APx500
User's Manual

for all APx Series audio analyzers
APx500 version 4.1

model APx555

8211.0238 rev 023

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Contents

Section II: Configuration

Section I: Introduction

1

General Information

Welcome to APx500

APx500 version 4.0 software

APx500 is the measurement software used with Audio Precision APx audio analyzers. APx500 version 4.0 brings big changes to the APx family, with a new look and two operating environments, Sequence Mode and Bench Mode. There is support for the APx555 and new hardware options, there are new measurements and new features for existing measurements and results.

See Chapter 3 for an in-depth tour of the APx analyzer software.

APx Analyzer Systems

Each member of the APx family of analyzers is a system comprised of a precision measurement instrument (the APx500 analyzer hardware) attached to a personal computer (PC) running APx500 measurement software. The instrument hardware provides analog and digital interface and system clocking. The APx500 software provides configuration, measurement, automation and reporting capabilities.

Hardware

Analyzer Instrument Hardware

APx500 series instruments include:

- the compact, fixed-configuration 2-channel APx515
- the 2-channel APx525
- the 2-channel output / 4-channel input APx526
- the 2-channel APx555. The APx555 analog input and output circuitry outperforms that of any other audio analyzer sold.
- the 2-channel output / 8-channel input APx582
- the 8-channel output / 8-channel input APx585, and
- the 8-channel output / 16-channel input APx586.

See your analyzer's *Installation Instructions and Specifications* booklet for safety, installation, and fusing information and personal computer hardware and operating system requirements.

Hardware Options

For the entire APx525/APx585 family:

- The APx-ADIO Advanced Digital I/O option (standard in the APx555).
- The APx-DSIO Digital Serial I/O option.
- The APx-HDMI HDMI+ARC option
- The APx-BT Bluetooth wireless technology option
- The APx-PDM Option
- The Advanced Master Clock option (standard in the APx555).

For instruments fitted with the 2-channel output module, including the APx582:

- The APx-AG52 analog generator option adds square wave generation and other improvements (standard equipment in the APx555).

For the instruments fitted with the 2-channel input module:

- The APx-BW52 high bandwidth option provides up to 1 MHz high-resolution analog input bandwidth (standard equipment in the APx555).

Auxiliary switchers, filters and the DCX-127 Multifunction Module are also available for all instruments.

See Chapter 2 for an in-depth tour of the APx analyzer hardware.

Analyzer Output signals

Analog signals

Audio output signals are, with some notable exceptions, generated in DSP in APx500 on the PC, and streamed down the USB cable to precision analog output circuits. The exceptions are the low-distortion analog sine generated in the APx555, the analog square wave generated in the APx555 and AG52, and Output

Equalization. These are implemented in instrument hardware.

Digital signals

Digital signals are also generated in DSP in the PC and streamed to a choice of digital interface transmitters in the instrument hardware.

Analyzer Input signals

Analog signals

Audio input signals are acquired using precision interfaces and eventually digitized. The data goes through some initial processing in the hardware, and then is streamed up the USB cable to the PC. The measurements are performed in APx500 DSP, using, for the most part, FFT technology. For THD+N measurements, the APx555 adds an analog notch filter implemented in instrument hardware.

Digital signals

Digital signals are input via a choice of digital interface receivers in the instrument hardware, and streamed to the PC for analysis in DSP.

Software

APx500 software supports all of the instruments in the APx family. APx can be addressed from Sequence Mode or from Bench Mode.

Getting Started

To make a measurement, you must have an APx analyzer attached to the PC. APx500 software will also run without hardware in "demo mode." No measurements can be made in demo mode, but you can explore the user interface and view the APx features and help content.

For complete installation, safety and fusing information and for detailed specifications, refer to the *Installation Instructions and Specifications* booklet provided with your instrument and available online at ap.com.

PC Requirements

APx500 requires a PC with the following characteristics and capabilities:

- Operating system: Microsoft Windows 8, Windows 7, or Windows Vista. Windows XP is no longer supported.
- A multi-core processor (at least dual-core) running at a clock speed of at least 2 GHz. Most current processors from Intel and AMD meet these requirements. Note: the Intel Atom processor does not meet our minimum specification.
- At least 2 GB of RAM. More is recommended.
- At least 300 MB of free hard disk space.
- A CD-ROM optical disc drive.
- A USB 2.0 port; two are required for optional switcher use.
- A color monitor and a video card with at least VGA capabilities. Video resolution of 1024 x 768 or greater is recommended.

System performance is sensitive to processor speed; faster processors will yield faster results.

APx500 is data intensive and it is recommended that other data-intensive applications not be run concurrently. This includes Audio Precision's AP2700, APWIN or ATS.

Embedded Help

You can access Help from the Help menu or by right-clicking and selecting Help, or pressing F1 from a context within the application. The Help system includes a Table of Contents, Index and Search capabilities.

Documentation and Support

The *APx500 User's Manual* (this document) is the primary documentation for the APx500 Series, in conjunction with the embedded help system accessible from within the application. The manual is provided as a PDF for the APx515 and as a PDF and a printed book for APx52x/555/58x users. Other documentation includes:

- *APx515 Installation Instructions and Specifications*
- *APx52x/58x Family Installation Instructions and Specifications*
- *APx555 Installation Instructions and Specifications*
- APx-DVD1
  APx-DVD1 is a playable video DVD with menu-driven linear and coded audio test signals for external source use with DVD players.
- APx-CD1
  APx-CD1 is a playable audio CD with linear audio test signals for external source use with CD players.

Audio Precision Web site at ap.com

For more information, go to ap.com/downloads/apx. You'll find manuals, audio test signals, Performance Check, utilities, APx sample projects, VB.NET / C#.NET / LabVIEW samples and more.

You can also contact our Technical Support staff at techsupport@ap.com, or by telephoning 503-627-0832 extension 4, or 800-231-7350 extension 4 (toll free in the U.S.A.).

2

Quick Tour: APx Analyzer Hardware

APx585 8-channel analog output, 8-channel analog input, DIO, DSIO, HDMI and Bluetooth

Introduction

The Audio Precision APx Series audio analyzers and the APx500 measurement software bring a fast, intuitive measurement-centric approach that is changing how audio test and measurement is done. This chapter provides a brief overview of the system hardware and will point you to more information.

For safety, fusing or installation information, and for detailed specifications for any of the following instruments, see the corresponding APx Installation and Specifications Booklet.

APx515

The APx515 is a fixed-configuration audio analyzer perfect for both bench and production test.

Analog I/O

The APx515 has two analog output channels, available with both balanced and unbalanced connections; and two analog input channels, each balanced or unbalanced.

Digital I/O

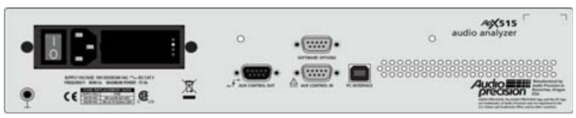
The APx515 is fitted with AES3/SPDIF digital input and output on balanced electrical, unbalanced electrical and optical connectors.

See the *APx515 Installation Instructions and Specifications* booklet for full specifications.

Rear Panel

Power entry module

The power entry module is mounted on APx515 rear panel, and provides the mains power switch, the connector for the mains power cable, the mains voltage selector and the line fuse assembly.



Power and fuse ratings are printed below the power entry module. See the *APx515 Installation Instructions and Specifications* booklet for more information about mains power and fusing.

PC interface

The PC interface connector provides a USB 2.0 interconnection to the controlling personal computer.

Aux control

The two Aux Control connectors (sometimes referred to as GPIO or General Purpose Input/Output) provide the capability to communicate with external devices, transmitting and receiving control commands. See Chapter 96 for more information about Aux Control.

Software options for the APx515

The APx515 enables a basic set of APx500 features and measurements in Sequence Mode. This set can be expanded to the full complement of instrument-appropriate features and measurements (including Bench Mode) by the installation of iButton software option keys. Factory-keyed iButtons are mounted in the APx software option module (pn BSW0.0000). The software option module is connected to the Software Options connector on the APx515 rear panel. Go to page 166 for more information about software options in the APx515.

APx52x/58x family

The APx52x/58x family of audio analyzers provides a modular framework for a wide selection of input/output configurations. Analog I/O is available with as few as 2 input and 2 output channels, up to 8 output and 16 input channels. The standard DIO module provides AES3/SPDIF digital input and output, to which you can add DSIO (chip-level digital serial interface), HDMI+ARC, PDM or Bluetooth wireless technology, provided there is an open module bay in the lower tier.

See the *APx52x/58x Family Installation Instructions and Specifications* booklet for full specifications.

Power switch and indicator light

The master power switch and the power indicator light for the APx52x/58x family are located in the blue module at the right end of the front panel.

APx525

The APx525 is the classic 2-channel configuration, fitted with a Model 103/203 2-channel analog output module, a Model 104/204 2-channel analog input module, and the Model 110/210 DIO module. Read about these I/O modules in detail in the next few pages.

APx526

The APx526 is a 4-channel (analog input) version of the APx525, with 2 analog outputs and 4 analog inputs.

On the analog side, the APx525 has balanced and unbalanced analog inputs and outputs, relay-implemented I/O loopback and common-mode rejection measurements. The digital I/O provides AES3 balanced input and output on XLR connectors, SPDIF/SMPTE unbalanced I/O on BNCs, and optical I/O on Toslink connectors, at sample rates up to 216 kHz. All APx500 generation and measurement features that are applicable to a 2-channel instrument are supported.

APx582

The APx582 is a popular combination of the analog output features of the APx525, and the analog input features of the APx585, described below.

APx585

The APx585 is the world’s first true multichannel audio analyzer.

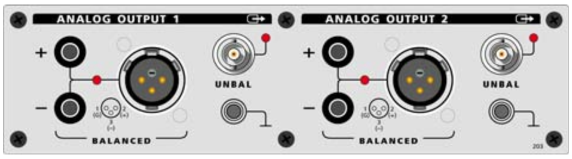
The typical configuration features 8 simultaneous channels of balanced or unbalanced analog inputs and outputs with 216 kHz AES3, SPDIF, and optical digital I/O.

APx586

The APx586 is similar to the APx585, but with 8 additional analog input channels on a Model 106/206 module. This provides a total of 16 analog input channels.

Analog I/O for the APx52x/58x family

2-channel analog outputs



The Model 103/203 2-channel analog output module is used in the APx520, 521, 525, 526 and 582 analyzers.

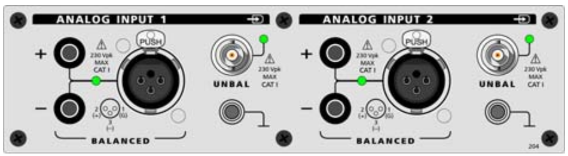
The module has two unbalanced analog output channels available on BNC connectors. Each unbalanced output has selectable source resistances and a maximum output level of 10.61 Vrms.

The module also has two balanced analog output channels available on XLR-M and dual-banana connectors (connected in parallel). Each balanced output has selectable source resistances and a maximum output level of 21.21 Vrms.

AG52 analog generator option

The AG52 analog generator option is available for the APx525 and 526 analyzers, and is provided as standard equipment in the APx555 and APx582 analyzers. AG52 improves THD+N residual performance, increases maximum generator amplitude (26.66 Vrms balanced, 13.33 Vrms unbalanced), and adds a fast rise time square wave generator for general purpose testing and for generating a DIM/TIM distortion stimulus.

2-channel analog inputs



The Model 104/204 2-channel analog input module is used in the APx525 and 526 for analog inputs 1 and 2. The electronically identical Model 107/207 module is used in the APx526 for Analog Inputs 3 and 4.

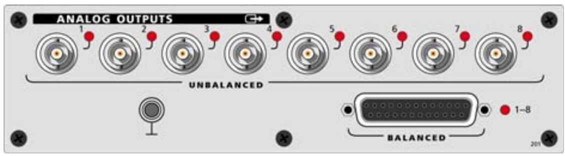
The module has two unbalanced analog input channels available on BNC connectors. Each unbalanced input can be DC or AC coupled, with selectable termination resistances and a maximum input level of 160 Vrms.

The module also has two balanced analog input channels available on XLR-F and dual-banana connectors (connected in parallel). Each balanced input can be DC or AC coupled, with selectable termination resistances and a maximum input level of 300 Vrms. Two ground lugs are provided.

BW52 high-bandwidth input option

The BW52 high bandwidth analog input option is available to the APx525 and 526, and the performance features of the BW52 are standard in the APx555. The BW52 expands the analyzer bandwidth of analog inputs 1 and 2 to up to 1 MHz, providing unparalleled 1 MHz 24-bit FFTs.

8-channel analog outputs



Unbalanced

The Model 101/201 is used in the APx585 and 586 analyzers. The module has eight unbalanced analog output channels available on floating BNC connectors. Each unbalanced output has a source resistance of 50 Ω and a maximum output level of 7.2 Vrms.

Balanced

The module also has eight balanced analog output channels available on a single 25-pin D-Sub connector; see page 11. Each balanced output has a source resistance of 100 Ω and a maximum output level of 14.40 Vrms. A ground lug is provided.

8-channel analog inputs

The Models 105/205 and 106/206 8-channel modules are functionally identical, their only difference being front-panel nomenclature. Model 105/205 is used on the APx582, 585 and 586 for input channels 1–8, while Model 106/206 is used on the APx586 for channels 9–16.

Unbalanced

Each module has 8 unbalanced analog input channels available on floating BNC connectors. Each unbalanced input is DC coupled, with a termination resistance of 100 kΩ and a maximum input level of 115 Vrms.

Balanced

Each module also has 8 balanced analog input channels available on a single 25-pin D-Sub connector; see page 11. Each balanced input is DC coupled, with a termination resistance of 100 kΩ, each leg to ground, and a maximum input level of 115 Vrms.

APx555

The APx555 is the most versatile, highest-performance audio analyzer Audio Precision has ever made. Built on the scaffolding of the APx525, the APx555 includes an analog High Performance Sine Generator and a hybrid High Performance Sine Analyzer, which together drop the residuals in a THD+N measurement to unmatched lows. The High Performance Sine Generator is of extraordinarily low distortion and noise, and has a maximum frequency of 204.75 kHz, well beyond the 80.1 kHz maximum of the DAC generator.

The Advanced Master Clock is standard on the APx555, and provides jitter generation and analysis capability when used with a supporting jitter-enabled digital module (currently ADIO and DSIO). The Advanced Master Clock also provides an AES11 Digital Audio Reference Signal (DARS) reference in and out, a clock sync in and out, and a trigger in and out, all on a connector panel on the rear of the instrument.

The Advanced Digital Input/Output (ADIO) is a two-channel AES3/SPDIF interface with balanced, unbalanced and optical I/O, which is similar to the Digital Interface Module (DIO) module but with additional features. ADIO adds metadata impairment, digital interface signal impairment, digital interface common mode measurement, and the provides digital interface level data to the analyzer. ADIO includes jitter generation and analysis capability when used with the Advanced Master Clock.

All APx500 generation and measurement features that are applicable to a 2-channel instrument are supported.

Analog I/O for the APx555

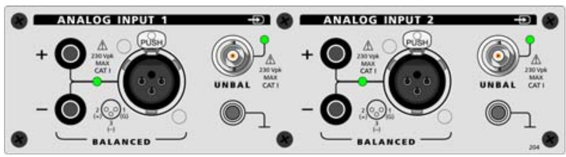
2-channel analog outputs

The Model 203 2-channel analog output module is also used in the APx555 analyzers, but the high performance features of the AG52 option are included, with the fast rise time square wave generator for general purpose testing, generating a DIM/TIM distortion stimulus and sine burst signals. Additionally, the High Performance Sine Generator provides an extraordinarily pure sine stimulus at frequencies as high as 204 kHz.

The module has two unbalanced analog output channels available on BNC connectors. Each unbalanced output has selectable source resistances and a maximum output level of 13.33 Vrms.

The module also has two balanced analog output channels available on XLR-M and dual-banana connectors (connected in parallel). Each balanced output has selectable source resistances and a maximum output level of 26.66 Vrms.

2-channel analog inputs



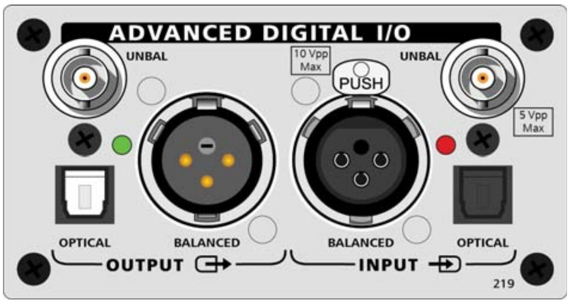
The Model 104/204 2-channel analog input module is also used in the APx555 analog inputs. The features of the BW52 option are included, expanding the analyzer bandwidth of the analog inputs up to 1 MHz, providing unparalleled 1 MHz 24-bit FFTs. Additionally, the High Performance Sine Analyzer uses stereo analog notch filters with a dedicated ADC to provide a system THD+N residuals specification unmatched in the industry.

The module has two unbalanced analog input channels available on BNC connectors. Each unbalanced input can be DC or AC coupled, with selectable termination resistances and a maximum input level of 160 Vrms.

The module also has two balanced analog input channels available on XLR-F and dual-banana connectors (connected in parallel). Each balanced input can be DC or AC coupled, with selectable termination resistances and a maximum input level of 300 Vrms. Two ground lugs are provided.

Digital I/O for APx modular analyzers

Advanced Digital I/O



The Model 219 Advanced Digital I/O module with AES3 interface is the standard digital interface for the APx555 audio analyzer, and is an option for the APx52x/58x instruments.

This module provides both digital input and output on balanced electrical, unbalanced electrical and optical connectors.

Impairments and interface measurements

The Advance Digital I/O differs from the standard Digital I/O in that it offers impairments to the transmitted digital interface signal, and the ability to measure certain characteristics of the received interface signal.

The transmitted interface signal can be impaired with false metadata, validity and parity errors, jitter, noise, common mode and risetime impairments (including cable simulation), to test the tolerance of a downstream device. Measured input interface voltage level can be measured and displayed.

Jitter

The transmitted interface signal can be impaired with jitter, and received jitter can be analyzed in great detail.

Balanced

The balanced interface satisfies AES3, EBU-3250 and IEC60958-4.

The balanced interface has a nominal carrier level of 5 Vpp. Output source resistance is nominally 110 Ω. Input termination resistance can be set to 110 Ω or off (~2 kΩ). When the balanced output connector is selected, the APx500 software embeds Professional status bits in the output signal.

Unbalanced

When the unbalanced output is set to Consumer mode (the default), the unbalanced output is compatible with SPDIF and satisfies IEC60958-3. Nominal carrier level is 0.5 Vpp and output status bits are set to Consumer.

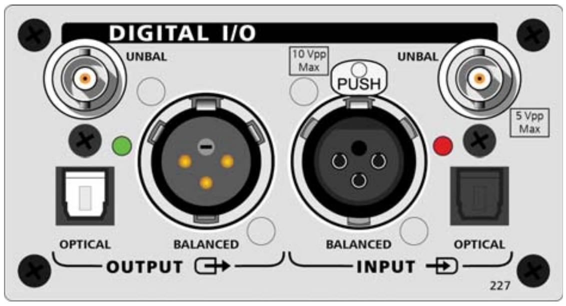
When the unbalanced output is set to Professional mode (see Advanced Settings), the unbalanced output satisfies AES3-id and SMPTE-276M, while remaining largely compatible with IEC-60958-3 and most SPDIF receivers. Nominal carrier level is 1.0 Vpp, and output status bits are set to Professional.

Output source resistance is nominally 75 Ω. Input termination resistance can be set to 75 Ω or OFF (~23 kΩ).

Optical

Compatible with the Toslink interface. When the optical output connector is selected, output status bits are set to Consumer.

Digital input/output



The Model 110/210 Digital I/O module with AES3 interface is the DIO for the modular APx500 series audio analyzers, and is a standard installation for the APx525, 526, 582, 585 and APx586 instruments.

This module provides both digital input and output on balanced electrical, unbalanced electrical and optical connectors.

NOTE: Early APx585 and 586 instruments may be fitted with a Model 109 Digital I/O module, which is not equipped with balanced digital input or output connectors.

Balanced

The balanced interface satisfies AES3, EBU-3250 and IEC60958-4.

The balanced interface has a nominal carrier level of 5 Vpp. Output source resistance is nominally 110 Ω. Input termination resistance can be set to 110 Ω or off (~2 kΩ). When the balanced output connector is selected, the APx500 software embeds Professional status bits in the output signal.

Unbalanced

When the unbalanced output is set to Consumer mode (the default), the unbalanced output is compatible with SPDIF and satisfies IEC60958-3. Nominal carrier level is 0.5 Vpp and output status bits are set to Consumer.

When the unbalanced output is set to Professional mode (see Advanced Settings), the unbalanced output satisfies AES3-id and SMPTE-276M, while remaining largely compatible with IEC-60958-3 and most

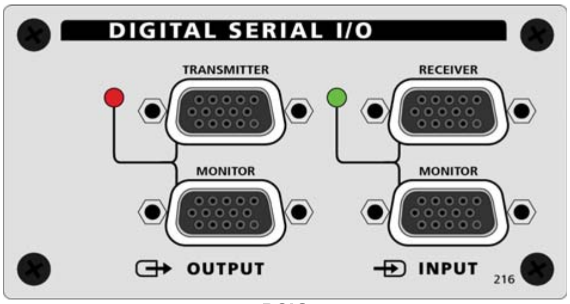
SPDIF receivers. Nominal carrier level is 1.0 Vpp, and output status bits are set to Professional.

Output source resistance is nominally 75 Ω. Input termination resistance can be set to 75 Ω or OFF (~23 kΩ).

Optical

Compatible with the Toslink interface. When the optical output connector is selected, output status bits are set to Consumer.

Digital Serial I/O option



DSIO

The Digital Serial Input/Output (or DSIO) option provides a flexible chip- or board-level serial input and output interface.

With separate Master Clock, Bit Clock, Frame Clock, Channel Clock and four Data lines, variable signal formats, variable word width, bit depth and synchronization options, the DSIO can address almost any serial interface need.

Formats include TDM, I^2S, DSP (bit-wide pulse) and custom formats. Up to 16 channels can be transmitted and received using the TDM format.

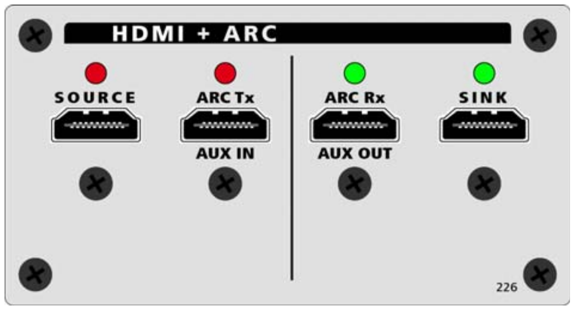
The DSIO option can be ordered with APx525, 526, 582, 585 or 586 instruments, provided there is an open module bay in the lower tier. A combination DIO / DSIO module (model 111/211) is also available.An upgrade kit for these instruments is also available.

Jitter

If the analyzer is fitted with the Advanced Master Clock (standard on the APx555), the DSIO transmitter can be jittered, and jitter in the received signal can be measured.

Operation of the DSIO is discussed in Chapter 14.

HDMI+ARC option



The APx525, 582 and 585 can be fitted with an optional model 114/214 HDMI+ARC (High Definition Multimedia Interface plus Audio Return Channel) I/O module.

> *NOTE: Earlier APx585 instruments may be fitted with a Model 112 HDMI module, which does not include ARC support.*

About HDMI+ARC

The model 114/214 HDMI+ARC module is fully compatible with HDMI 1.3a; additionally, it supports a subset of HDMI 1.4a, the ARC (Audio Return Channel) feature.

HDMI (High Definition Multimedia Interface) is designed to carry high-bandwidth digital streams providing an audio/video interface that includes content protection and a bi-directional channel for interaction with connected electronic devices.

ARC (Audio Return Channel) provides an additional digital audio channel, which can simplify interface cabling in certain applications, for user convenience.

HDMI+ARC Hardware description (214)

The HDMI option is fitted with four HDMI Type A connectors:

- The Source connector outputs APx-generated monocolor video with APx generated audio and HDCP. Alternatively, user video (from Aux In) can be substituted for monocolor video. This connector is inactive when the Signal Path Setup Output Configuration is set to HDMI ARC Tx.
- The ARC Tx / Aux In connector provides input for user video (such as a test pattern) to be substituted for APx-generated video available at Source connector.
  When Signal Path Setup Output Configuration is set to HDMI ARC Tx, this function is disabled; instead, the connector outputs digital audio and CEC negotiation in support of HDMI ARC.
- The ARC Rx / Aux Out connector provides a monitor function, outputting the video and audio present at the Sink connector. EDID metadata from downstream device (typically an HDTV monitor) can be selectively blocked (the default) or passed

to the upstream Device Under Test.
When Signal Path Setup Input Configuration is set to HDMI ARC Rx, this function is disabled; instead, the connector inputs digital audio in support of HDMI ARC.

- The Sink connector provides measurement input for HDMI from the Device Under Test.
  This connector is inactive when the Signal Path Setup Input Configuration is set to HMI ARC Rx.

CEC

For new instruments and upgrades sold after September, 2013, CEC functionality has been added to Source and Sink connections. ARC Tx and ARC Rx CEC functionality has been expanded. APx500 version 3.4 or later is required. See More About CEC on page 120.

HDMI (112)

Older APx585 instruments may have been fitted with the model 112 HDMI module, which requires APx500 v2.2 or later.

The model 112 HDMI module is fully compatible with HDMI 1.3a.

The front panel is fitted with four Type A HDMI connectors:

- The Source connector outputs APx-generated monocolor video with APx generated audio and HDCP. Alternatively, user video (from Aux In) can be substituted for monocolor video.
- The Aux In connector provides input for user video (such as a test pattern) to be substituted for APx-generated video available at Source connector.
- The Aux Out connector provides a monitor function, outputting the video and audio present at the Sink connector. EDID metadata from downstream device (typically an HDTV monitor) can be selectively blocked (the default) or passed to the upstream Device Under Test.
- The Sink connector provides measurement input for HDMI from Device Under Test.

Audio on HDMI

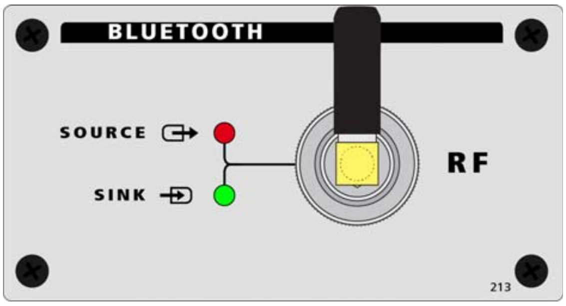
APx500 provides generation and measurement capabilities for audio signals carried on the HDMI interface.

Video is implicit in HDMI, and the AP HDMI interface transmits and receives the required video and metadata signals for proper operation of the HDMI interface. These video and metadata signals can be altered in APx500 in some ways to examine the effect of bitstream changes on audio quality. However, APx500 does NOT provide any measurement or diagnostic capabilities for HDMI video signals, or for all HDMI metadata.

Audio on ARC

ARC (Audio Return Channel) is a feature of HDMI 1.4a that enables digital audio (IEC 60958 / SPDIF) transmission and reception on existing but previously unused conductors in standard HDMI connectors and cables. See More About ARC on page 119.

APx Bluetooth Option



Hardware description

The APx Bluetooth Option module (APX-BT) can be fitted into any instrument in the APx52x/555/58x families, provided there is an open module bay in the lower tier. A combination DIO / Bluetooth module (model 217) is also available.

The Type N connector on the front of the module provides RF (radio frequency) connectivity. Typically, a small antenna is connected using an adapter. A 50 Ω cable can also be connected to route the RF signal to a Bluetooth device with an RF cable connection.

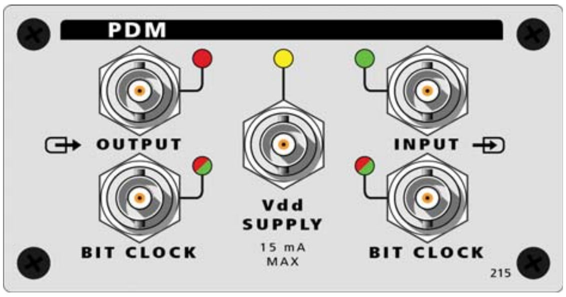
When cabling directly to a device (no antennas), the use of an RF level attenuator of approximately 50 dB is recommended, to be connected between the APx instrument and the device.

Bluetooth codecs

Bluetooth wireless technology uses codecs to compress the audio to usable data rates. All Audio Precision Bluetooth Option modules support the CVSD, SBC and aptX codecs. A revision to the Bluetooth Option module hardware (APX-BT-WB) adds the mSBC (wideband speech) codec.

To check the codec support of your Bluetooth module, in APx500 go to Help > About > Bluetooth Module. Scroll to the “Codecs:” entry to read the supported codecs.

APx PDM Option



The PDM option provides a complete solution for addressing circuits or devices with a PDM input or output. The nomenclature for the PDM option module is 215. The PDM module can be installed in any APx instrument except the APx515, provided there is an open module bay in the lower tier. A combination DIO / PDM module (model 218) is also available.

Output

The PDM Option provides a PDM signal output, which consists of an APx generator audio signal, interpolated by a broad choice of oversampling ratios, and modulated into a 1-bit PDM bitstream. A 4th-order modulator is the default; a 5th-order modulator can be selected.

Bit Clock

A PDM transmitter (output) typically acts as a slave, with the downstream receiver acting as master. Therefore, in the APx implementation the Bit Clock connection associated with the PDM Output is, by default, configured as a clock input; however, it can also be configured as a clock output by changing the clock direction in software. The LED associated with the Bit Clock connector indicates the clock direction: red when Bit Clock is a clock output, green when it is a clock input.

Bit clock frequencies can range from 128 kHz to 24.576 MHz, depending upon the interpolation rate (baseband audio sample rate) and oversampling ratio. This constrains the higher oversampling ratios to Decimated Rates at or below 96 kHz.

Input

The PDM Option also provides a signal input with its associated clock connection. The input accepts a 1-bit PDM bitstream, which is then decimated by one of a wide range of oversampling ratios and filtered into baseband audio at the Decimated Rate. The input bitstream can also be analyzed directly (before decimation) in the Signal Analyzer to view out-of-band components.

Bit Clock

A PDM receiver (input) typically acts as a master, with the upstream transmitter acting as a slave. Therefore, in the APx implementation, the Bit Clock connection associated with the PDM Input is, by default, configured as a clock output; however, it can also be configured as a clock input by changing the clock direction in software. The LED associated with the Bit Clock connector indicates the clock direction: red when Bit Clock is a clock output, green when it is a clock input.

Bit clock frequencies can range from 128 kHz to 24.576 MHz, depending upon Decimated Rate (baseband audio sample rate) and oversampling ratio.This constrains the higher oversampling ratios to Decimated Rates at or below 96 kHz.

Vdd Supply

PDM MEMS microphones devices are typically externally powered. The Vdd Supply connector can provide DC current up to 15 mA, with a voltage range of 0.8 VDC to +3.60 VDC.

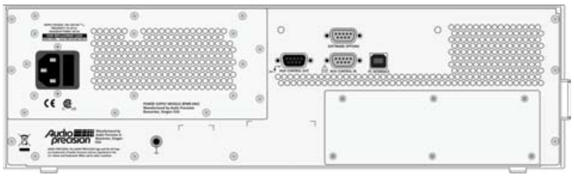
PSR testing

AC signals can be imposed upon the Vdd supply dc voltage to test the PSR (Power Supply Rejection) in the DUT. This connection can also be used for PSRR testing with certain analog DUTs. Use PSR or PSR Sweep to measure the PSR or PSRR.

APx52x/555/58x family rear panel

Power entry module

The power entry module is mounted on the power supply module. The power entry module provides the connector for the mains power cable and the line fuse assembly.



Power and fuse ratings are printed to the right of the power entry module. See the instrument Installation and Specifications booklet for more information about mains power and fusing.

PC interface

The PC interface connector provides a USB 2.0 interconnection to the controlling personal computer.

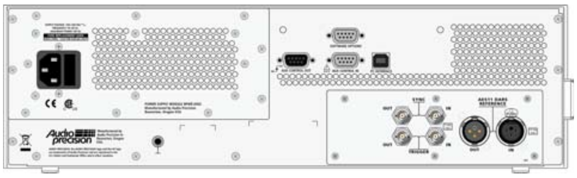
Configuration label

Configuration, serial number and date of manufacture are listed on the configuration label.

Aux control

The two Aux Control (sometimes referred to as GPIO or General Purpose Input/Output) connectors provide the capability to communicate with external devices, transmitting and receiving control commands. See Chapter 96 for more information about Aux Control.

Advanced Master Clock

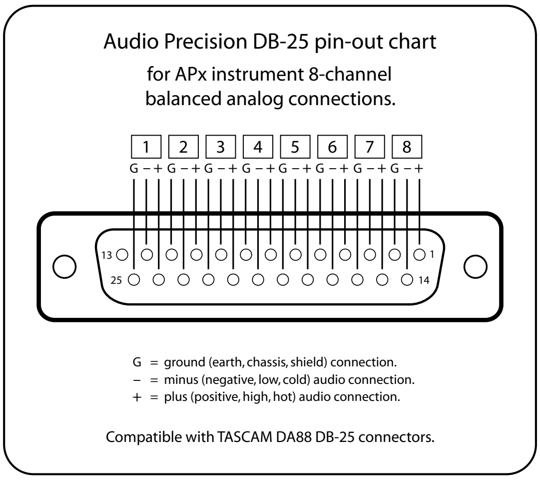


Analyzers fitted with the Advanced Master Clock (standard on the APx555 and optional on other models) will have Sync, Trigger and Reference connectors on the rear panel.

The Advance Master Clock also provides jitter generation and measurement capabilities when used with a jitter-compatible digital I/O module.

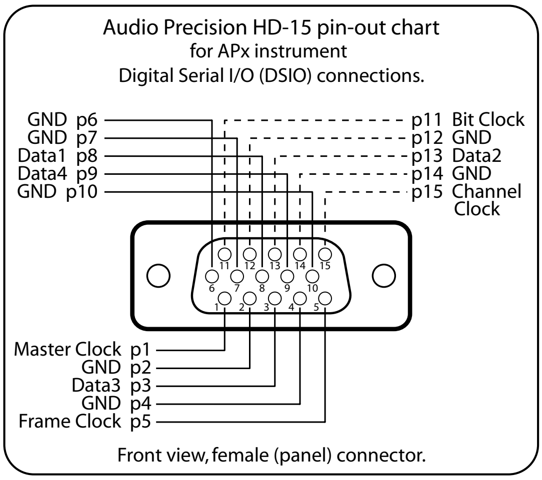
DB-25 Analog Balanced Connector
(used in 8-channel analog I/O modules)

Audio Precision's 8-channel analog input and 8-channel analog output modules use a female DB-25 connector for balanced multichannel inputs and outputs. Pin connections are shown below. Compatible cables and adapters of high performance are available from Audio Precision.



HD-15 Connector
(used in the DSIO module)

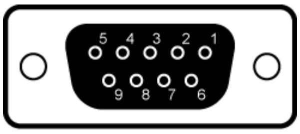
The Digital Serial I/O option is fitted with four female HD-15 connectors for multichannel serial interface transmitter, receiver and monitor connections. Pin connections are shown below. Compatible cables are available from Audio Precision.



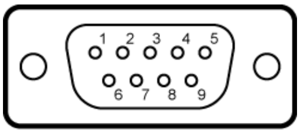
DE-9 connectors

used in all analyzers (rear panel)

Aux Control Out is available on a female DE-9 connector on the rear panel of all APx analyzers. The female DE-9 has the following pin-out:



Aux Control In is available on a male DE-9 connector on the rear panel of all APx analyzers. The male DE-9 has the following pin-out:



Accessories

AUX-0025 filter

The AUX-0025 switching amplifier filter is a 2-channel switching amplifier measurement filter for use with the APx analyzers when testing high slew rate signals, typical of switching amplifier (Class D) outputs.

The AUX-0100 filter

The AUX-0100 filter (previously known as the APx581 filter) provides the same filtering in a 8-channel product, and may be a more appropriate choice for use with the APx582, 585 or 586 analyzers.

For more information about the AUX-0025 or AUX-0100 switching amplifier filters, contact your Audio Precision distributor or representative or visit our Web site at ap.com.

SWR-2755 switchers

Audio Precision manufactures optional audio switchers that can be used to connect and disconnect many signal channels (up to 192 input and 192 output channels) to the instrument inputs or outputs, under software control. See Chapter 97 for more information about switchers.

DCX-127 Multifunction Module

The DCX-127 Multifunction Module is an Audio Precision accessory that provides interface and control features not otherwise available in Audio Precision analyzers. Download the *DCX-127 Multifunction Module User's Guide* from the Audio Precision Web site at ap.com for complete operational information for the DCX-127.

The DCX-127 provides a digital multimeter that can be configured to measure DC voltages or resistance; it has two programmable DC voltage sources, and four GPIO Auxiliary Output connections.

3

Introduction to the APx500 Software

Welcome to APx500 v 4.0

APx500 is the measurement software used with Audio Precision APx500 audio analyzers. APx500 analyzers are systems, comprised of a precision measurement instrument (the APx500 analyzer hardware) attached to a personal computer (PC) running APx500 measurement software.

What's New?

APx500 v 4.0 is a major release that brings an entirely new paradigm for user interaction. There are now two operational modes for APx500: Sequence Mode and Bench Mode.

Sequence Mode and Bench Mode

Sequence Mode and Bench Mode each have their own workspace. APx normally launches into Sequence Mode. You can switch modes using the selector in the upper right corner of the workspace.

Sequence Mode

This chapter begins with a tour of Sequence Mode on the next page. If you are an experienced APx500 user, you will already be familiar with Sequence Mode.

If you are new to APx, Sequence Mode brings a number of pre-defined measurements to hand. For many tests, it's as simple as opening a measurement and clicking Start. Measurements can be arranged in Signal Paths and Sequences for easy automation and comprehensive reporting.

Sequence Mode is already familiar to many of our users as the only APx interface. Sequence Mode provides easy access to a large number of pre-configured measurements, which can be quickly arranged into an automated sequence that can generate sophisticated reports.

Bench Mode

Bench Mode, on the other hand, is largely unconfigured. Rather than using pre-defined measurements, Bench Mode makes a number of settings and tools available, and allows you to simply adjust and tweak and try different approaches without imposing constraints.

This chapter will look at the key features of Bench Mode beginning on page 20, and will also visit other APx features that are shared by both Sequence Mode and Bench Mode.

Signal Path Setup revised

Signal Path Setup now includes a menu to select a number of related panels: Input/Output, References, and Switchers. If an Audio Precision DCX-127 is attached, there is a new menu item to control it. If your instrument includes the Advanced Master Clock (AMC), Ref, Sync and Jitter selections are included in the menu.

The References panels, which used to reside in the Navigator, are now consolidated into a menu item in the new Signal Path Setup menu.

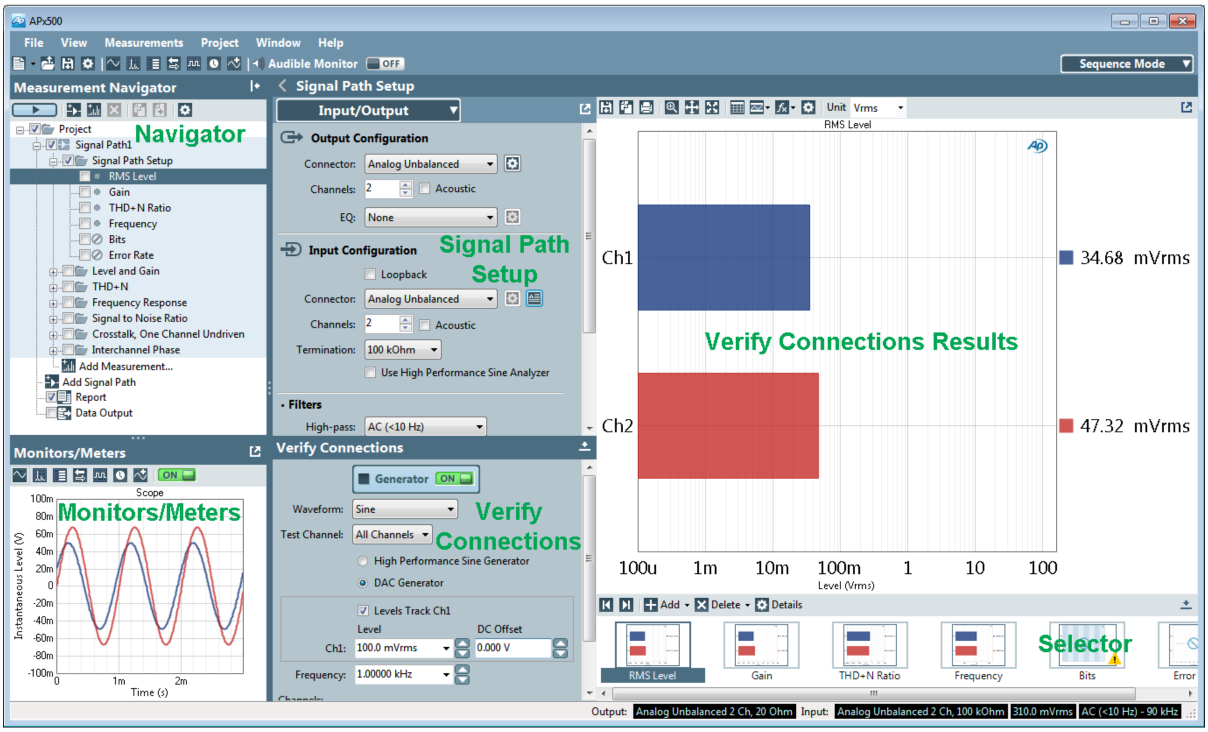
Hardware Support

Additionally, APx 4.0 supports the hardware improvements introduced in the APx555 instrument, including the 555's High Performance Sine Generator and High Performance Sine Analyzer.

APx 4.0 also supports the new Advanced Digital Input/Output and the Advanced Master Clock hardware, which is standard equipment for the APx555, and optionally available for the rest of the APx modular family (which excludes the APx515).

The Sequence Mode Workspace: Signal Path Setup

SEQUENCE MODE WORKSPACE (1)



All APx500 workspaces have a Menu bar and a Toolbar at the top, and a Status bar at the bottom.

The Measurement Navigator panel is on the left, above the Monitors/Meters panel. The Signal Path Setup menu runs down the center of the workspace, and the measurement Results, Selector and Data Sets grid are on the right.

The Measurement Navigator is key to Sequence Mode. Here you add pre-defined measurements to Signal Paths, and select the results you'd like to see. You can create multiple Signal Paths, with different combinations of measurements or even different input/output configurations.

Once you've created a Signal Path or two and populated them with measurement you can run them all in any order as an automated sequence. You can add stops and prompts and run-time configurations to your sequence, and customize a report to be generated upon completion.

The Signal Path Setup menus

The center panel is Signal Path Setup, and the menu at the top of the panel provides configuration settings for the current signal path. Depending upon the connected APx instrument hardware, these menus may include

- Input/Output
- References
- Output Switchers
- Input Switchers
- DCX
- Clocks
- Triggers

The Signal Path Setup menus are also available in Bench Mode.

Signal Path Setup is discussed in detail in Chapter 6.

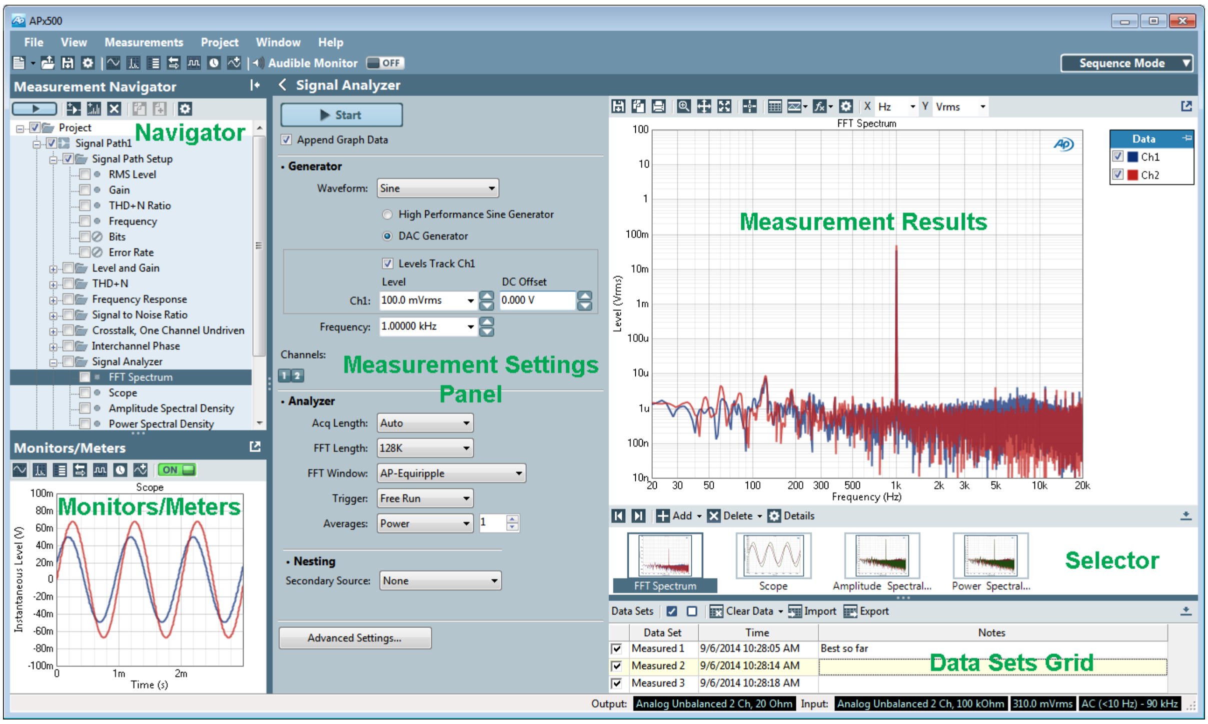
Verify Connections (Sequence Mode only)

Verify Connections provides generator and analyzer results enables you to easily check your connections and make simple diagnostic check by sending and receiving signals on the channels you are using.

Verify Connections is discussed in detail in Chapter 7.

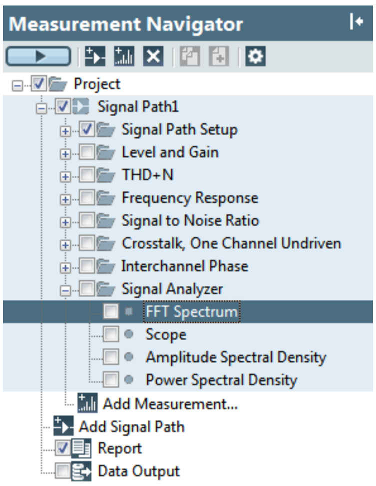
The Sequence Mode Workspace: Making a Measurement

SEQUENCE MODE WORKSPACE (2)



When you select a Measurement view, Signal Path Setup and Verify Connection disappear, replaced by the Measurement Settings Panel, the Results and Selector, and for some measurements, the Data Sets grid.

The Navigator (Sequence Mode only)



This panel has two key functions.

It serves as a Measurement Navigator, where you can add, delete or rename measurements and results, displayed in a tree structure.

It also acts as a Sequencer, an automation interface where you can stage signal paths and measurements, add prompts and build a sophisticated automated Sequence, all from within the APx500 graphical interface. See Chapter 80 for detailed information about creating and using an automated Sequence.

You will find this panel referred to as simply the Navigator, or sometimes just the Sequencer, depending upon the context and the task at hand.

The Navigator panel is at the left of the APx500 workspace. Click the **Hide** button to hide the panel; click **Show Measurement Navigator** to restore it to view. Note that the show and hide buttons also affect the visibility of the Signal Monitors when they are docked.

What is a Signal Path?

In Sequence Mode, a Signal Path is a combination of input and output format and connector settings, reference level settings and a collection of measurements and measurement settings.

A new project opens with one signal path. You can add, move, rename and delete signal paths from the Navigator panel. Multiple signal paths in a project are useful for:

- testing DUTs that have more than one signal path; for example, analog in / digital in, or 5.1 channels out / stereo out;
- testing the same signal path on a DUT with a different configuration; for example, all levels at 0 dBV versus all levels at –20 dBV;
- cycling through different switcher settings.

A sequence can run through some or all of the signal paths within a project. Use the sequence step checkboxes in the Navigator to add or remove measurements in different signal paths, just as you would add or remove measurements within one signal path.

Remember to change physical connectors as needed when you change signal paths. If a connection change is required in a sequence, you should insert a user prompt in the sequence to remind the operator and to provide time to change the connectors.

Measurements (Sequence Mode only)

In Sequence Mode, APx500 provides many different testing tools called *measurements*, such as Signal-to-Noise Ratio or THD+N. Chapter 20, Making Measurements, covers general topics regarding making Sequence Mode measurements. Each Sequence Mode measurement is discussed in detail in Chapters 21 through 79.

You can add any number of measurements to a Signal Path, including more than one instance of the same measurement. You can remove measurements from a Signal Path, and you can rename each measurement to match your testing requirements.

In a new Sequence Mode Project, there are six key measurements initially available:

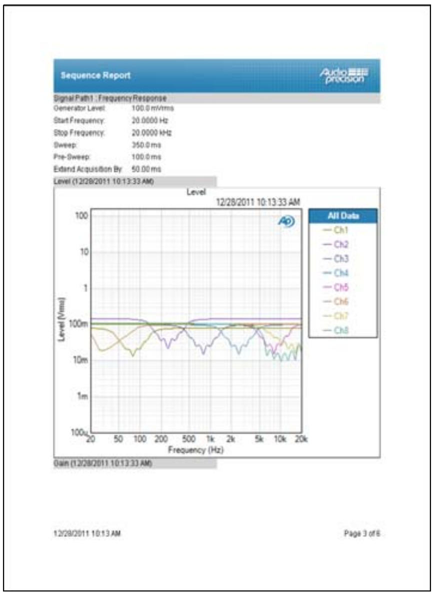
- Level and Gain (see Chapter 52)
- THD+N (see Chapter 78)
- Frequency Response (see Chapter 43)
- Signal to Noise Ratio (see Chapter 74)
- Crosstalk (see Chapter 34)
- Interchannel Phase (see Chapter 49)

Note: some measurements are unavailable in certain configurations, and are marked with a symbol in the Navigator.

Result branches in the Navigator

Each measurement has one or more primary results and optional derived results associated with it. Results are represented in the Navigator tree as branch nodes and are also shown as thumbnails in the Selector. See Chapters 93 and 95 for more information about results.

Reports (Sequence Mode only)



A report is the primary output document of a Sequence, listing setup, measurements and results. A report is created when a Sequence is run, or when a measurement is made using the **Start Measurement** command from the Navigator context menu or the Project menu.

You can create a report showing the results in the current view; you can run a measurement and create a report showing results of all the selected views; or you can run a sequence and create a report showing the results of all the selected measurements. You can view, edit and format a report, and then export or print it.

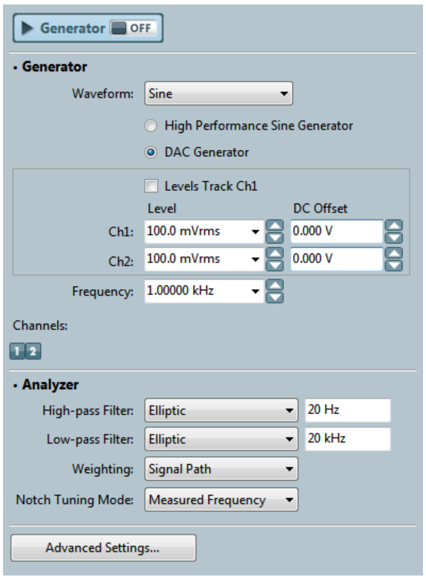
Click on the **Report** node in the Navigator to create a report, or right-click on the node to edit the report's properties. Using APx500's Microsoft Word report feature, you can also create custom reports and report layouts.

See Chapter 81 for more information about reports.

Data Output (Sequence Mode only)

The Data Output feature can save certain sequence results to a CSV file as tabular data. Each time the sequence is run, the target file is opened, and the new data is appended to the table.

The Measurement Settings panels (Sequence Mode only)



A typical Measurement Settings panel

In a Sequence Mode measurement, the Measurement Settings panel runs down the middle of the APx500 workspace. The contents of the panel vary with the measurement or configuration view selected. In general, there are generator settings at the top of the panel (when supported) and analyzer settings (if settings are available) below that.

Start On/Off Button

Most measurements have a Start or Generator On/Off button, usually at the top of the panel.

Generator settings

Depending upon the measurement selected, the Measurement Settings panel may contain generator waveform source, level, frequency and channel settings. Sweep measurements may include start and stop frequencies or levels, step size, points, logarithmic or linear settings and more. See Chapter 5 for more information about the APx Generator.

Analyzer settings

Depending upon the measurement selected, the Measurement Settings panel may contain acquisition and analysis settings including filters, measured or reference channel settings, append settings, regulation controls, FFT settings, reading rate settings and more.

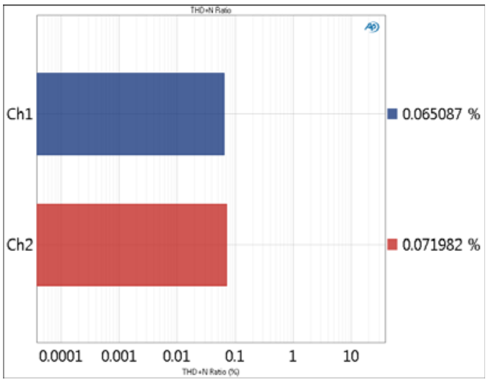
Advanced settings

An Advanced Settings button opens a dialog that allows you to view and set other generator and analyzer settings for the measurement.

Results and the Graph panel

Measurement results are represented as thumbnail views in the measurement's Selector filmstrip display (see page 19) and as branches in the Navigator tree display (page 15).

When a result is selected, the meter bar graph or XY line graph is shown in the main Graph panel (below). The Graph panel can be undocked and repositioned and resized as an independent window. The Selector thumbnails can also be undocked, and become full-featured graph windows.

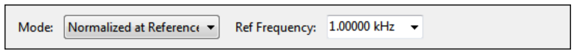


Typical Bar Graph Result

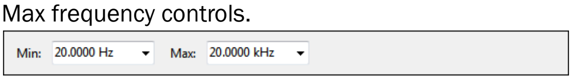
The Result Settings bar

Some measurements present a Result Settings bar between the results Graph panel and the Selector. Different results may offer different controls in the Result Settings bar.

For example, Frequency Response: Relative Level shows the Ref Frequency control, the necessary reference for the relative result view.



Frequency Response: Deviation shows the Min and Max frequency controls.



Adjusting a results setting changes the results view, but this is non-destructive; the original measurement acquisition data are retained.

See Chapter 20 for more information about making measurements.

Working with the Navigator/ Sequencer

Moving around in the Navigator

To expand a branch (or *node*) of the Navigator tree, click on a plus-sign expansion box ⊞. To contract an expanded branch, click on a minus-sign contraction box ⊟.

Or, right click on a Navigator node and select Expand All or Collapse.

Selecting a measurement

To select a measurement, click the measurement branch. The primary result for that measurement will open in the APx500 main Result panel.

Selecting a result

Many measurements have more than one result, represented as sub-branches beneath the Measurement. Click on the sub-branch and the result will open in the APx500 graph panel.

Adding Navigator elements

Add a Signal Path

To add a Signal Path, click on the Add Signal Path button at the top of the Navigator or the Add Signal Path branch at the bottom of the Project tree. You can also right-click on the Project branch and choose Add Signal Path from the context menu.

Add a measurement using a context menu

To add a measurement to a selected Signal Path, right-click in the Navigator area and then choose a measurement from one of the fly-out menus: Add Measurement, Insert Measurement Before Selection, or Insert Measurement After Selection.

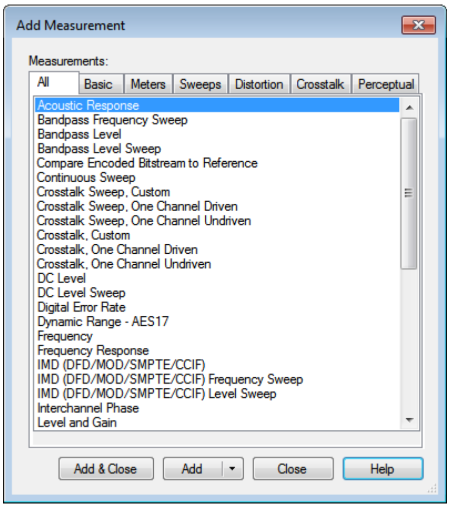
Add a measurement from the Main menu

You can add a measurement from the Main menu. Choose Project and then choose a measurement from one of the fly-out menus: Add Measurement, Insert Measurement Before Selection, or Insert Measurement After Selection.

The Add Measurement dialog

You can open the Add Measurement dialog by clicking the Add Measurement button at the top of the Navigator or the Add Measurement branch at the bottom of the Signal Path tree.

The tabs at the top of the dialog allow you to filter the list. Select one or more measurements and click Add and Close, or choose one of the items on the Add fly-out menu to insert the measurement(s) at a particular place in the signal path.



Typical Add Measurement dialog

Add a result

To add a result to a measurement, right-click on a result and choose Add Primary Result or Add Derived Result.

Copying and pasting Navigator elements

To copy a Signal Path or a Measurement, right-click on the branch you wish to copy and choose Copy from the context menu. To paste a Signal Path or a Measurement, right-click in the Signal Path you wish to copy into and choose Paste from the context menu.

Only one Signal Path Setup branch and Reference Levels branch can exist within each Signal Path. Signal Path Setup and Reference Levels cannot be copied or pasted.

Moving Navigator elements

You can change the order of Navigator elements by moving the branches within the tree. This can provide a more orderly project, but more importantly it also changes the order of operations within an automated sequence.

To move a Signal Path, select the branch you wish to move and drag it up or down to a new location in the Project.

To move a Measurement, select the branch you wish to move and drag it up or down to a new location within the Signal Path. Measurements cannot be dragged to a different Signal Path. Instead, use the Copy, Paste and Delete functions.

The Signal Path Setup branch and Reference Levels branch must be at the top of *each Signal Path tree. Signal Path Setup and Reference Levels cannot be moved.*

Renaming Navigator elements

To rename a Signal Path or a Measurement, click twice on the branch you wish to rename, or right-click on the branch and choose **Rename** from the context menu. The branch name will become available for keyboard editing (indicated by highlighting and a cursor bar).

Deleting Navigator elements

To delete a Signal Path, select the Signal Path branch you wish to remove and click on the **Delete Selected Item** button at the top of the Navigator. You can also right-click on the Signal Path branch you wish to remove and choose **Delete Signal Path** from the context menu.

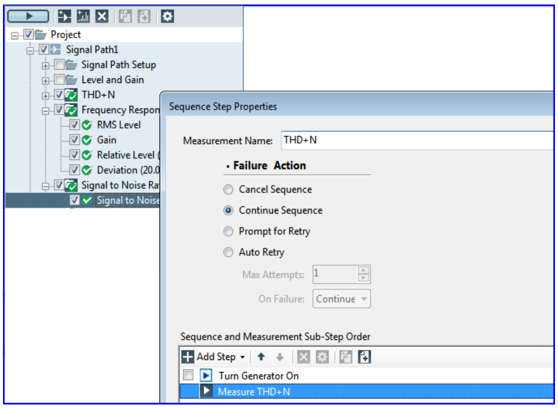
To delete a Measurement, select the Measurement branch you wish to remove and click on the **Delete Selected Item** button at the top of the Navigator. You can also right-click on the Measurement branch you wish to remove and choose **Delete Measurement** from the context menu.

There must be one Signal Path Setup branch and one Reference Levels branch in each Signal Path. Signal Path Setup and Reference Levels cannot be deleted.

Starting a measurement

To run a Measurement, right-click on the branch you wish to run and choose **Start Selected Measurement** from the context menu. This action runs the only the selected measurement and produces a Report for that measurement's results.

The Sequencer (Sequence Mode only)



The Sequencer is an easy-to-use automation feature accessed through the Navigator. Using the mouse, set a green check mark in the measurements you would like to run in a sequence. Click the **Start Sequence** button, and all checked measurements will be performed against limits (if set), and a report will be generated. Prompts, delays and external routines can be added to the sequence. Pass/Fail results are shown in the Navigator and in the Report.

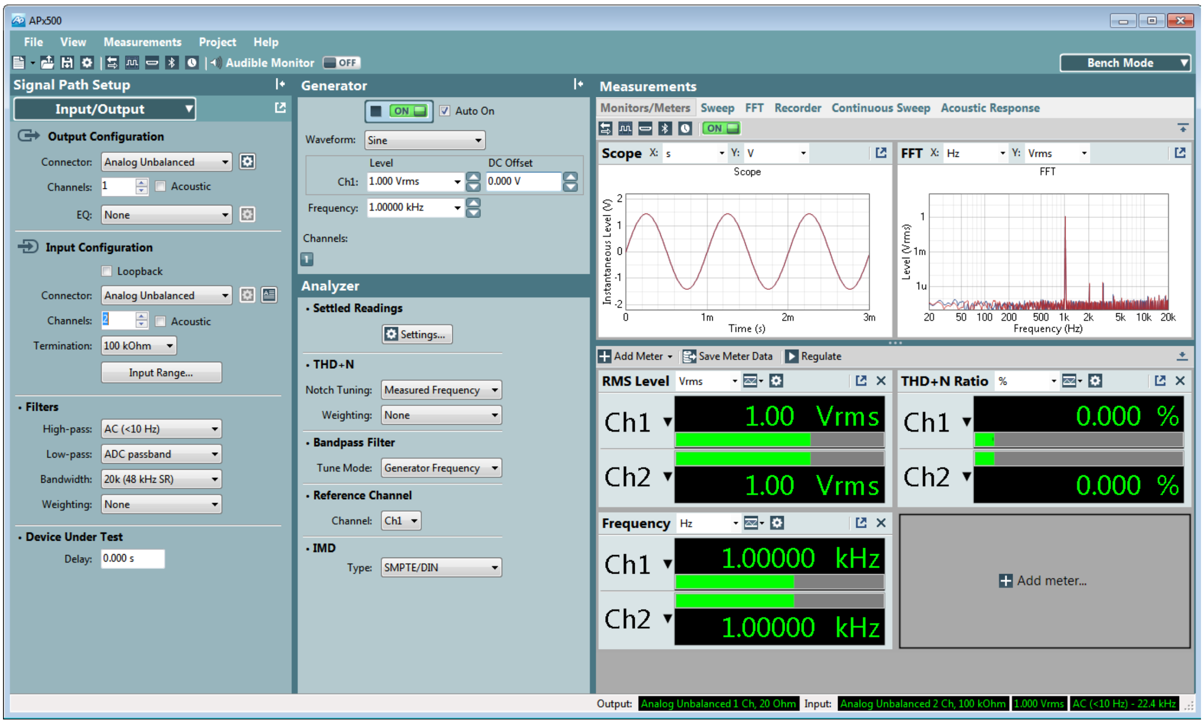
See Chapter 80 for more information about using the Sequencer to automate your measurements.

Starting a Sequence

You can select measurements in the Navigator to run in an automated sequence using the APx500 Sequencer, a flexible, powerful and easy-to-use automation tool.

The Bench Mode Workspace

BENCH MODE WORKSPACE



SIGNAL PATH SETUP

GENERATOR AND ANALYZER

MONITORS / METERS AND MEASUREMENT TABS

Bench Mode is an alternative user interface introduced in 2014. Unlike Sequence Mode, Bench Mode does not offer defined measurements. Instead, Bench Mode provides a set of tools that can be assembled and used in many different ways. AP2700 and APWIN users will recognize similarities between the Bench Mode paradigm and earlier Audio Precision control software.

The Bench Mode Workspace consists of three columns. The column to the left contains a collection of Signal Path Setup panels, very similar to the Signal Path Setup panels in Sequence Mode. The central column contains Generator and Analyzer settings. The right panel provides a selection of Tools displays, with a choice of meter, graph, sweep and other analysis tools.

Bench Mode measurement tools include a number of bar graph/numeric meters that can each be set to one of over 12 signal parameters, real time monitors, and an extremely flexible set of sweep engines for stepped sweep, FFT, time recorder, continuous sweep and acoustic response sweeps. X and Y axes can be set to a broad range of parameters, and nested sweeps are supported.

Bench Mode shares a number of features with Sequence Mode, including the Menu Bar, the Toolbar across the top of the workspace, and the Status Bar across the bottom.

The Signal Path Setup menus are essentially the same as in Sequence Mode, and the Scope and FFT Spectrum Monitors are identical. A toolbar in the Monitors/Meters tabbed view provides access to the same tools that are available in Sequence Mode.

The Signal Path Setup menus

The left panel is Signal Path Setup, and the menu at the top of the panel provides configuration settings for the Bench Mode signal path. Depending upon the connected APx instrument hardware, these menus may include

- Input/Output
- References
- Output Switchers
- Input Switchers
- DCX
- Clocks
- Triggers

The Signal Path Setup menus are also available in Sequence Mode.

Signal Path Setup is discussed in detail in Chapter 6.

The Generator and Analyzer

The Generator and Analyzer in Bench Mode are independent of each other, and to some extent independent of the Bench Mode measurements. The Generator, for example, can be turned On or Off, or have settings made or changed at any time. A measurement sweep can take control of the generator for the sweep duration, and then returns the generator to its previous state. Read more about the Bench Mode Generator in Chapter 82.

The Bench Mode Analyzer provides a number of analysis tools that you can customize for your test. These include settling, filtering and IMD analysis settings. Read more about the Bench Mode Analyzer in Chapter 82.

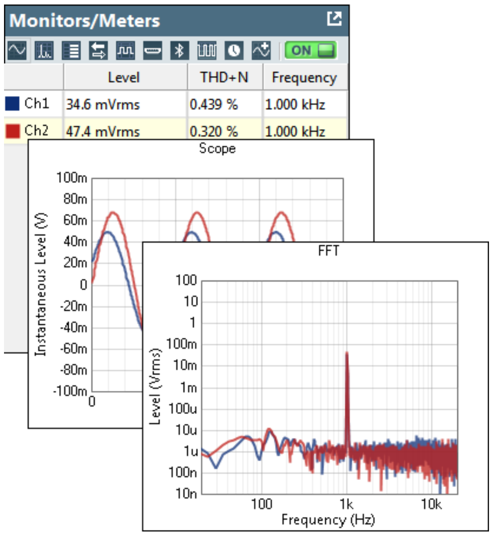
The Bench Mode Measurements

Bench Mode provides tabbed measurement panels, each displaying different measurement tools, including Monitors and Meters, a stepped Sweep, an FFT Scope and Spectrum analyzer, a time Recorder, and Continuous Sweep and Acoustic Response suites. Chapter 82, Bench Mode Overview, covers general topics regarding making Bench Mode measurements. Each Bench Mode measurement tool is discussed in detail in Chapters 83 through 88.

Common Features to Both Sequence Mode and Bench Mode

Some APx features and tools are common to both Sequence Mode and Bench Mode, including many Menus and Menu items, Project Properties, the Monitors and the Monitors/Meters tool bar, the Status Bar, the Selector, and the Audible Signal Monitor. Sweep features like importing sweep tables or nesting sweeps are the same, and graph features such as cursors or limits operate in the same way.

The Monitors/Meters



APx500 has a number of Monitors and Meters (depending upon hardware and software options) to provide continuous detailed information about the signals at the analyzer's inputs, and to monitor and set other features, all without changing the primary measurement view. Read more about the Monitors and Meters in Chapter 4.

In Sequence Mode, these are normally docked just below the Navigator, but each signal monitor can be undocked and moved around the screen.

In Bench Mode, the Scope and FFT Spectrum Monitors are at the top of the Monitors/Meters panel, and the other Monitors are available from the Monitors/ Meters toolbar.

When turned ON, the signal monitors operate continuously. The displays are refreshed many times per second, creating near real-time signal monitors.

The signal monitors can be turned OFF, which improves computer performance by reducing the CPU usage.

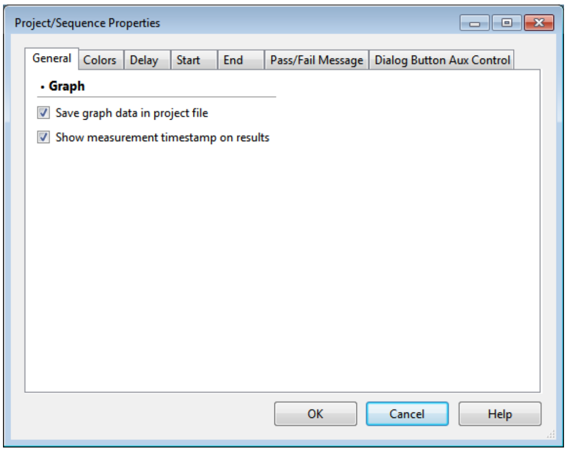
Use the Monitors/Meters toolbar to select the monitor view.

Choose

-  Scope Monitor
-  Meters Monitor
-  FFT Spectrum (disabled for APx515 unless software option SW-AML is installed.)
-  Aux Control

- Metadata Monitor: Status Bits/User Bits
- Metadata Monitor: HDMI (available for instruments fitted with HDMI I/O)
- Bluetooth Monitor (available for instruments fitted with Bluetooth I/O)
- Metadata: Status Bits/User Bits (for instruments with a digital interface only).
- Metadata: HDMI (for instruments fitted with the HDMI Option).
- PDM Control Codes Monitor (for instruments fitted with the PDM Option.
- Digital Serial Reference Clock (for instruments fitted with the Digital Serial I/O Option.
- Analog Sine Generator (for the APx555)

Project and Project/Sequence Properties settings



Sequence Mode

In Sequence Mode, the Project/Sequence Properties are global, project-wide settings. These include enabling saving of graph data, time-stamping results and graph and trace colors.

Project properties settings share a dialog with sequence properties settings. Access the dialog from the Main menu by selecting **Project > Project/ Sequence Properties**, or by choosing **Project/ Sequence Properties** from the **Project** context menu (right click on the Navigator **Project** node). You can also access this dialog by clicking **Project Colors...** in the **Channel Labels** dialog. See page 22.

The Project/Sequence Properties dialog contains several tabbed pages. The **General** and **Colors** tabs offer global, project-wide settings, and are discussed below. The remaining tabs offer sequence settings, and are discussed beginning on page 480.

Bench Mode

Bench Mode has a reduced set of Project Properties settings, with only the **General** and **Colors** tabs available.

General tab

These properties are project-wide (global) settings. They affect all measurements and signal paths within the project.

Save graph data in project file

By default, this checkbox is checked, and the graph data for all results in the project is saved in the project file.

Graph data that requires recalculation is not saved. For example, when saving **Relative Level** data, only the data in memory when saving (**Normalized** or **Centered**) will be saved. When a project file that contains result graph data is opened, all graph data that do not require recalculation are displayed in their various measurement results.

Show measurement timestamp on results

Measurement data received into the project, whether as Measured data or Imported data, is stamped with the time and date it entered the project. When this checkbox is checked the measurement timestamp is displayed on the graph panel and in the report.

Colors tab

These properties are project-wide (global) settings, but can be overridden locally for any result. For local bar graph (meter) color overrides, see page 564. For local XY graph color and line style overrides, see page 572.

Colors for bar graph (meter) displays and colors, line style and line width for XY graph traces can be customized from a wide palette, and automatic cycling of colors and styles can be specified for appended and imported measurements.

Graph Colors and Styles

Rows

Each row in this grid applies to an input channel, to a maximum of 16 channels.

Columns

Each column in this grid shows color, line style and line width for one **Color Cycle**. **Color Cycle** columns can be added to the grid, to a maximum of 16.

Color Cycle behavior

For measurements that support **Append Graph Data** and **Import Graph Data**, each **Color Cycle** column applies to a **Data Set**. Within a measurement result, these **Color Cycles** are applied to **Data Sets** in the order the **Data Sets** are added to the result. If the last defined **Color Cycle** has been applied, the next **Data Set** added will have **Cycle 1** applied, and further **Data Set** additions will progress through the **Cycle** columns.

These features allow you to specify colors and styles that can show channels and appended data in a clear and understandable display.

Grid Toolbar

Add Cycle button

Add a **Cycle** column to the grid.

Duplicate Selected Cycle button

Copies the currently selected **Cycle** column and adds it to the grid.

Move Cycle right or left button

Moves a **Cycle** column. This affects the order of the cycles applied to appended and imported data.

Automatically choose new colors button

Changes the color of the selected cells by an arbitrary amount. A quick way of adding different colors to the **Cycle** grid.

Set Color button

Allows selection of a custom color for the selected cells, using the color picker dialog.

Set Line Style button

Allows selection of one of five line styles for the selected cells.

Set Line Width button

Allows selection of one of five line widths for the selected cells.

Import button

Opens an import file browser to select and import a **Trace Style Cycle** file (*.tsc). The styles in the file will overwrite the styles currently in the grid.

Export button

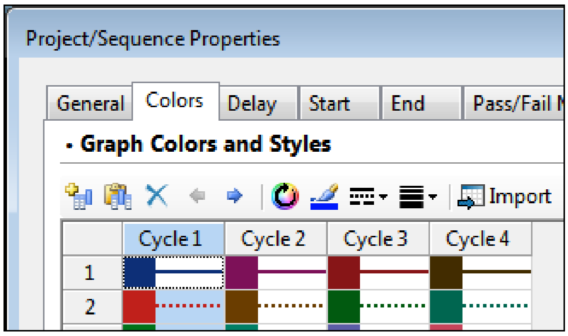
Opens an export file browser to select and export the current **Cycle** grid contents to a **Trace Style Cycle** file (*.tsc).

Reset to Default button

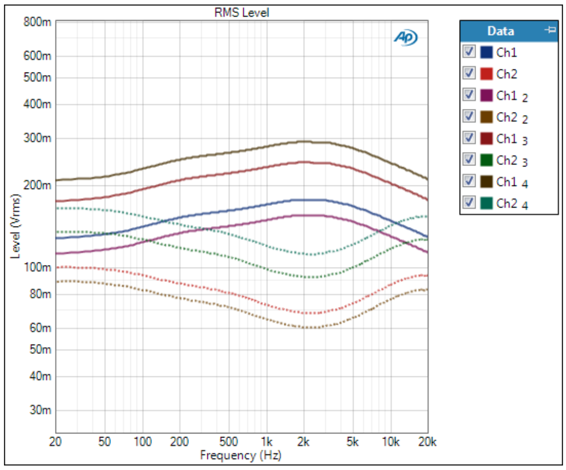
Resets the highlighted cells to the default colors.

Example

For example, a user might want to display noise vs. time for 2 channels, across 4 appended acquisitions. The illustrations show a set of Color Cycles that would provide a bold, solid line for Channel 1 and a bold, dotted line for channel 2. The appended acquisitions would cycle through the four colors.



Color Cycles example



The resultant Noise Recorder graph

The Menu bar

The Menu bar extends across the top of the workspace and provide access to the following menus.

The File menu

New Project (Sequence and Bench Modes)

Opens a new project file. New projects always use a template, either the Standard APx500 Project or another template of your choice. Available templates are shown in the New Project dialog.

Open Project (Sequence Mode)

Opens an existing project file from disk. You will be prompted to save the current project.

Lock Project (Sequence Mode)

Allows you to set a password to lock the current project. This function is useful to prevent accidental project changes by a production operator. When locked, all user control of signal path, measurement and sequence settings is disabled. A sequence can still be run, and display and navigation features are available.

Unlock Project (Sequence Mode)

Allows you to unlock a project by entering the correct password.

Save Project (Sequence and Bench Modes)

Saves the current project file. If this is the first time the project is being saved, you will be prompted for a project filename.

Save Project as (Sequence and Bench Modes)

Saves the current project under a new filename.

Save Project as Template (Sequence and Bench Modes)

You can make a new project template by saving your current project as a template file. In the dialog, name your template file and write an optional description.

> *You can create project template files to help manage and streamline your measurement work. A project template file contains all project definitions and settings.*

Manage Project Templates (Sequence and Bench Modes)

Opens a dialog in which you can rename or delete a project template. You can also make a template the default project.

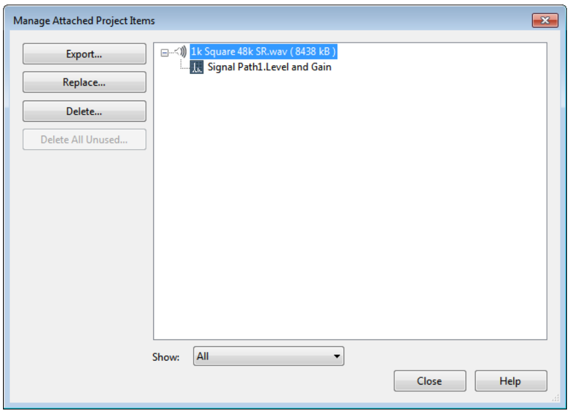
You can add, rename or delete project groups from the Edit Groups dialog. You can move a project template from one group to another using the Move to Group dialog.

Templates are saved under My Documents > APx500 Templates, where the subfolders are managed as Template Groups. Choose or create a Templates Group from the Group Name field.

Manage Attached Project Items (Sequence and Bench Modes)

Image files, generator waveform files, Microsoft Word report layouts, and multitone signal definition files can be attached to an APx500 project for use in a measurement or a user prompt. Attached items that have been removed from current use remain attached to the project, increasing the project's memory and disk use.

The Manage Attached Project Items dialog allows you to view all the current image and generator waveform attachments, see where they are used, and manage their usage.



Also see Chapter 14 for more information about generator waveforms, Chapter 60 for multitone signal definition files, Chapter 81 for Microsoft Word report layouts, and page 501 for image files.

Manage Attached Project Items: Export

Export allows you to save an attached file to disk, outside of the APx project file. Custom report layouts, in particular, are created, edited and maintained within the project, and must be exported for use by other projects.

Manage Attached Project Items: Replace

Replace allows you to choose a new file for a particular project item. The new file replaces the current file in all its project uses.

Manage Attached Project Items: Delete

Delete allows you to remove a specific file from the project.

Manage Attached Project Items: Delete all unused

Delete all unused examines the current usage of all the files in the project. The files that are not currently being used are removed from the project.

Import Graph Data (Sequence and Bench Modes)

Imports graph data from a file on disk to be displayed in the current graph.

Export Graph Data (Sequence and Bench Modes)

Exports the graph result data as an Excel spreadsheet (*.xls), a comma separated value text file (*.csv) or a Matlab file (*.mat).

Operating Mode (Sequence and Bench Modes)

Choose Sequence Mode or Bench Mode.

Recent Projects (Sequence and Bench Modes)

Displays a list of recently opened project. Select a project to open.

Exit (Sequence and Bench Modes)

Closes the application.

The View menu

Scope Monitor (Sequence Mode)

Switches the Monitor to the Scope view.

FFT Spectrum Monitor (Sequence Mode)

Switches the Monitor to the FFT Spectrum view.

Meters Monitor (Sequence Mode)

Switches the Monitor to the Meters view.

Aux Control (Sequence and Bench Modes)

Opens the Aux Control Monitor view.

Metadata Monitor: Status Bits/User Bits (Sequence and Bench Modes)

Switches the Monitor to the Status Bits/User Bits view. This selection is not available if the instrument not fitted with DIO or HDMI+ARC.

HDMI: Metadata and CEC (Sequence and Bench Modes)

Switches the Monitor to the HDMI metadata/CEC view. This choice is only available when the instrument is fitted with the HDMI or HDMI+ARC option installed.

Bluetooth Monitor (Sequence and Bench Modes)

Switches the Monitor to the Bluetooth view, if the Bluetooth option module is installed.

Digital Serial Reference Clock (Sequence and Bench Modes)

Switches the Monitor to the Digital Serial Reference clock view. This choice is only available when the instrument is fitted with Digital Serial I/O, and Input Configuration is not DSIO.

Analog Sine Generator (Sequence and Bench Modes)

Switches the Monitor to the Analog Sine Generator View (APx555 only). This allows secondary analog sine output while primary generator output is digital.

Hide/Show Measurement Navigator (Sequence Mode)

Toggles the Navigator visibility.

Hide/Show Selector (Sequence Mode)

Toggles the Selector visibility.

Turn Monitors On/Off (Sequence and Bench Modes)

Toggles the Signal Monitors On or Off. When Signal Monitors are Off the measurement PC will show better performance.

Audible Signal Monitor (Sequence and Bench Modes)

Allows you to turn the Audible Signal Monitor On or Off, or to adjust settings.

The Measurements menu

Turn Generator On F9 / Turn Generator Off F12 (Sequence and Bench Modes)

If the current measurement is a meter measurement, Turn Generator On F9 and Turn Generator Off F12 appear on the Measurements menu. Click these, or press the associated keys (F9 or F12) to turn the Generator On or Off in the current measurement

Active Measurements List (Sequence and Bench Modes)

Displays a list of Active Measurements in the current Signal Path.

The Project menu

Add Signal Path (Sequence Mode)

Adds a signal path to the project

Add Measurement (Sequence Mode)

Choose a measurement to add from one of the fly-out menus: Add Measurement, Insert Measurement Before Selection, or Insert Measurement After Selection.

Add Primary Result (Sequence Mode)

Adds a primary result to the selected measurement.

Define New Result (Sequence Mode)

Chose X, Y or Bar from a submenu to add a user-defined results.

Add Derived Result (Sequence Mode)

Adds a Derived Result to the selected result.

Delete (Sequence Mode)

Delete items selected on submenu.

Start Sequence (Sequence Mode)

Starts entire sequence defined in the Navigator.

Start Selected Signal Path (Sequence Mode)

Starts a sequence for the selected Signal Path only.

Start Selected Measurement (Sequence Mode)

Starts the selected measurement only.

Start Sequence from Selected Measurement (Sequence Mode)

Starts a sequence beginning at the selected measurement.

Edit Report Properties (Sequence Mode)

Opens the Edit Report Properties dialog.

Edit Data Output Properties (Sequence Mode)

Opens the Edit Data Output Properties dialog.

Project/Sequence or Project Properties (Sequence and Bench Modes)

Displays the Project/Sequence or Project Properties dialog.

The Window menu

The View menu is not available in Bench Mode.

The Window menu is available from the main Menu bar. Measurements that have been visited are listed in the Window menu, with a check mark indicating the active measurement. Up to ten measurement are listed in the menu; if there are more than ten measurements in the Project, the ten that have been most recently active are listed here. Click the **Windows...** button to open the Windows dialog and view all the measurements.

The Windows dialog (Sequence Mode)

Open the Windows dialog by selecting **Windows...** from the bottom of the Window menu. All measurements in the Project are shown in the Windows dialog. The active measurement is highlighted in the dialog. To activate a different measurement, select the measurement and click **Activate**.

The Help menu

Help on this topic F1 (Sequence and Bench Modes)

Displays context-sensitive Help for the current measurement. Equivalent to pressing function key F1.

Contents (Sequence and Bench Modes)

Displays the Help Contents view.

Index (Sequence and Bench Modes)

Displays the Help Index view.

Register Product (Sequence and Bench Modes)

Register your product with the Audio Precision Web site.

About (Sequence and Bench Modes)

Provides detailed information about APx500 and the current instrument hardware and firmware.

The Toolbar

The Toolbar provides a convenient access to often-used commands.

- New Project / New Project from Template
- Open Project
- Save Project
- Project/Sequence Properties
- Scope Monitor
- FFT Spectrum Monitor
- Meters Monitor
- Aux Control
- Metadata Monitor: Status Bits/User Bits
- Metadata Monitor: HDMI (available for instruments fitted with HDMI I/O)
- Bluetooth Monitor (available for instruments fitted with Bluetooth I/O)
- Audible Monitor Configure Audible Signal Monitor
- OFF Audible Signal Monitor On/Off

The Project File

APx500 projects are stored in a project file with the filename extension *.approjx.

*.approj files created in earlier versions of APx500 are opened and converted to the current format.

A project file has one or more signal paths (for Sequence Mode), and contains all the project measurement information, including signal path setup, all settings for each measurement and graph along with limit information and sequence instructions. Generator waveforms, user prompt images and report layout files are also saved in the project file.

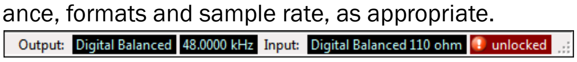
A co-worker with APx500 and a compatible Audio Precision APx500 series analyzer can open a project file, make the same connections to a DUT and perform the same set of tests.

The following data is NOT saved in an APx project file:

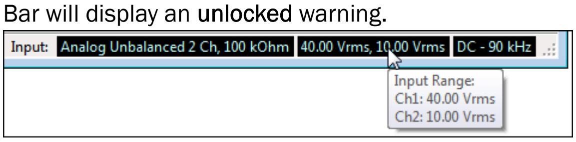
- Report results. Reports can be exported in a number of formats, but the result data is not included in a project file.
- Audio acquisition data.
- External programs called by a sequence step.

Status bar

The Status Bar (at the bottom of the APx500 window) displays the output and input connections, impedance, formats and sample rate, as appropriate.

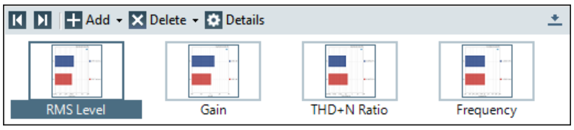


When the Input Configuration is set to a digital input and no signal is present, or if the signal is corrupted or out of range, the sample rate indicator in the Status Bar will display an **unlocked** warning.



When the Input Configuration is set to an analog input, the input range is displayed. If channels are set to different ranges, place the cursor over the range field and the tool tip will display the ranges set for the various channels.

The Selector



The Selector runs across the bottom of each measurement view panel, providing a visual means to browse and choose measurement views from thumbnail icons.

Read about the Bench Mode panels beginning on the next page.

Selecting a result

To select a result, click on the result thumbnail.

In addition, you can select an existing derived result by first selecting the source result and then choosing **Go to Derived** from the right-click context menu.

You can also select a source result by first selecting the derived result and then choosing **Go to Source** from right-click context menu.

Moving a result

In the Selector, grab a result thumbnail with the mouse cursor and drag it to a new position in the Selector. The result positions in the Selector determines the relative result locations in a report.

Adding or deleting a result

Select a result and click **Add** or **Delete** in the Selector toolbar.

In addition, you can add a derived result by right-clicking on the result thumbnail and choosing **Add Derived Result**. You can delete a result by right-clicking on the result thumbnail and choosing **Delete**.

Viewing result details

To view result details, select the result and click **Details** in the Selector toolbar.

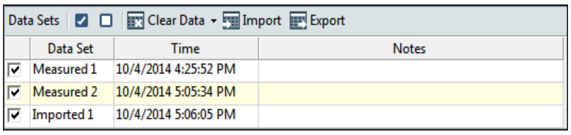
Derived Results

A derived result is an additional result, computed from data in one or more existing results. Examples include smoothing, calculating means, comparisons of data, etc. Read more about Derived Results in Chapter 95.

The Data Sets panel

A **Data Set** is created whenever a batch measurement is run. Real-time measurements do not produce a **Data Set**. A **Data Set** is the set of results measured from an acquisition, or imported from a data file.

Data Sets are managed in the **Data Sets** panel, located below the **Selector** in the APx workspace.



See Managing Data Sets on page 573.

The Audible Signal Monitor

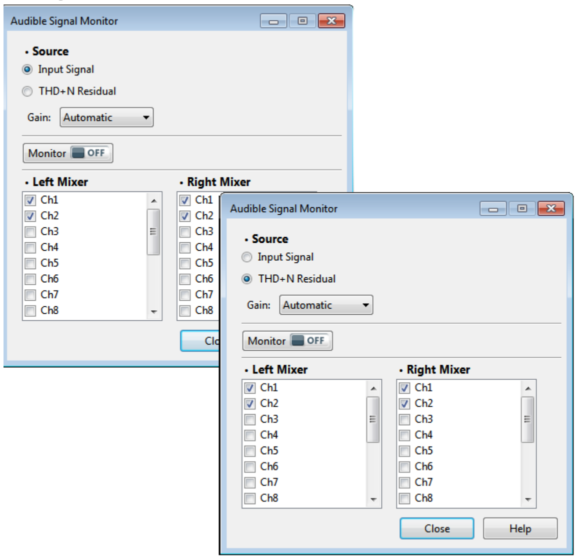
File input signals cannot be monitored using the Audible Signal Monitor.

You can listen to the signal applied to the APx500 analyzer inputs by using the Audible Signal Monitor. For some signals and measurements, you can choose to listen to the input signal's THD+N residual.

The Audible Signal Monitor uses the attached PC's software and hardware sound components to play the audio. You must have a PC sound card or an equivalent device, and you must connect headphones, speakers or an audio amplifier to the sound card outputs.

Using the Audible Signal Monitor

Click the Audible Signal Monitor Settings button on the toolbar, or choose **View > Audible Signal Monitor > Settings...** to open the Audible Signal Monitor Settings dialog.



Source

Input Signal

When **Input Signal** is selected, the audio at the system inputs chosen in Signal Path Setup is available for monitoring, subject to the settings made in the Left and Right Mixer panels, below.

THD+N Residual

The **THD+N Residual** setting is only useful when a steady sine signal is present at the system inputs. This setting applies a very deep, narrow notch at the stimulus signal frequency, effectively removing the stimulus. The residual consists of distortion products, hum and noise.

Gain

The APx500 system can measure a very wide range of signal levels. When monitoring the input signals, some signals may be very loud, some very soft. You can choose gain settings here.

- **Automatic** (the default) When **Automatic** is checked, the software automatically adjusts the Audible Signal Monitor volume for a constant level.
- x1 (0 dB gain)
- x10 (20 dB gain)
- x100 (40 dB gain)
- x1k (60 dB gain)
- x10k (80 dB gain)
- x100k (100 dB gain)

Monitor ON / OFF

The OFF ON button toggles the Audible Signal Monitor **On** and **Off**. This button is also available on the toolbar.

Left Mixer / Right Mixer

When **Source** is set to **Input Signal**, the **Left Mixer** and **Right Mixer** panels are available.

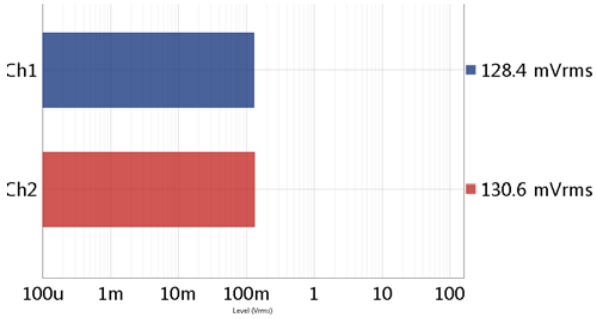
All input channels checked in the Left Mixer panel will be summed and routed to the Left PC sound output. Unchecked channels will not be sent to this output.

All input channels checked in the Right Mixer panel will be summed and routed to the Right PC sound output. Unchecked channels will not be sent to this output.

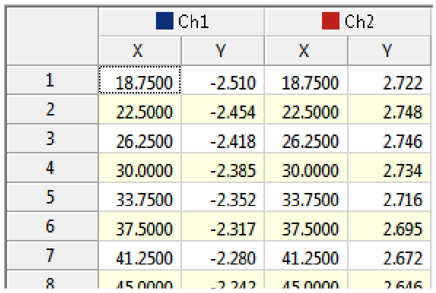
THD+N Monitor Channel

When **Source** is set to **THD+N Residual**, the THD+N Monitor Channel selector is available. Choose the input channel to monitor for THD+N Residual.

Measurement Result Displays

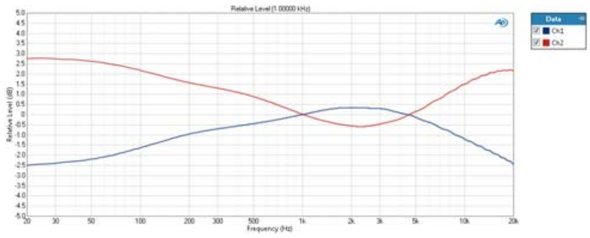


Meter bar single value result

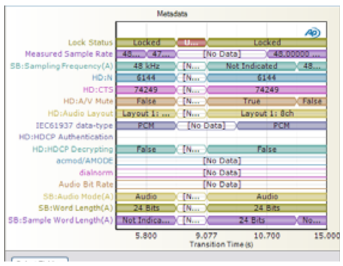


	Ch1 X	Ch1 Y	Ch2 X	Ch2 Y
1	18.7500	-2.510	18.7500	2.722
2	22.5000	-2.454	22.5000	2.748
3	26.2500	-2.418	26.2500	2.746
4	30.0000	-2.385	30.0000	2.734
5	33.7500	-2.352	33.7500	2.716
6	37.5000	-2.317	37.5000	2.695
7	41.2500	-2.280	41.2500	2.672
8	45.0000	-2.242	45.0000	2.646

Tabular result view



XY graph result



"Logic analyzer" result

APx500 has many measurement result views using different methods and display techniques. Read more about Results in Chapter 93.

Meter Bar Displays

Meter bar displays are used for those measurements that measure only one parameter per channel, providing a clear view of this parameter across many channels. APx500 also uses a meter bar display for distortion product measurements, where several parameters are displayed but the view is limited to one channel.

XY Graphs

APx500 graphs display two parameters of a signal measurement on an XY grid. In a frequency domain display, these parameters are typically level versus frequency, with level on the Y axis (vertical, left), and frequency on the X axis (horizontal, bottom). In a time domain display, these parameters are typically level versus time, with level on the Y axis and time on the X axis. Other relationships can be graphed: phase angle versus frequency, DUT output level versus generator level (linearity) and so on.

A second Y axis can be added to show three parameters simultaneously.

"Logic analyzer" result

The Metadata Recorder logs metadata status changes against time, shown in a logic analyzer type result.

Tabular Displays

Meter and graph measurement values are available as tabular displays in the meter or graph Data Grid.

Undocking Views

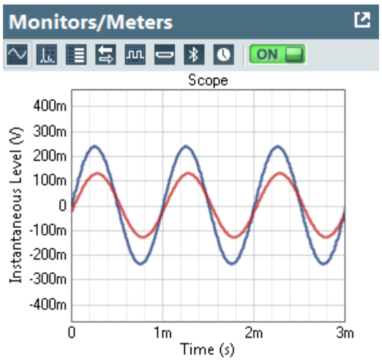
One or more measurement views can be undocked to compare readings. Click the Undock button on a view, and then select another view to Undock. Undocked views, shown here, can be independently moved and resized, revealing all graph settings for independent adjustment.

4

Monitors/Meters

Introduction

Scope Monitor



The Scope Signal Monitor provides an oscilloscope view on an XY graph.

Settings

The additional settings shown here, and graph display options, are available from a right-click context menu or from the graph panel when **Undocked**.

Interpolation

For time-domain displays, the APx500 graphs are normally plotted using sinc function interpolation. Interpolation adds points to the displayed trace that do not exist in the measured data, to make data trends more easily visualized. The default in APx is to have Interpolation **On**. However, digital domain signals are sometimes best understood when viewing actual samples, with interpolation switched **Off**. Also see "Interpolation and limit failure markers" on page 581.

Residual Display

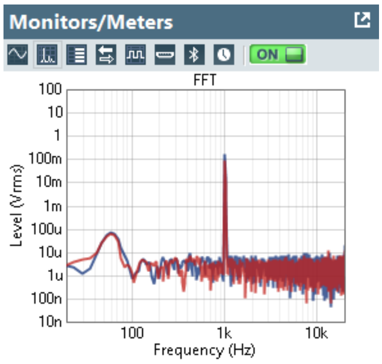
- **Off**
  When Residual Display is **Off** (the default), a time-domain view of the full input signal is displayed.
- **x1 through x100K**
  These selections display a time-domain view of the THD+N residuals after the fundamental is removed, with selectable gain applied. This view requires a periodic signal for meaningful results.

Show Cycles

For periodic waveforms, you can choose to display 1 to 10 cycles of the waveform. 3 cycles is the default.

For non-periodic waveforms, the X-axis is set to 200 ms. When Graph Properties X-Axis is not set to Auto, **Show Cycles** is not available.

FFT Spectrum Monitor



This monitor view is disabled for APx515 unless software option SW-AML is installed.

The FFT Spectrum Signal Monitor displays a frequency-domain spectrum view on an XY graph. Right-click on the monitor to access graph properties.

Settings

Graph display options and these additional settings are available from a right-click context menu, or from the graph panel when **Undocked**.

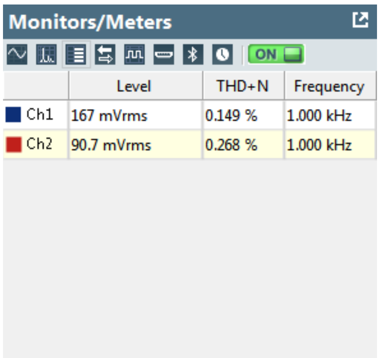
- **Show Residual**
  Adds distortion residual FFT display for each channel.
- **Averages**
  For non-periodic waveforms such as noise, averaging multiple acquisitions can provide a more useful view. The FFT monitor will display the average of the last *n* acquisitions. Default is 1 (no averaging); maximum is 1000. **Reset** clears the averag-

ing history and restarts averaging with the next acquisition.

- Window
  FFT acquisitions must either be synchronous or have one of a number of amplitude windows applied to provide useful data for interpretation. Each window function brings advantages and disadvantages. The default selection, AP-Equiripple, is a proprietary Audio Precision FFT window that is an excellent choice for most FFT measurements.
- FFT Length
  Set the FFT record length here. Options from 256 K samples to 1.2 M samples are available; default is 8 K samples.

Go to page 442 for more information about FFTs and FFT windows.

Meters Monitor



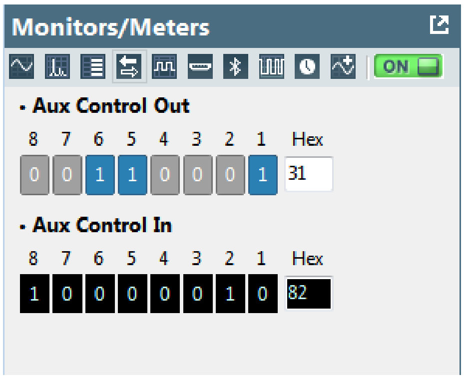
Meters

The Meters Signal Monitor displays a grid with Level, THD+N and Frequency measurements for each channel. The number of channels displayed is set in the Input Configuration settings in the Signal Path Setup view.

Changing Units for the Meters monitor

Right-click on the Meters monitor display to open a context menu, where you can select the units for each of the meter results.

Aux Control Monitor



Reading Aux Control Out and In bit states

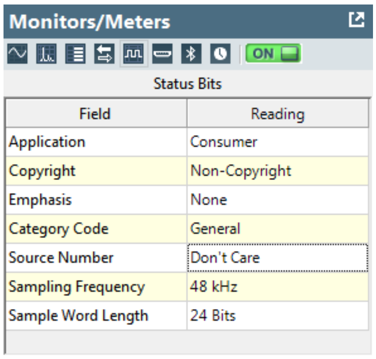
The Aux Control Monitor displays the current state of the Aux Control Out and Aux Control In bits. Bit states are shown both in a binary display and as the hex equivalent.

Setting Aux Control Out bit states

Additionally, Aux Control Out bits can be set from this monitor window, either by clicking on the binary display to toggle the state of a bit, or by entering the hex equivalent in the **Hex:** field.

For more information about Aux Control go to Chapter 96.

Metadata Monitor: Status Bits/User Bits



This monitor is available only for instruments with Digital Input/Output or HDMI.

The Metadata Monitor: Status Bits displays the channel status bits embedded as metadata in the digital input signal. Status bits are supported in the transport streams for the Digital Balanced, Digital Unbalanced, Digital Optical, HDMI ARC Rx and HDMI Sink inputs.

The Digital Serial Input/Output (DSIO) protocol does not support embedded metadata.

Consumer status bits

For consumer application status bits, the docked Monitor view displays this set of status bit fields:

- Application
- Copyright
- Emphasis
- Category Code
- Source Number
- Sampling Frequency
- Sample Word Length

Undock the Monitor to display all available consumer application status bits fields, in both plain text and in hex.

Professional status bits

For professional application status bits, the docked Monitor view displays this set of status bit fields:

- Application
- Emphasis
- Sampling Frequency

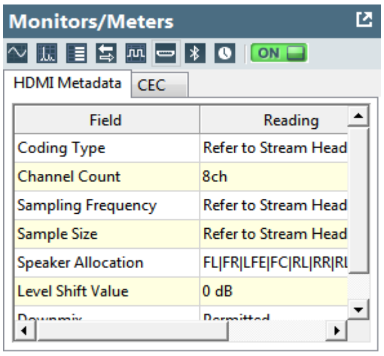
Undock the Monitor to display all available professional application status bits fields, in both plain text and in hex.

User bits

In the undocked Metadata Monitor views, user bits are displayed in hex. User bits are the same for both consumer and professional applications.

Go to page 354 for more information about Status Bits and User Bits.

HD Monitor: Metadata tab



This monitor is available only for instruments fitted with the HDMI Option.

This topic discusses *reading* the Audio InfoFrame data on a received HDMI transport stream in the HDMI Monitor.

For information about *setting* the Audio InfoFrame data to be embedded in the transmitted HDMI transport stream, see page 113.

For HDMI input signals, the docked Monitor view displays these Audio InfoFrame fields:

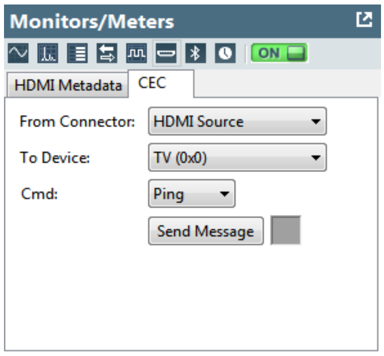
- Coding Type
- Channel Count
- Sampling Frequency
- Sample Size
- Speaker Allocation
- Level Shift Value
- Downmix

Undock the Monitor to display the Audio InfoFrame fields in both plain text and hex, plus these additional HDMI metadata fields:

- N
- CTS
- A/V Mute
- High Bit Rate
- Audio Layout
- HDCP Decrypting

Go to page 356 for more information about HDMI Audio InfoFrame metadata.

HD Monitor: CEC tab



This tab is available if the HDMI+ARC Option (+ CEC enabled) is installed. See More about CEC on page 120.

The CEC tab provides settings and controls to send CEC commands via either the sync or source connectors, and to receive replies acknowledging receipt.

From Connector:

Select the HDMI connector (Source, Sync, ARC Tx or ARC Rx) that will send the command and receive the acknowledgement.

To Device:

Select the device to be addressed. The menu lists the 16 defined CEC logical addresses.

Cmd:

Select the command to send, **Ping** or **Custom**. A **Ping** is a specific CEC polling message that any addressed device should acknowledge.

When **Custom** is selected, the **Opcode** and **Operands** fields are available to add payload information to the message.

Opcode:

Enter an arbitrary opcode (operational code) here.

Operands (hex):

Enter arbitrary operands here, in hex.

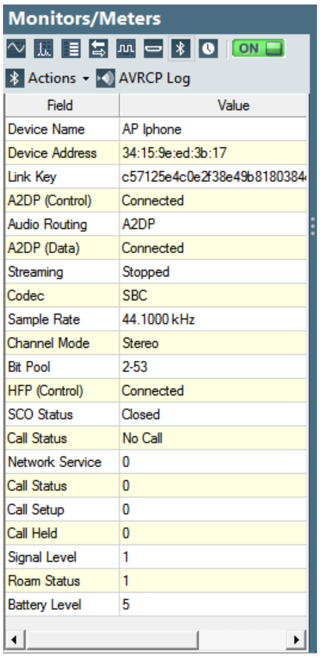
Send Message:

Click this button to sent the message.

The field to the right will remain grayed out until the message is acknowledged (ACK) by the device, then will display a passed graphic. If the message fails, a failed graphic will be displayed.

Note: ARC Tx and ARC Rx can also send certain ARC-specific CEC commands. See CEC for ARC on page 119.

Bluetooth Monitor



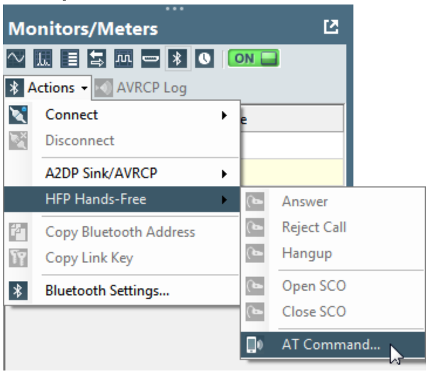
This monitor is available if the Bluetooth Option is installed and active.

The Bluetooth Monitor displays a number of Bluetooth status fields and their current values. Additionally, Bluetooth actions, settings and utility functions are available through a context menu (right-click in the Bluetooth Monitor display.)

Bluetooth Status display

Fields shown are dependent upon profile, connected device and other variables. Here is an example:

Bluetooth actions, settings and utility functions (from monitor context menu)



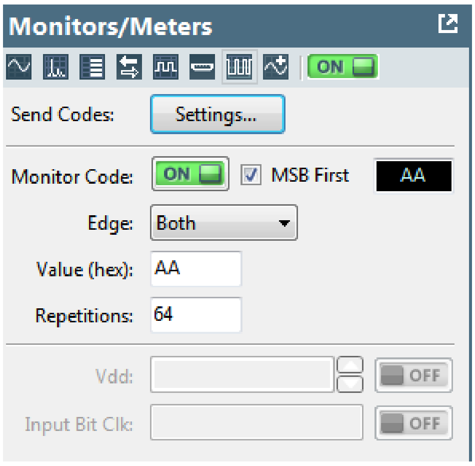
Right-click on the Bluetooth monitor display to open the context menu. Options include

- Connect.
- Disconnect.
- [One or more Actions, depending upon active profile].
- Copy Bluetooth Address.
- Copy Link Key.
- Open Bluetooth Settings dialog.

AVRCP Log

If APx is configured with Bluetooth profile that includes AVRCP Target, the AVRCP Log feature will record AVRCP activity. Click on AVRCP Log in the Bluetooth Monitor panel to open the Received AVRCP Command Log panel. You can view and save the logged commands.

PDM Monitor



See more about PDM on page 139.

This monitor is available if the PDM Option is installed and active.

The PDM Monitor enables monitoring and sending amplifier control codes embedded in a PDM bit-stream.

Send Codes:

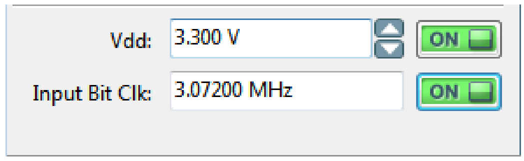
- Settings...
  Click this button to open the Send PDM Control Codes dialog. See page 36.

Monitor Code: ON/Off

When Monitor Code is ON, APx500 monitors the incoming PDM bitstream, looking for a control code that matches the criteria set in the next 4 fields. When a matching code appears, a confirmation field appears in the reading field here and in the Status Bar, displaying the hex value of the desired code. Read more about Control Codes on page 141.

- MSB First
  When MSB First is checked, the expected control code must be formatted with MSB first. When MSB First is not checked, the desired control code must be formatted with MSB last.
- Edge
  Select the data edge to monitor.
- Value (hex)
  Enter the expected code value in hex.
- Repetitions
  Enter the expected number of repetitions of the control code.

Vdd and Input Bit Clock



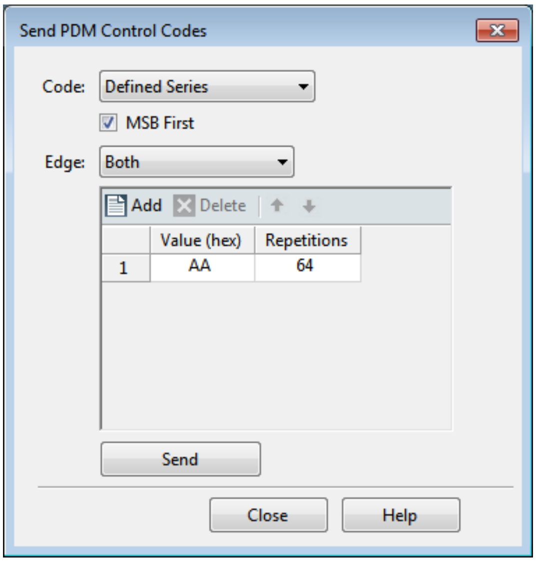
The Vdd feature on this panel is only available when the instrument is fitted with a PDM Option module, and when neither Output Configuration nor Input Configuration is set to PDM.

The Input Bit Clock feature is only available when the instrument is fitted with a PDM Option module, and when Input Configuration is not set to PDM.

These features enable power and clock to be supplied to a DUT such as a MEMS microphone or an integrated circuit, while generating and/or analyzing audio in a format other than PDM.

- Vdd:
  Set the voltage required for operating power the DUT here. This is available on the PDM module at the Vdd Supply BNC connector, providing DC current up to 15 mA, with a voltage range of +0.8 VDC to +3.60 VDC.
- ON/OFF
  This switch turns the Vdd DC power supply On or Off.
- Input Bit Clk
  Set the bit clock rate required by the DUT here. Bit clock is available on the PDM module at the Input Bit Clock BNC connector. The range is 128 kHz to 24.576 MHz.
- ON/OFF
  This switch turns the Input Bit Clock On or Off.

Send PDM Control Codes dialog



The PDM transmitter can embed amplifier control codes in the PDM bitstream. Click the **Send Codes: Settings** button in the PDM Monitor panel to open the Send PDM Control Codes dialog.

Also see the **Control Codes** discussion in More About PDM on page 141.

This dialog enables configuration for transmitting PDM control codes, and provides a **Send** button to insert the codes into the PDM bit stream in real time.

Code

Choose

- **Defined Series** to send one code value for a defined number of repetitions, or to send a series of different code values. Enter the code values and the number of repetitions in the grid below. Add more rows to the grid to specify other codes.

or

- **Single Code Indefinitely** to repeat one code value indefinitely, until **Stop** is invoked. Enter the code in the **Value** field.

MSB First

Check **MSB First** to send the Control Code MSB (Most Significant Bit) first. When **MSB First** is not checked, the MSB is sent last, and the LSB first.

Edge

Choose **Both**, **Rising** or **Falling** to select the PDM bitstream edge to carry the Control Code data.

Value (hex)

Enter the control code value as a 2 digit hex number.

Repetitions

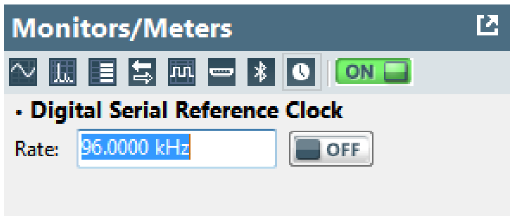
For **Defined Series**, set the number of repetitions here.

Send/Stop

Click the **Send** button to insert the control code(s) into the PDM bit stream. If **Code** is set to **Single Code Indefinitely** when **Send** is invoked, the Send button becomes a Stop button.

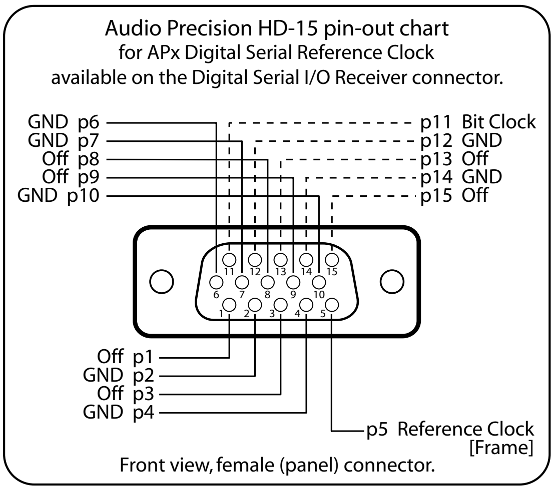
> *Note: PDM Control Codes can be configured and sent during an automated sequence. See Sequencer: Add a Send PDM Control Codes step on page 490 and Sequencer: Add a Send PDM Control Code Indefinitely step on page 490.*

Monitor: Digital Serial Reference Clock



If you have a Digital Serial I/O module installed, you can output a reference clock signal from the module in any configuration where the signal path is not Digital Serial Input. The rate is arbitrary, unrelated to the internal converter rates or other digital I/O rates.

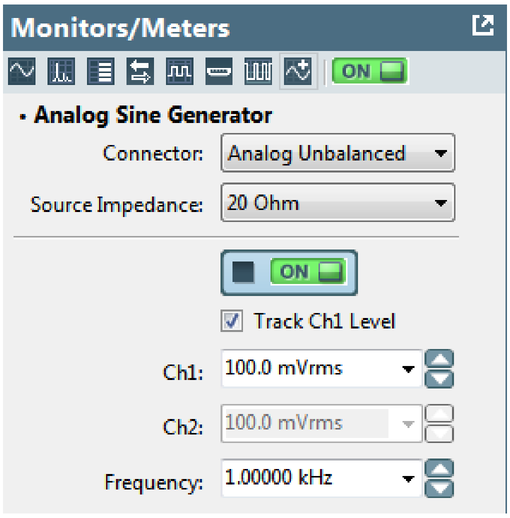
The signal is a 50% duty cycle square wave of 3.3 Vpp with a range of 4 kHz to 216 kHz. Accuracy is ±0.0003% (3 PPM). The output impedance is 50 Ω.



The clock signal appears on pin 5 of the Digital Serial I/O Receiver HD15 jack. Pins 2, 4, 6, 7, 10, 12 and 14 are grounded. Pins 1, 3, 8, 9, 13, 15 are Off. Bit Clock (64x) appears on pin 11.

If you have an Audio Precision CAB-DSIO cable kit, use the “FRAME” cable, colored green. This connects to pin 5 and to a grounded pin.

Analog Sine Generator



This feature is only supported by the APx555.

The APx555 has two generators, the standard DSP/ DAC generator used across the APx family, and the analog High Performance Sine Generator.

When the APx555 analog outputs are not selected, it is possible to use the High Performance Sine Generator as a second generator. In this case you could use the DAC generator to feed a digital output, while simultaneously using the High Performance Sine Generator to feed a different signal to the analog outputs.

In both Bench Mode and Sequence Mode, you can open a panel to control the Analog Sine Wave Generator feature from the Menu Bar or the Monitors/Meters area.

Choose the desired analog output connector, the source impedance, the channel levels and frequency, and turn the Analog Sine Generator On or Off.

Section II: Configuration

5

Generator Waveforms and Controls

Overview

APx analyzers generate a number of audio stimulus signals for both analog and digital outputs.

Signals for digital output are always generated in the digital domain, using digital signal processing (DSP) in the attached computer. Generally, signals for analog output are generated in the digital domain and converted to analog signals using precision digital-to-analog converters (DACs). The APx555 has an additional analog oscillator to generate very low distortion sine signals, and the 555 and analyzers equipped with the AG52 option also have an analog square wave generator.

This chapter looks first at the waveforms generated in DSP and used for digital and analog testing. Next we look at signals used exclusively for digital testing, beginning on page 47. On page 49 we look at the APx555's analog oscillator and its capabilities, and finally on page 49 we examine the AG52 and 555 square wave and DIM waveforms.

DSP signal generation

For the most part, APx stimulus signals are generated in digital signal processing (DSP). Signals are created mathematically or played from digital audio files on disk. These signals can be embedded in digital output streams, or converted to analog output signals using precision digital-to-analog converters (DACs).

Not all selections are available for every hardware or software configuration.

Signals available

For analog (via DAC) or digital output:

- Sine (frequency range 0.1 Hz to 80 kHz; optional DC Offset). See page 42.
- Sine, Burst (available only for the APx555, analog) See page 50.
- Sine, Dual (2 frequencies, on separate channels or summed). See page 43.
- Sine, Var Phase (1 frequency. An optional phase offset can be applied to other channels). See page 43.
- IMD signals (2 sine frequencies summed according to specific definitions). See page 43.
- Noise (various shapes available). See page 45.
- Generator Waveform (arbitrary waveform from file on disk). See page 45.

Some measurements use specific, batch-mode stimulus signals:

- Continuous Sweep (Farina log-swept sine "chirp," used in Acoustic Response, Continuous Sweep, Frequency Response Impedance Thiele-Small and Loudspeaker Production Test.) See page 47.
- Multitone (several tones summed into one complex signal; used in the Multitone Analyzer) See page 47.

For digital output only:

- Square. See page 47.
- Bit test (digital diagnostic). See page 48.
- Constant Value (digital diagnostic)
- Walking Ones (digital diagnostic)
- Walking Zeros (digital diagnostic)

Common Controls

Most DAC generated signals have these common controls:

- Levels Track Ch 1
  If Level Tracks Ch1 is checked, all output channels are set to the level entered for channel 1. If it is unchecked, each channel can be set to a different output level.
- Level
  Set the generator level here.

Bit Test, Walking Ones and Walking Zeros have no level controls.

- **DC Offset**
  For a DAC generator continuous sine waves, a DC offset can be added to the signal. For analog output, when sine level is zero the maximum offset is $\sqrt{2}$ times the maximum (±) RMS level for the current instrument and output connector. For digital output, when sine level is zero the maximum offset is 1 D (±).
  If **Level Tracks Ch1** is checked, all output channels have a DC voltage added at the level set in the channel 1 DC Offset field. If it is unchecked, each channel can be offset by a different DC voltage.
  If the generator level is set to zero (0.000 Vrms or 0 FS), the generator will output DC only (pure DC).

Bit Test, Constant Value, Walking Ones and Walking Zeros have no DC offset controls.

Channels

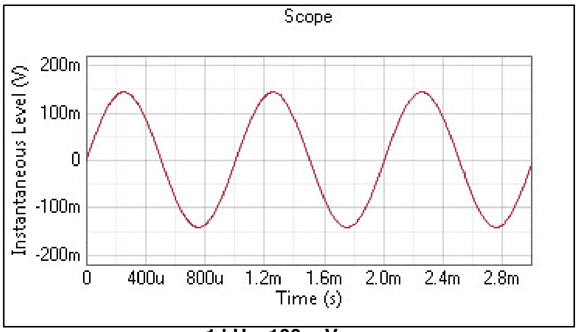
Select the channels to output the generated signal. By default, all available channels are selected.

Waveform Descriptions

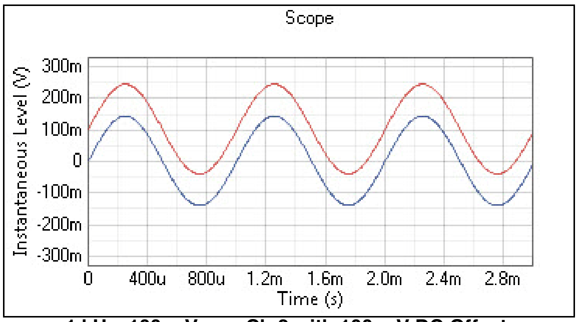
The waveforms cited in the generator descriptions above are examined in more detail here.

DAC Sine Waveform (digitally generated)

The basic sine waveform is a pure tone, a single frequency. Sines are useful in test and measurement because of their purity. When passed through a device under test (DUT), the output signal can be analyzed to see what the DUT has added to the pure sine, whether noise, distortion, non-linear level changes, or other changes.



1 kHz, 100 mVrms



1 kHz, 100 mVrms; Ch 2 with 100 mV DC Offset.

DAC Sine

The digitally-generated DAC sine is discussed here; for the APx555, see High Performance Sine Generator on page 49.

The DAC sine waveform has a frequency range up to 80.1 kHz (see the table below). The purity of the waveform is indicated in the system residual THD+N, to which the generator sine signal is a contributor.

The system residual figure varies with Audio Precision hardware, but is better than:

–102 dB, for the APx515

–103 dB for the APx585/586

–105 dB for the APx525/526 (and APx555, when using the DAC generator).

Please check the APx *Installation Instructions and Specifications* booklet for your instrument for detailed specifications.

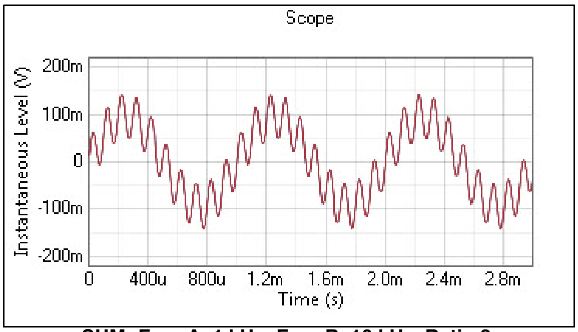
Controls

- **Levels Track Ch 1**
  If Level Tracks Ch1 is checked, all output channels are set to the level entered for channel 1. If it is unchecked, each channel can be set to a different output level.
- **Level**
  Set the generator level here.
- **DC Offset**
  For a DAC generator continuous sine waves, a DC offset can be added to the signal. For analog output, when sine level is zero the maximum offset is $\sqrt{2}$ times the maximum (±) RMS level for the current instrument and output connector. For digital output, when sine level is zero the maximum offset is 1 D (±).
  If **Level Tracks Ch1** is checked, all output channels have a DC voltage added at the level set in the channel 1 **DC Offset** field. If it is unchecked, each channel can be offset by a different DC voltage.
  If the generator level is set to zero (0.000 Vrms or 0 FS), the generator will output DC only (pure DC).

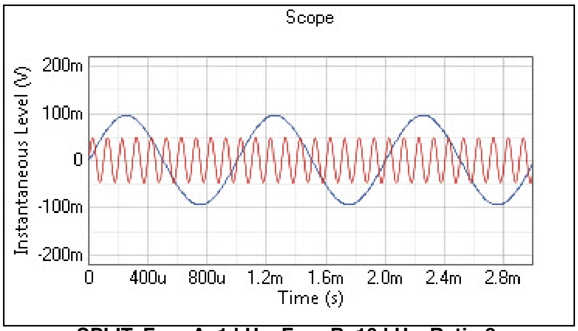
- **Frequency**
  Set the generator frequency here. The DAC generator frequency range is:
  - APx515: 2.00000 Hz to 80.1 kHz
  - APx525/526/582: 0.10000 Hz to 80.1 kHz
  - APx585/586: 5.00000 Hz to 80.1 kHz
  - APx555: 0.00100 Hz to 80.1 kHz
- **Channels**
  Toggle output channels **ON** or **OFF**.

Sine, Dual

Bench Mode and a number of Sequence Mode measurements provide a **Sine, Dual** DSP/DAC generator option, which allows you to modify the sine output in several ways.



SUM: Freq A, 1 kHz; Freq B, 10 kHz. Ratio 2.



SPLIT: Freq A, 1 kHz; Freq B, 10 kHz. Ratio 2.

Controls Specific to Sine, Dual

Frequency A and B

Sine, Dual allows you to set the DSP generator to produce two signals of different frequencies. These can be output independently or summed; see below. Enter the values for **Frequency A** and **Frequency B** here.

Sum

Select **Sum** and enter different frequencies in **Frequency A** and **Frequency B**. The two sine waves are summed to become the output waveform. Use the **FB:FA Level Ratio** control to set the level ratios of the two component sine waves.

Split

Select **Split** and enter different frequencies in **Frequency A** and **Frequency B**. The A and B signals can be routed to different output channels. See Channel Assignments, below.

FB:FA Level Ratio

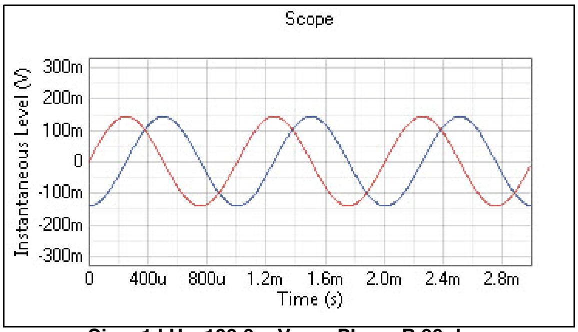
For both Sum and Split, this control sets the ratio of levels between Frequency A and Frequency B signals.

Channel Assignments

For **Split**, use the **Channel Assignments** menu to map frequencies A and B to output channels. By default, frequency A is mapped to odd-numbered channels, and frequency B is mapped to even-numbered channels. For assignments across multiple channels not available in the Menu, select **Custom**.

Sine, Var Phase

Bench Mode and a number of Sequence Mode measurements provide a Sine, Var Phase generator option, which allows you to modify the sine output in several ways.



Sine, 1 kHz, 100.0 mVrms. Phase B 90 deg.

Controls Specific to Sine, Var Phase

Phase B

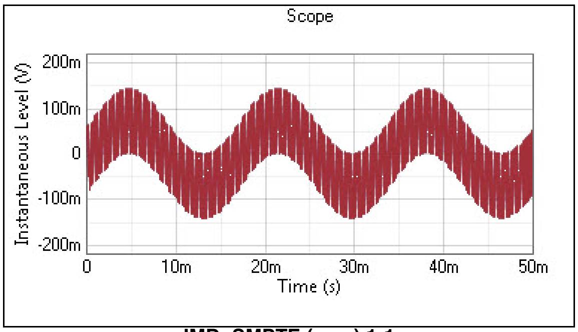
The channel assigned as A is the reference channel. Channels assigned as B can have a phase shift relative to A.

Channel Assignments

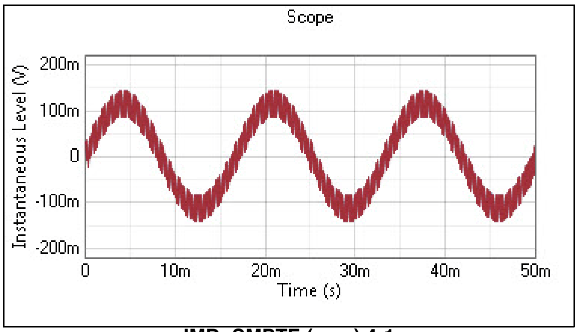
Use the **A/B Channel Assignments** menu to map phase A and phase B to output channels. By default, phase A is mapped to odd-numbered channels, and phase B is mapped to even-numbered channels.

IMD waveforms

Bench Mode and a number of Sequence Mode measurements provide an IMD DSP/DAC generator option, which allows you to modify the sine output to provide IMD (intermodulation distortion) waveforms.



IMD: SMPTE (sum) 1:1



IMD: SMPTE (sum) 4:1

Controls Specific to IMD

DIM signals require the APx analog square wave as a component, restricting DIM to the APx555 and analyzers with the AG52 option installed.

Type

See More about IMD on page 273 for a more detailed description of IMD signals and measurement techniques.

SMPTE 10:1
This is a SMPTE-type stimulus, with a F1:F2 ratio of 10:1. See Frequency 1 and Frequency 2 below.

SMPTE 4:1
This is the standard SMPTE stimulus, with a F1:F2 ratio of 4:1. See Frequency 1 and Frequency 2 below.

SMPTE 1:1
This is a SMPTE-type stimulus, with a F1:F2 ratio of 1:1. See Frequency 1 and Frequency 2 below.

DIN
This is the standard DIN stimulus, with a F1:F2 ratio of 4:1. DIN is quite similar to SMPTE 4:1, using somewhat different frequencies. See Frequency 1 and Frequency 2 below.

DFD/CCIF
DFD and CCIF use the same stimulus tones, with slightly different reporting, discussed in More about IMD on page 273. The tones are of high frequency and close together, placed around a mean frequency and separated by a difference frequency. See Mean Frequency and Diff Frequency, below.

DIM

The DIM waveform is only available for analog outputs in instruments fitted with an analog square wave generator: the APx555, or an APx520, 521, 525, 526 or 582 with option AG52.

DIM 30
DIM 30 mixes a square wave (fixed at 3.15 kHz) with a lower-amplitude sine wave (fixed at 15 kHz). The square wave is filtered by a single-pole low-pass filter at 30 kHz. The ratio of the peak voltage of the square wave to the peak voltage of the sine wave is 4:1.

DIM 100
DIM 100 mixes a square wave (fixed at 3.15 kHz) with a lower-amplitude sine wave (fixed at 15 kHz). The square wave is filtered by a single-pole low-pass filter at 100 kHz. The ratio of the peak voltage of the square wave to the peak voltage of the sine wave is 4:1.

DIM B
DIM B (broadcast) mixes a square wave (fixed at 2.96 kHz) with a lower-amplitude sine wave (fixed at 14 kHz). The square wave is filtered by a single-pole low-pass filter at 30 kHz. The ratio of the peak voltage of the square wave to the peak voltage of the sine wave is 4:1.

DIM B8
DIM B8 mixes a square wave (fixed at 2.96 kHz) with a lower-amplitude sine wave (fixed at 8 kHz). The square wave is filtered by a single-pole low-pass filter at 30 kHz. The ratio of the peak voltage of the square wave to the peak voltage of the sine wave is 4:1.

Frequency 1 and Frequency 2

These controls are for SMPTE and DIN IMD stimulus signals.

For SMPTE, **F1** is typically 60 Hz; **F2** is typically 7 kHz. **F1** range is 40 Hz to 1 kHz; **F2** range is 2 kHz to 60 kHz.

For DIN, **F1** is typically 250 Hz; **F2** is typically 8 kHz. **F1** range is 40 Hz to 1 kHz; **F2** range is 2 kHz to 60 kHz.

Mean Frequency and Diff Frequency

These controls are for DFD/CCIF stimulus signals. **Mean Freq** is typically 12.5 kHz; **Diff Frequency** is typically 80 Hz. **Mean frequency** range is 250 Hz to 60 kHz; **Diff frequency** range is 80 Hz to 2 kHz.

Square Freq and Sine Freq

These are display fields for DIM stimulus signals. **Square Freq** is fixed at 3.15 kHz (2.96 kHz for B signals); **Sine Freq** is fixed at 15 kHz (14 kHz for B, 8 kHz for B8).

Sum

Select Sum for typical IMD testing. The two signals are summed to become the output waveform. For SMPTE/DIN, the ratio of F1:F2 is set by the IMD Type selected.

Split

Select Split for microphone distortion testing, or other IMD testing that mixed the signals in the DUT or acoustic environment.

Split is useful for microphone distortion testing because a loudspeaker is typically used as a sound source for microphone testing, and loudspeakers have high THD levels compared to the microphone under test, making THD testing impossible. The alternative is to perform IMD testing on the microphone, using two loudspeakers, each outputting a different component of the IMD signal. The mixing of the signals occurs in the microphone.

For SMPTE/DIN, the ratio of F1:F2 is set by the IMD Type selected.

Channel Assignments

For Split, use the Channel Assignments menu to map frequencies 1 and 2 to output channels. By default, frequency 1 is mapped to odd-numbered channels, and frequency 2 is mapped to even-numbered channels. For assignments across multiple channels not available in the Menu, select Custom.

More About Noise Signals

The Noise signals in APx500 are generated in DSP. They have the following characteristics:

White noise

White noise has a power spectrum ideally constant per unit bandwidth from just above DC to half the sampling rate. The crest factor is between 3 and 5, and the probability distribution is close to Gaussian. The generation period (before the waveform repeats) is 2^{32-1} samples long, which is about a day at 48 kHz.

Pink noise

Pink noise is generated from white noise using a filter, and inherits the statistics of the white noise. Pink noise has a power spectrum ideally constant per fractional bandwidth from just above DC to half the sampling rate. The signal power drops off below 10 Hz. The ideal pink power spectrum is maintained ±1.0 dB in the range 2×10^{-4} SR to 0.45 SR.

IEC 60268-1 noise

Noise according to the IEC 60268-1 standard is generated from pink noise passed through a weighting filter. IEC 60268-1 has a spectrum that mimics program material, including voice and music.

BS EN 50332-1 noise

BS EN 50332-1 is IEC 60268-1 noise with a reduced crest factor, accomplished by applying soft clipping. The crest factor is between 1.8 and 2.2.
BS EN 50332-1 noise supports headphone testing.

Setting Generator levels when using noise signals

The RMS units assume a sine wave, which has a crest factor of the square root of 2 (approximately 1.414). None of the noise signals have this crest factor, so actual noise values will not be equal to the generator level in RMS-based units.

White noise is generated using a method that guarantees that the peak value is equal to the generator level in peak units; Vp and Vpp will accurately specify the noise level. Note that for digital outputs there is no generator level peak unit.

The filters used in pink noise, IEC 60268-1 noise and BS EN 50332-1 noise alter the signal so that the maximum output cannot be guaranteed in the same way that the white noise can. Signal levels for these noise types will not be equal to the generator level setting for any units, but will be bounded by the Vp setting.

Generator Waveform Files (arbitrary waveforms)

Test signal generation from a waveform file

In addition to the DSP-generated waveforms, APx500 also has the capability to use custom or arbitrary waveforms, loaded into the generator from waveform files available to the computer running APx500. This is particularly useful when coded audio files (dts, Dolby or others) are required to stimulate a decoder in your DUT.

The loaded waveform is “looped” for generator output, so that the end of the waveform is immediately followed by the beginning, providing continuous playback without a jump discontinuity.

Generator waveform files are not restricted to sine waveforms; other waveforms can be used. The Waveform Generator Utility (APxWfmGenerator.exe) allows you to generate .wav files for a wide range of test signals, with selectable sample rates, bit depths and channel counts. It is available from the Audio Precision Web site at “www.ap.com/downloads/file/273”. You are free to copy this utility to other computers or media, where you can generate the audio files you may need as generator waveforms or for external source testing.

You can also create your own generator waveform files. To be useful for measurement, user-created files must be designed with the requirements of the measurement context in mind. Not all audio signals are compatible with certain analyzer measurements nor are useful for measurement. APx500 supports a number of file formats, sample rates and multichannel file formats for Generator Waveform files. See Supported audio waveform file formats... for a detailed list.

Using a generator waveform file

A measurement that supports generator waveform files displays a Waveform selection list in the Signal Generation panel. The default is Sine (generated in DSP). Click the Waveform menu in the Generator and choose Browse for file... to select an audio file for the generator to use. You can select multiple files.

A selected file becomes the source for that measurement's generator output and is attached to the project file. Once attached to the project, an audio file is available to all other supported measurements as a selection on the Waveform menu. Files attached to the project but not currently used by any measurement can be removed by navigating to the dialog at File > Manage Attached Project Items...

Signal Output

Analog output: linear audio only

Coded waveforms cannot be decoded to analog audio within the APx500 system.

Linear mono or stereo audio files can be sent to the analog outputs. Mono waveforms are sent to all available output channels. By default, stereo waveforms map channel A audio to odd output channels, channel B audio to even output channels. These assignments can be remapped in the Advanced Settings dialog.

Analog output: level adjustment

The level of linear audio files sent to the analog outputs can be adjusted.

Consider that the signal level of the embedded audio in any file is unknown: the file may contain a sine wave at 0 dBFS, or it may contain pink noise at –42 dBFS. This being the case, the signal level at the generator output is also indeterminate. Level adjustments for generator waveform files are calibrated assuming the maximum possible level in the file.

For analog output, a 1 Vrms level setting would output a 0 dBFS sine wave at 1 Vrms; lower signal levels in the file would be output at correspondingly lower voltages.

Digital output: linear or coded audio

Linear mono or stereo audio files and coded audio files can be sent to the digital outputs. Whenever a generator waveform file is selected, the APx500 output sample rate is set to the rate of the file.

Mono waveforms are sent to all available output channels. By default, stereo waveforms map channel A audio to odd output channels, channel B audio to even output channels. These assignments can be remapped in the Advanced Settings dialog. Coded waveforms are not decoded in the APx500 system, but are sent bit exact.

Digital output: level adjustment

When Bit Exact is not set, the level of linear audio files sent to the digital outputs can be adjusted. Level adjustments are not available for coded waveforms, which are send bit exact.

Consider that the signal level of the embedded audio in any file is unknown: the file may contain a sine wave at 0 dBFS, or it may contain pink noise at –42 dBFS. This being the case, the signal level at the generator output is also indeterminate. Level adjustments for generator waveform files are calibrated assuming the maximum possible level in the file.

For digital output, a 0 dBFS level setting would output a 0 dBFS sine wave at 0 dBFS; lower signal levels in the file would be output at correspondingly lower digital levels. A –15 dBFS signal in the file with a –12 dBFS level setting would be output at –27 dBFS.

Bit Exact

When a digital output is selected, the Bit Exact option is available when a linear generator waveform is in use. Checking Bit Exact forces the APx generator to output the waveform with no changes whatsoever, with every audio bit exactly as it is in the disk file. Level adjustment is unavailable and dither is off. Coded audio waveforms are always output as Bit Exact.

Dither

For digital output (when Bit Exact is not set), a generator waveform file signal is always redithered.

Generator waveform length constraints

File Truncation

Generator waveforms are limited in file length by APx instrument memory resources. A waveform file that exceeds the maximum length will be truncated as it is loaded. A truncated waveform is marked with a warning icon when selected as a generator source.

A generator waveform file is downloaded from the PC into the APx analyzer hardware as needed for a measurement. The hardware memory has a capacity of 128 MB, and audio waveforms that exceed 128 MB will be truncated.

Since APx500 converts 8- and 16-bit PCM waveforms to 32-bit for generator output, and expands coded audio for playback, generator waveform files may use considerably more memory than indicated by their file size on disk.

Truncated waveforms should be avoided, as a jump discontinuity will be introduced on playback at the point of truncation. Best practice is to use the shortest generator waveform file consistent with your measurement requirements.

Maximum File Length table

File formats and sample rates affect the amount of memory used. This table is provided as a guide, indicating the running times that represent the maximum memory available for various formats.

DTS transmit rates

When a legacy dts file (*.dts or *.cpt) with an embedded sample rate of 44.1 kHz is transmitted via HDMI or DSIO, it can sent using an interface transmit rate of 44.1 kHz or 176.4 kHz (4x). If the dts file has an embedded sample rate of 48 kHz, it can be sent using an interface transmit rate of 48 kHz or 192 kHz.

These settings do not convert the sample rate of the embedded audio but only affect how the audio is carried on the interface signal. For DSIO, the Digital Serial Output Sample Rate setting must agree with setting made here.

Select

- Auto (the default)
  For a 44.1 kHz sample rate file, Auto sets the interface transmit rate to 44.1 kHz.
  For a 48 kHz sample rate file, Auto sets the interface transmit rate to 48 kHz.
- 4x
  For a 44.1 kHz sample rate file, 4x sets the interface transmit rate to 176.4 kHz.
  For a 48 kHz sample rate file, 4x sets the interface transmit rate to 192 kHz.

Continuous Sweep method

The Continuous Sweep measurement outputs a special broadband stimulus signal to the DUT. The DUT output is acquired by the analyzer and is processed in DSP (digital signal processing) to provide a number of sweep results, all from one acquisition. (The Deviation result is not graphed as a sweep, but is viewed as a single value for each channel).

The Continuous Sweep, Frequency Response and Acoustic Response measurements use the continuous sweep method. Read more about continuous sweep beginning on page 220.

Multitone method

The multitone method uses a stimulus signal made up of many sine waves (a multitone) and analyzes the DUT output with FFT techniques to measure the amplitude and phase of each component sine wave in the result. These values can provide frequency response, phase response, crosstalk response, distortion response, noise response and other results.

Multitones are useful for fast testing applications, and since a multitone stimulus can be brief and unobtrusive, multitone can be considered for in-service testing. Read more about multitone beginning on page 373.

Square wave

The square wave has been used in audio test for many years as a diagnostic tool. A glance at a square wave passed through a DUT and displayed in the time domain (scope) view can reveal a lot about the DUT's coupling, frequency response, slew rate, power supply sag and other characteristics to the experienced eye. In some APx instruments, a square wave is used as a component of the DIM/TIM waveform.

Digitally generated square waves (DSP/DAC) exhibit Gibbs phenomenon ringing artifacts, an unavoidable consequence of mathematics of the DSP generation. In APx instruments, digitally generated square waves are not available for analog output.

The APx555 and an AG52-equipped APx525/526/582 have an analog square wave generator that produces fast rise time square waves with little ringing or overshoot. This waveform is only available to the analog outputs, and also serves as a component of the DIM/TIM waveform.

Controls

Levels Track Ch 1

If Level Tracks Ch1 is checked, all output channels are set to the level entered for channel 1. If it is unchecked, each channel can be set to a different output level.

Level

Set the generator level here.

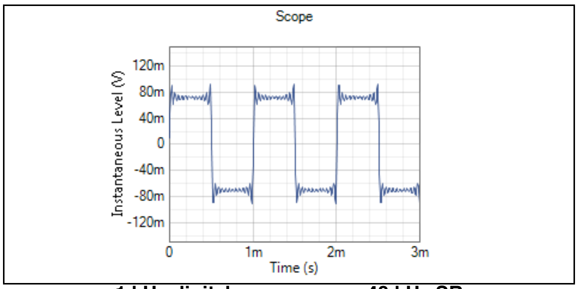
DC Offset

Set the DC Offset here. This control is only available for digitally generated square waves.

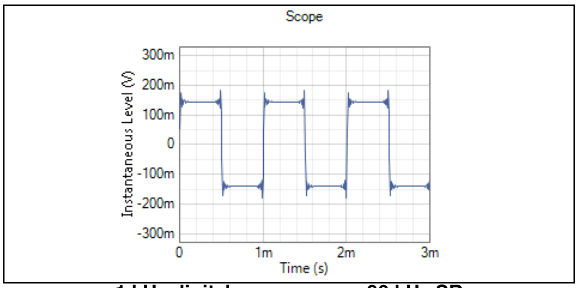
Frequency

Set the generator frequency here. For digital square waves, the range is 100 mHz to SR/6. For analog square waves, the range is 100 mHz to 30 kHz.

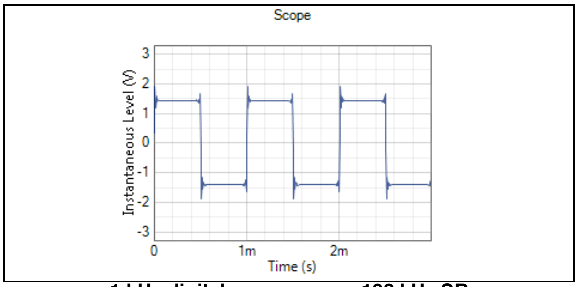
Comparison, digitally generated square waves and analog generated square wave



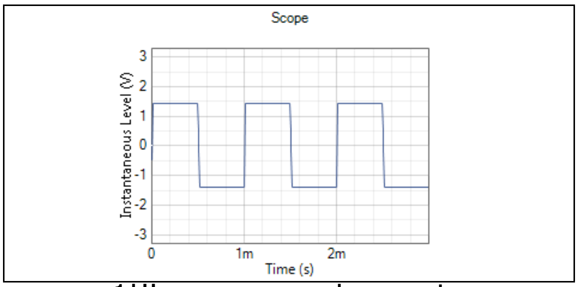
1 kHz digital square wave, 48 kHz SR



1 kHz digital square wave, 96 kHz SR



1 kHz digital square wave, 192 kHz SR



1 kHz square wave, analog generator

Special waveforms for Digital Diagnostics

Walking Bits patterns

The walking bit waveforms are used in diagnosing digital problems in the **Verify Connections** feature of Signal Path Setup. The walking pattern makes it easy to observe "stuck" bits in digital hardware. The walking bits waveforms will also show no errors when passed through a bit-accurate system.

Bit test

Bit Test is a pseudo-random waveform with values uniformly distributed between plus and minus full scale. Algorithms in the generator and analyzer determine sample values.

Bit test exercises a wide range of levels and frequencies and is the most thorough of the bit-accurate waveforms.

Walking Zeros

In the Walking Zeros waveform, all bits are set to 1 except one bit, which is set to 0. This single low bit is continuously incremented from the least significant bit (LSB) to the most significant bit (MSB). When it reaches the MSB it wraps around to the LSB of the selected word length.

Walking Ones

In the Walking Ones waveform, all bits are set to 0 except one bit, which is set to 1. Like Walking Zeros, this single high bit is continuously incremented from LSB to MSB and wrapped back to LSB.

Samples/Step

The time required for one complete cycle of the walking pattern depends upon the output sample rate and the generator Samples/Step value. For example, with Samples/Step set to 48000 and sample rate set to 48 kHz, each step will last for 1 second.

Constant Value waveform

The Constant Value waveform is a continuous stream of data samples at the same fixed value (digital dc). Constant Value mode aids in the investigation of data-dependent errors in digital systems.

Value

Enter the value for the digital data here, in either hexadecimal (base 16) or D units. Specifying the value in hex is often easier, especially if the output bit depth is configured for less than 32 bits. For instance, if the output is configured for 16 bits, the lowest positive constant value is 00010000 hex; this is easier to specify than 30.52 µD.

Note: Constant Value signals of all ones (1s) or all zeros (0s) are treated as invalid Constant Value signals, and will return errors. This implementation serves to flag digital connection errors, which can incorrectly produce a stream of constant ones or zeros. When using Constant Value signals, set the signal to any value except ones or zeros.

Notes on hex values in APx500 Constant Value

1. Truncation of hex numbers at lower bit depths.

The output bit depth selected in Signal Path Setup defines the range of valid hex values for digital levels. For example, a bit depth of 24 requires 6 hex digits (1 hex digit = 4 binary digits). If the bit depth is reduced to 16, only 4 hex digits are required, so the 2 least significant hex digits are not used. If, while bit depth is set to 16, you enter 6 digit hex value (12345600), APx will return the truncated value 12340000.

2. Conversion of hex values to two's complement.

As is almost universal in digital audio systems, APx500 uses two's complement notation for digital audio sample values. Two's complement simplifies digital processing by expressing bipolar signals as positive-only numbers. In APx, hex values are always shown in their two's complement form.

For example, entering '-12345600' with the output configuration in 24-bit mode results in the display 'EDCBAA00 hex', which is the two's complement representation of -12345600 hex.

Units available for Constant Value are

- hex
- D

See Chapter 98 for more information about units of measurement.

Active Bits

When Active Bits is set, any bit that changes state during the diagnostic period (about 1/4 second) is shown as highlighted with color. Inactive bits are shown as white.

Data Bits

When Data Bits is set, the logical state (binary 1 or binary 0) of each bit in the audio word is shown. This is updated at the end of each diagnostic period (about 1/4 second). Binary 1 is shown in highlighted color; binary 0 is shown as white.

Note: The audio word may be any length between 24 bits and 8 bits, depending upon the Bit Depth setting in Signal Path Setup.

Error Rate (digital only)

Read more about bit-accurate measurements beginning on page .

The Error Rate display recognizes Audio Precision bit-accurate waveforms, including

- Bit test (usually the best choice for digital error rate measurements.)
- Walking Zeros
- Walking Ones
- Constant Value

When one of these waveforms is received with every sample at precisely the same value as was generated, zero errors are reported and an error rate of 0.000% is displayed. When one of these waveforms is received with sample values different from the values in the original waveform, each differing sample is reported as one error.

Analog High Performance Sine Generator (APx555)

The APx555 has a low-distortion RC oscillator which can be selected for sine generation. It is considerably lower in residual harmonic distortion and noise than the DSP/DAC generator, and in combination with the APx555's High Performance Sine Analyzer enables the 555's signature low system residuals.

Signals available

- Sine (frequency range 5 Hz to 204 kHz). Very low distortion and extended frequency range. See page 49.
- Sine, Burst. See page 50.

Analog square wave generator (APx555 and AG52)

Digitally synthesized square waves (DSP/DAC) exhibit Gibbs phenomenon ringing artifacts. The APx555 and an AG52-equipped APx525/526 have an analog square wave generator that produces fast rise time square waves with little ringing or overshoot.

Signals available

- Square
  A fast rise time analog square wave.
- DIM
  The APx555 and AG52 use the analog square wave generator in combination with the sine generator to produce a DIM IMD waveform.

Sine

The APx555 has a low-distortion RC oscillator which can be selected for sine generation, called the High Performance Sine Generator. It is considerably lower in residual harmonic distortion and noise than the DSP/DAC generator, and in combination with the APx555's High Performance Sine Analyzer enables the 555's signature low system residuals.

The analog sine waveform has a frequency range of 5 Hz to 204.475 kHz. The purity of the waveform is indicated in the system residual THD+N, to which the generator sine signal is a contributor. For the APx555, this figure is better than:

–116.5 dB.

Please check the *APx555 Installation Instructions and Specifications* booklet for detailed specifications.

Controls

Levels Track Ch 1

If Level Tracks Ch1 is checked, all output channels are set to the level entered for channel 1. If it is unchecked, each channel can be set to a different output level.

Level

Set the generator level here.

Frequency

Set the generator frequency here. Range is 5 Hz to 204.475 kHz.

Comparison of the High Performance Sine Generator to the DAC sine generator

Advantages

- Very low THD+N residuals.
- Extended high-frequency range (up to 204 kHz).

Disadvantages

- Split modes, summed modes, offset and other modifications or impairments are not available.
- Frequency accuracy is less precise. (DAC generator 3 ppm; RC oscillator 0.3%).
- RC oscillator cannot be synchronized to a reference.

Sine, Burst

The APx555 provides a general purpose sine burst waveform, which can have an arbitrary sine frequency, an arbitrary High Time and Interval time, and an arbitrary ratio between On and Off levels. Sine, Burst is only available in the APx555, and only for analog outputs.

Read more about burst signals beginning on page 339.

Controls

Levels Track Ch 1

If Level Tracks Ch1 is checked, all output channels are set to the level entered for channel 1. If it is unchecked, each channel can be set to a different output level.

Level

Set the generator level for the On portion of the burst here.

Frequency

Set the generator frequency here. The Sine, Burst generator frequency range is 1 mHz to 80.1 kHz.

Cycles/Seconds

Choose whether to set the burst High Time and Interval in cycles (at the current sine frequency) or seconds.

High Time

For Cycles, set the number of cycles of the sine waveform for the On portion of the burst. Minimum is 1; maximum is 16,777,214.

For Seconds, set the duration in time for the On portion of the burst.

Interval

For Cycles, set the number of cycles of the sine waveform for the total interval between burst onsets. Minimum is 2; maximum is 16,777,215.

For Seconds, set the duration in time for the total interval between burst onsets.

Low/High Ratio

Set the ratio of the level of the Off portion of the burst to the On portion of the burst. The level of the On portion of the burst is set in the Level control.

The Off portion can be set as high as unity (0 dB), to as low as –80 dB. The low level can also be set to –∞ dB (completely off). To set –∞ dB, enter a value of –10000 dB or smaller.

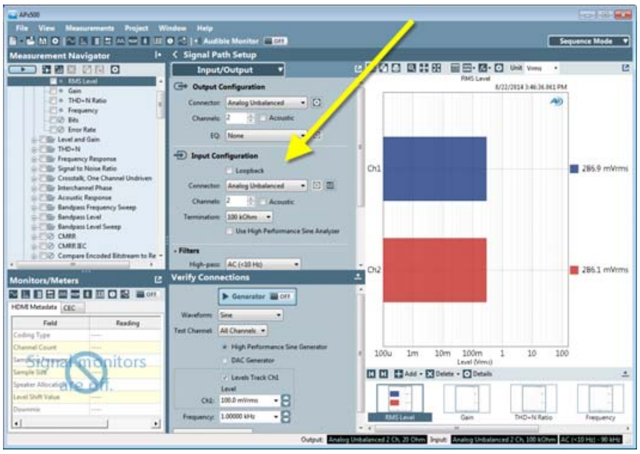
Channels

Toggle output channels ON or OFF.

6

Signal Path Setup

Overview: Signal Path Setup



See Chapter 3, **Introduction to the APx Software** to read more about Signal Paths.

For both Sequence Mode and Bench Mode, **Signal Path Setup** provides a series of menu-selected panels, with access to a collection of tools to select and configure the physical connections between the analyzer and the Device Under Test (DUT) for the current **Signal Path**. This is where you choose your input/output (I/O): digital or analog, balanced or unbalanced, specify impedances or communications protocols, set the number of channels in use, name your inputs and and so on. You can also configure references, switchers, DCX, clock sync, triggers and jitter.

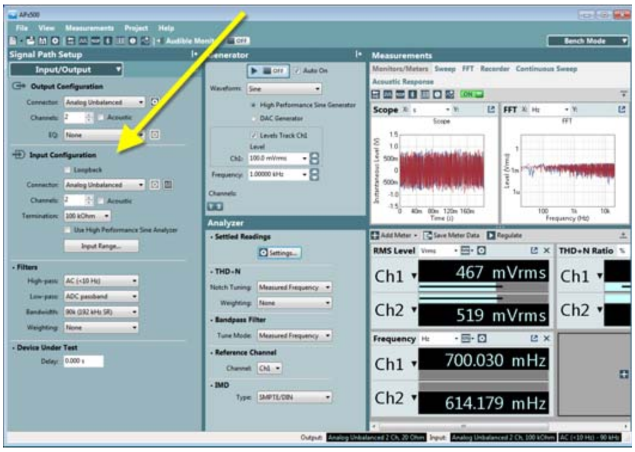
Individual Signal Path Setup panels can be undocked and placed anywhere on the screen.

In **Sequence Mode**, **Signal Path Setup** is in the center of the workspace, along with the control panel for **Verify Connections**. See Chapter 7 for more information about Verify Connections.

In **Bench Mode**, **Signal Path Setup** is on the left side of the workspace.

A drop-down menu provides these selections:

- **Input/Output**, page 52.
  Input and Output configuration, acoustic mode selectors, labels and colors, global input filters, device delay.
- **References**, page 54.
  Output and input level references, mic calibration, frequency reference.
- **Output Switchers**, page 57.
  Output switcher configuration. Available if an output switcher is connected.
- **Input Switchers**, page 57.
  Input switcher configuration. Available if an input switcher is connected.
- **DCX**, page 58.
  Settings for DCX-127 module. Available if a DCX-127 is connected.

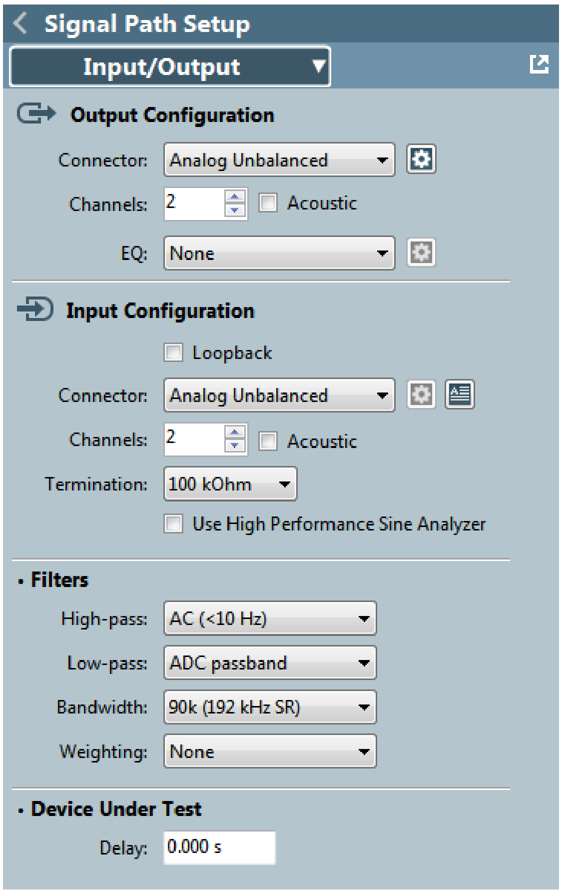


- **Clocks**, page 59.
  Configuration for external reference and synchronization signals, and jitter generation. Available if the Advanced Master Clock is installed.
- **Triggers**, page 61.
  Configuration for trigger output and external trigger input. Available if the Advanced Master Clock is installed.

The Back button

In Sequence Mode, if you have jumped to Signal Path Setup from a measurement, the back button will return you to the measurement.

Signal Path Setup: Input/Output



Dependency upon Analyzer Hardware

APx500 software works with all APx analyzer hardware models and options. The differences in using APx500 with different analyzer hardware models are most apparent in Signal Path Setup, where input and output format, configuration and channel count are dependent upon the analyzer hardware model and options.

APx500, for example, will not show HDMI connection selections when connected to an analyzer that has no HDMI interface installed.

This chapter provides an overview of Signal Path Setup and discusses features common to all analyzer hardware configurations.

I/O Configurations

The Signal Path Setup I/O Configuration panels are where you select the generator output and analyzer input connectors and formats. For each selection, a **Settings** dialog provides complete control over output settings.

Each of these I/O configurations are discussed in detail in the following chapters:

- **Analog I/O** (input/output) configurations for the APx515 analyzer, Chapter 8.
- **Analog I/O** configurations for the APx525 family of analyzers, Chapter 9. APx582 output configuration is also covered in Chapter 9.
- **Analog I/O** configurations for the APx555 analyzer, Chapter 10.
- **Analog I/O** configurations for the APx585 family of analyzers, Chapter 11. APx582 input configuration is also covered in Chapter 9.
- **DIO** (digital I/O) configurations for all analyzers, Chapter 12.
- **ADIO** (advanced digital I/O) configurations, Chapter 13.
- **DSIO** (digital serial I/O) configurations for the APx525, 582 and 585 families, Chapter 14.
- **HDMI** (high definition multimedia interface) and **HDMI ARC** I/O configuration for some models in the APx525, 582 and 585 families, Chapter 15.
- **Bluetooth** wireless technology configurations for APx525, 582 and 585 families, Chapter 16.
- **PDM** (pulse density modulation) configuration for APx525, 582 and 585 families, Chapter 17.
- **ASIO** (audio stream input/output) configuration for all analyzers, Chapter 18. APx515 requires a software option to enable this interface.

For Output Configuration, there is this additional selection:

- **External Source** (open-loop, no generator output) configuration for all analyzers, Chapter 5.

For Input Configuration, there are these additional selections:

- **Loopback**
- **File (Analog Units)** and **File (Digital Units)**, Chapter 19.

Output EQ

For analog outputs, **Output EQ** allows you to insert an equalization filter in the output signal path, after the APx generator. This enables compensation for a non-flat response in the device under test. Read more about **Output EQ** on page 168.

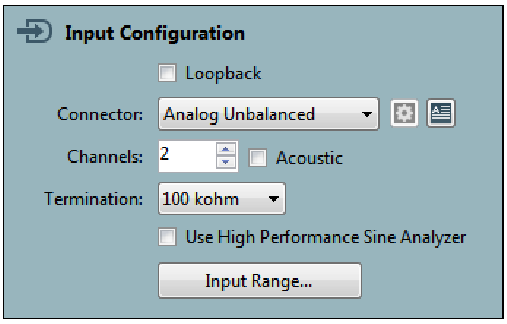
Acoustic Output Configuration

For convenience when performing acoustic tests, APx500 can be set to **Acoustic** output configuration. **Acoustic** output configures all generator level setting fields and generator level graph displays to units of pascals or dBSPL. Read more about **Acoustic** output configuration on page 167.

Acoustic Input Configuration

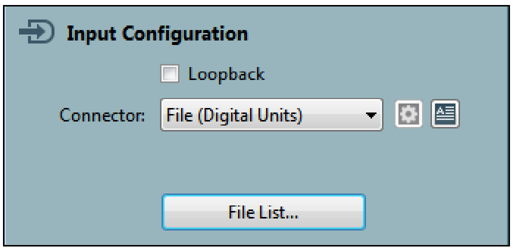
For convenience when performing acoustic tests, APx500 can be set to **Acoustic** input configuration. Acoustic input configures all APx500 measurements in the signal path to display levels relative to channel-specific acoustic reference levels. This is useful for making acoustic measurements with multiple microphones. Read more about **Acoustic** input configuration on page 167.

Input Range (Bench Mode only)



For analog inputs in Bench Mode, the **Input Range...** dialog is available to set input ranging. This feature is available in Sequence Mode in the Advanced Settings dialogs for individual measurements. See Chapter 92, Ranging and Settling.

File List (Bench Mode only)



For File Input configuration in Bench Mode, the **File List...** browser is available. This feature is available in Sequence Mode in the Analyzer settings for individual measurements. See Chapter 19, Open Loop measurements.

Filters

Signal Path **Filters** (high-pass, low-pass and weighting) can be set, which affect the input signal for the entire Signal Path.

Note: AC/DC coupling is set by the high-pass filter control.

Read more about Signal Path Filters starting on page 545.

Note: for digital inputs when jitter is selected as the measurement source, the Signal Path filters provide different filtering options.

Device Under Test: Delay

Overview

This field helps enable problem-free measurements on high-delay systems (such as Bluetooth links).

- In a high-delay system, the signal from the APx generator takes some time to propagate through the system to the APx analyzer input. APx may then attempt to make a measurement before the generator signal has arrived. This can result in poor measurements. Some examples:
- A Frequency Response, Continuous Sweep, or Acoustic Response measurement may show very low gain at higher frequencies, because the end of the stimulus has been cut off.
- A settled meter measurement such as THD+N may show an incorrect value because of the inclusion in the settling of measurements made before the generator signal arrived.

Although these problems can typically be corrected by adjusting the “Extend Acquisition By” control in the continuous sweep measurements, or by adjusting the settings parameters in meter measurements, this requires the user to change every measurement. These solutions are also less than optimal in that measurements will still be made before the generator signal arrives. This takes extra CPU time.

This field allows you to inform APx of the device delay. APx500 will then refrain from making measurements for that delay time after the generator has turned on. This allows the generator signal enough time to propagate through the device before analysis occurs. There is then no need to adjust acquisition or settling parameters.

Once the Delay field has been set, it applies to all measurements in the signal path. There is no need to further adjust individual measurements to account for the delay.

Determining the Delay value

There is no need to set the Delay field with complete accuracy. Many devices have a delay that varies with frequency anyway. The aim is to instruct APx to ignore the bulk of the data that arrives at the analyzer input before the generator signal has had a chance to propagate through the device. It does not matter if this is off by up to around 10 ms.

When the Device Delay is known

For devices with short delays (<10 ms), APx will perform correctly without any need to add delay in this field. For longer delays, it is best to set the field to the approximate device delay. If the delay is known (it may be a published parameter of the device), then the delay value can be entered directly into this field. Note that for digital devices, the delay may depend on the sample rate.

When the Device Delay is unknown

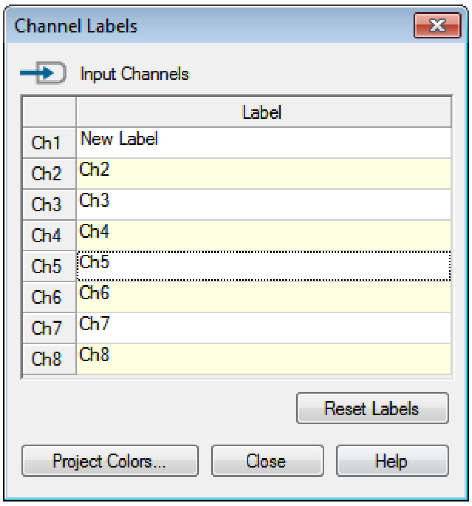
If the delay is not known, it can be measured using the Continuous Sweep or Acoustic Response measurements. The Group Delay, Impulse Response, or Delay results may be used for this. Note that if the device delay is large, the end of the chirp sweep may be cut off. The effect of this can be seen in the Level result, which will drop off very sharply at a frequency below the Stop frequency of the signal. This is unlikely to affect the computed delay. However, the Extend Acquisition By value can be increased to verify this.

Delay measurements with DUT delay set

Note that any delay measurements in the signal path do not take the Device Under Test Delay setting into account. For instance, if the Delay field is set to 50 ms, and the measured delay of the device is 2 ms according to Continuous Sweep, then the actual device delay is 52 ms.

Input Labels

In **Signal Path Setup**, click the Labels button to open the Channel Labels dialog.



Here you can rename the input channels with names of your choice. These settings are project-wide, but can be overridden for any measurement result.

For local bar graph (meter) label overrides, see page 564. For local XY graph label overrides, see page 572.

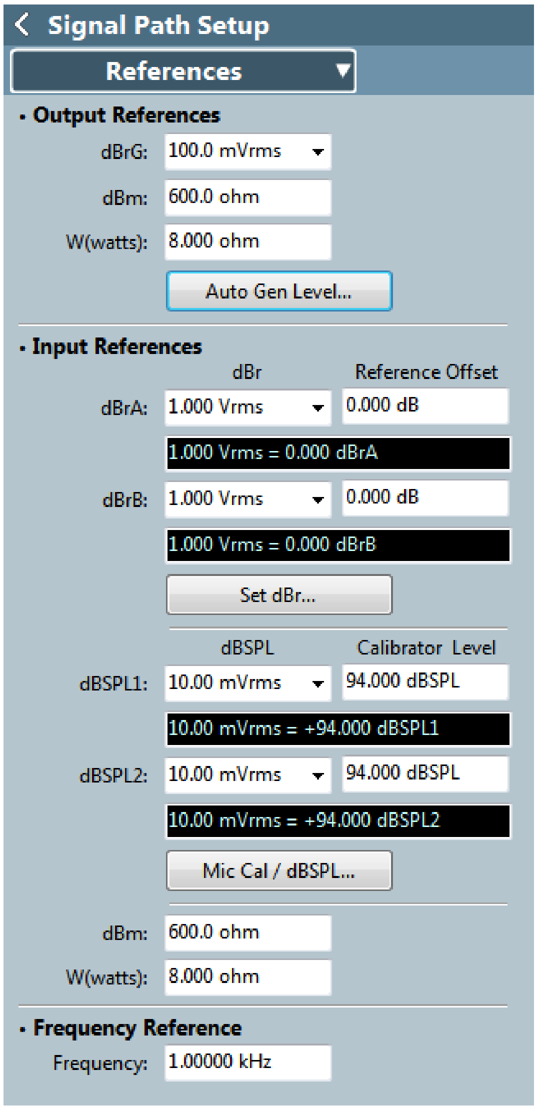
To set project colors and line styles, click the **Project Colors** button in the **Labels** dialog, This opens the Project Properties Colors tab, where you can set graph colors for each channel, or set a color cycle for multiple Data Sets. Read about setting colors on page 22.

Signal Path Setup: References

The References dialogs and options are quite different in a normal I/O mode, when compared to an Acoustic I/O mode. References in normal mode are discussed first.

The References in acoustic mode (when an Acoustic checkbox checked in **Signal Path Setup > Input/ Output**) are discussed beginning on page 56.

References: (Normal Mode)



A signal path through a DUT will have an optimal operating level for each measurement. This level may be specified in a device's specifications, required for a standard measurement method, or may be determined on the bench by a monitoring a parameter such as distortion.

If, for example, you select a non-referenced unit such as dBV as the unit in a meter display, you will see decibels relative to 1 volt; but if you select the reference unit dBrA, you will see decibels relative to the reference value you have entered.

Output References

If you would like to force all generator settings to acoustic units referenced to an acoustic level, see Acoustic Output configuration on page 56.

dBrG:

You can save a generator output reference level in each domain for a signal path. The saved generator level is called dBrG, and can be referenced throughout the signal path when making generator settings by selecting dBrG from the units menu.

You can enter a known value as dBrG here, or you can use the Auto Gen Level feature (see below) to find the optimal generator level setting for your test.

Alternatively, you can determine dBrG manually by adjusting the generator level in Verify Connections while observing the measured level or measured distortion on the graph, until you achieve your desired results. Once you have found your optimal level, you can copy the value and paste it in the dBrG field.

Default values are

- analog generator: 0 dBrG = 100.0 mVrms
- digital generator: 0 dBrG = –20 dBFS

Note: When using dBrG as a unit in a measurement, a setting greater than 0 dBrG is not allowed.

dBm:

Enter a dBm output reference impedance here. dBm is available as a level unit throughout APx. dBm is a power unit, and as such must be referenced to a nominal circuit impedance to be meaningful. dBm is typically referenced to 600 Ω. You can enter a different value here.

W (watts):

Enter a W (watts) output reference impedance here. W (watts) is available as a level unit throughout APx. W (watts) is a power unit, and as such must be referenced to a nominal circuit impedance to be meaningful. By default, W (watts) is referenced to 8.000 Ω. You can enter a different value here.

Auto Gen Level

The Auto Gen Level... button opens a dialog where you can automatically set the dBrG by regulating to a target level or distortion. You can read more about the Automatically Set Generator Level dialog at the end of this chapter, on page 62.

Input References

If you would like to force all results to acoustic units referenced to an acoustic level, see Acoustic Input configuration on page 56.

dBrA and dBrB:

You can save two analog input reference levels and two digital input reference levels for a signal path, so that results can be evaluated relative to references. These saved input level references are called dBrA and dBrB, and are independently set for analog and digital inputs. These can be referenced throughout the signal path when making result settings by selecting dBrA or dBrB from the units menu. You can set dBrA and dBrB arbitrarily, or you can set them interactively with your DUT, while observing measured level or distortion.

Reference Offset

Once a reference has been entered or set, you can offset the level by some arbitrary amount.

Set dBr

This button opens the Set dBrA, B dialog, where you can interactively set dBrA or dBrB from input levels. You can read more about the Set dBrA, B dialog at the end of this chapter, on page 63.

dBSPL1 and dBSPL2

If you are making acoustic measurements, you may want to use dBSPL references instead of dBr references.

If you would like to force all results to acoustic units referenced to an acoustic level, see Acoustic Input configuration on page 56.

You can save two analog input reference levels and two digital input reference levels for a signal path using sound pressure level (SPL) units, so that results can be evaluated relative to acoustic levels.

These saved input acoustic level references are called dBSPL1 and dBSPL2, and are independently set for analog and digital inputs. These can be referenced throughout the signal path when making result settings by selecting dBSPL1 or dBSPL2 from the units menu. You can set dBSPL1 and dBSPL2 arbitrarily, or you can set them interactively with your DUT, while observing measured level.

Typically, **dBSPL** is referenced to a measurement microphone and a microphone calibrator, in APx External Source configuration.

Calibrator Level

Microphone calibrators operate at one or more specific acoustic levels, typically 94 dBSPL or 114 dBSPL. Enter the calibrator level here.

Mic Cal / dBSPL

This button opens the **Microphone Calibration /Set dBSPL1, dBSPL2** dialog, where you can interactively set dBSPL1 or 2 from input levels. You can read more about this dialog at the end of this chapter, on page 64.

dBm:

Enter a dBm input reference impedance here. The default is 600.0 Ω.

W (watts):

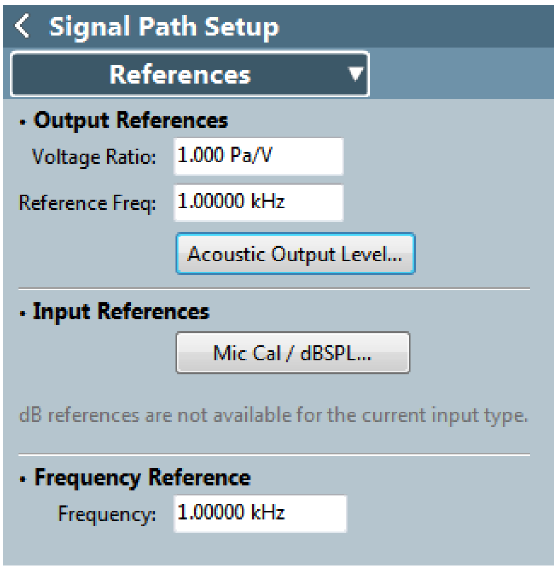
Enter a watts input reference impedance here. The default is 8.000 Ω.

Frequency Reference

Frequency:

Enter a global reference frequency here. The default is 1.0000 kHz.

Signal Path Setup: References (Acoustic Modes)



Output References (Acoustic)

When the **Acoustic** checkbox in Signal Path Setup: Output Configuration is checked, these Output References selections are available. Checking **Acoustic** expresses the generator levels in acoustic units throughout the signal path.

Voltage Ratio

Voltage Ratio displays the ratio of acoustic level in Pa to electrical level in V, which is established in the Set Acoustic Output Level dialog.

Alternatively, you can enter a known voltage ratio directly in the Voltage Ratio field.

Reference Freq

Reference Freq displays the frequency at which the voltage ratio was established, which is established in the Set Acoustic Output Level dialog.

Alternatively, you can enter a known reference frequency directly in the Reference Freq field.

Acoustic Output Level...

Click this button to open the **Set Acoustic Output Level** dialog. You can read more about this dialog at the end of this chapter, on page 65.

Input References (Acoustic)

When the **Acoustic** checkbox in Signal Path Setup: Input Configuration is checked, this Input References selection is available. Checking **Acoustic** expresses the analyzer levels in acoustic units throughout the signal path.

Mic Cal / dBSPL...

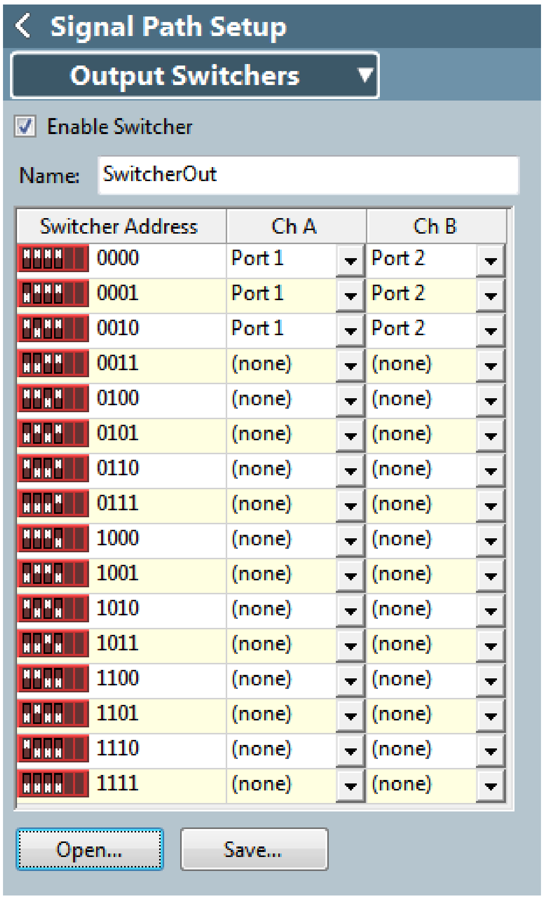
Click this button to open the **Microphone Calibration / Set dBSPL Per-channel** dialog. You can read more about this dialog at the end of this chapter, on page 65.

Frequency Reference

Frequency:

Enter a global reference frequency here. The default is 1.0000 kHz.

Signal Path Setup: Output and Input Switchers

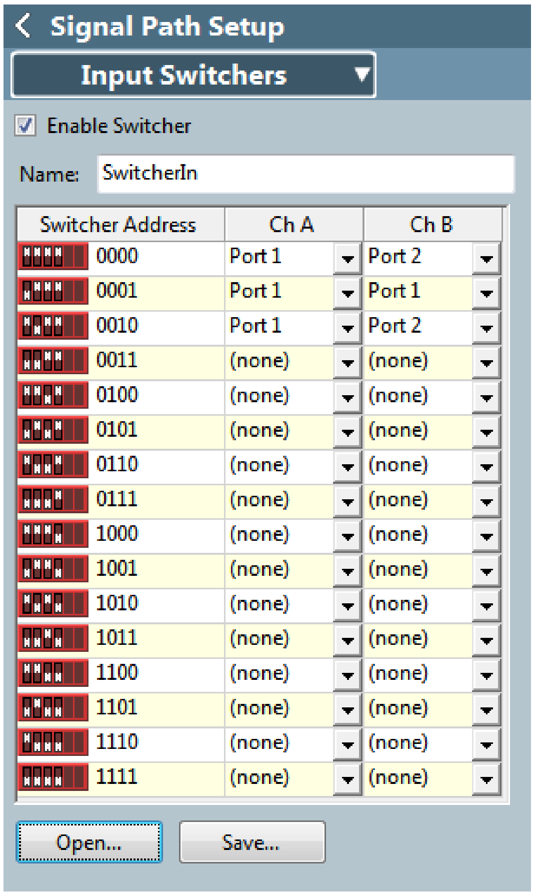


Audio Precision manufactures optional switchers that can be used to connect or disconnect many signal channels (up to 192 input and 192 output channels) to or from the instrument, under software control. See general switcher operating information beginning with the Switchers topic.

To configure a switcher, go to Signal Path Setup and choose Output Switchers or Input Switchers.

Name

You can apply a custom name to a switcher setup in the Name field. This name becomes the default name in the Save dialog. When opening a file, the name of the file opened file appears in the Name field.



Switcher Address

The switcher address is the binary address that enables communication from the APx500 software to a particular switcher hardware unit. On the rear panel of each switcher is a DIP switch with 6 switches; the four switches to the left set the binary address for that unit. To address a switcher unit from the APx500 software, the binary address must match on the switcher hardware and the switcher address controls in the software.

Note that the DIP switch orientation on the hardware is such that switch order is reversed from the digit order in ordinary binary notation, and that 0=UP and 1=DOWN: binary number 0111 corresponds to switch settings of DOWN-DOWN-DOWN-UP. For clarity, both DIP switch position and binary number are shown in the control panel.

Switcher Channels A and B

Each switcher has 2 channels called Channel A and Channel B, that are used to connect to 2 APx outputs (for an output switcher) or inputs (for an input switcher). These are connected to the switcher ports by internal relays, under the APx500 software control.

Switcher Ports

Each switcher has 12 ports that are used to connect to DUT channels according to your test requirements. An input switcher's ports are designed to connect to a DUT's output channels, and an output switcher's ports are designed to connect to a DUT's input channels. For each APx500 signal path, you can make a port selection for Channel A and Channel B for each addressed switcher.

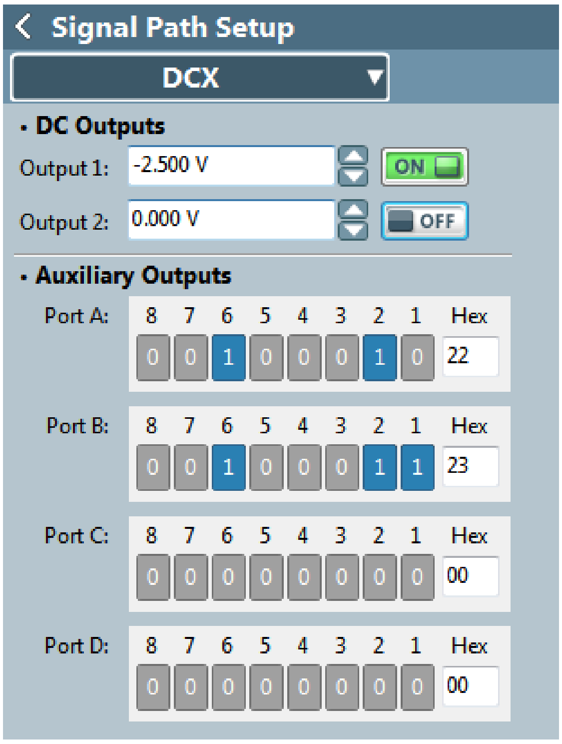
All But ChA

The Output Switcher Channel B port selection list includes All But ChA as a choice. This selection connects switcher Channel B to all ports except the current port connected to Channel A.

Audio Precision manufactures optional switchers that can be used to connect or disconnect many signal channels (up to 192 input and 192 output channels) to the instrument inputs or outputs, under software control.

See Chapter 97 for information about using switchers with APx500 and the APx family of instruments.

Signal Path Setup: DCX



The DCX-127 Multifunction Module is an Audio Precision accessory that provides interface and control features not otherwise available in Audio Precision analyzers. Download the *DCX-127 Multifunction Module User's Guide* from the Audio Precision Web site at ap.com for complete operational information for the DCX-127.

- In Bench Mode, you can configure the DCX DMM (digital multimeter), the DCX DC Outputs and Auxiliary Outputs from the Signal Path Setup > DCX panel.
- In Sequence Mode, you can only configure the DCX DC Outputs and Auxiliary Outputs from the Signal Path Setup > DCX panel. However, DMM functions are available in DC Level (DCX) and Resistance (DCX) Sequence Mode measurements.

DMM (Bench Mode only)

The DCX has a digital multimeter (DMM) that can be configured as DC voltmeter or an ohmmeter.

Mode

The Mode control allows you to select the DCX meter mode.

- Off
  The DCX meter is not enabled.
- Resistance
  The DCX meter is configured to measure resistance.
- DC Level
  The DCX meter is configured to measure DC Level.

Range

The Range control allows you to select one of 5 fixed maximum meter input ranges, or Auto. The ranges are:

Resistance	***DC Level***
2 MOhm	500 V
200 kOhm	200 V
20 kOhm	20 V
2 kOhm	2 V
200 Ohm	200 mV

Rate

The Rate control allows you to select one of two meter reading rates.

- 6/sec
  Approximately 6 meter readings are taken per second.
- 25/sec
  Approximately 25 meter readings are taken per second.

At 6/sec, DCX meter resolution is a full 4 1/2 digits. At 25/sec, the resolution is reduced. The number of digits displayed is the same but the least significant digit is always 5 or 0.

DC Outputs (Sequence Mode and Bench Mode)

The DCX has two DC outputs, which you can configure here.

- Output 1
  Set the DC Output 1 voltage and ON/OFF status here.
- Output 2
  Set the DC Output 2 voltage and ON/OFF status here.

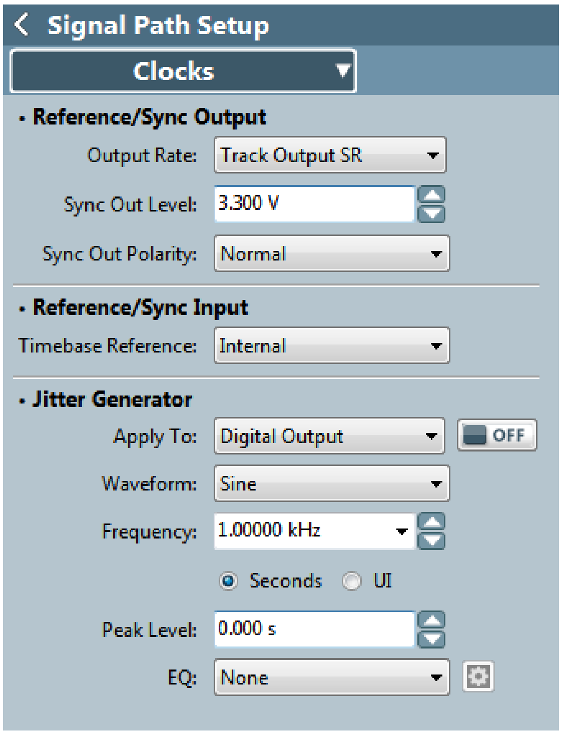
The voltage range for both outputs is ±10.50 V. These outputs have relatively low current capability (20 mA) and should be used as control or signal voltages, not as supply voltages.

Auxiliary Outputs (Sequence Mode and Bench Mode)

The DCX has four auxiliary (GPIO) output ports, similar in function, capability and user interface to the APx Aux Control Out port.

Configure the output ports here.

Signal Path Setup: Clocks



The features and controls described here require the Advanced Master Clock, a standard feature on the APx555 and an option on modular APx analyzers. When the Advanced Master Clock is present, these inputs and outputs are available on connectors on the APx rear panel.

Overview

Signal Path Setup: Clocks is the control panel for selecting the reference for the APx master clock time base, for making clock reference input and output settings, and for applying an internally generated jitter signal to APx digital outputs.

Reference/Sync Output

Output Rate:

- **Track Output SR**
  When **Track Output SR** is set, the Reference and Sync outputs track the current APx output sample rate.
  When the APx output is digital, the Reference and Sync output rates are set to the current output sample rate, as set in Signal Path Setup: Output Configuration.
  When the APx generator is playing a waveform, the Reference and Sync output rates are set to the sample rate of the waveform.
  When the APx output is the DAC generator, the Reference and Sync output rates are fixed at 192 kHz, unless an arbitrary Generator Waveform of a different rate is loaded. When the APx output is the High Performance Sine Generator, the Reference and Sync outputs are muted.
- **Custom**
  When **Custom** is set, the **Custom Rate** field is available. The rate set in this field will be the **DARS Reference** and **Sync Output Rates**, which track together from 8 kHz to 216 kHz. If you enter a rate between 216 kHz and 50 MHz, the **Sync Output** will use that rate, but the **DARS Reference Output** will mute.
- **Custom Rate**
  Enter the **Custom Rate** here. The acceptable range is 8 kHz to 50 MHz. The **DARS Reference Output** range is restricted to a maximum frequency of 216 kHz.

Sync Out Level

Set the Sync Out level here. Minimum is 0.8 V; maximum is 3.6 V; default is 3.3 V.

Sync Out Polarity

Select **Normal** or **Inverted**.

Reference/Sync Input

Timebase Reference:

- **Internal**
  The APx timebase is referenced to an internal crystal oscillator. Signals at the Reference In and Sync In connectors are ignored.

- Ext. AES11 DARS (XLR)
  The APx timebase is referenced to the digital signal at the AES11 DARS In XLR jack on the rear panel.
- Ext. Sync (BNC)
  The APx timebase is referenced to the square wave sync signal at the Sync In BNC jack on the rear panel.

Ext. Reference Rate:

When using an external Timebase Reference, you must specify the nominal rate of the external signal.

For the DARS input, the range is 8 kHz to 216 kHz. For the Sync input, the range is 4 kHz to 49.1520 MHz.

DARS Termination:

The DARS signal is a special version of an AES3 (IEC60958-4) signal. AES3 signals should be terminated in an impedance of 110 ohms, or bridged by a high impedance.

- High Impedance
  This setting is useful when bridging a digital connection (as with a “Y” cable). A balanced digital connection should be properly terminated in 110 ohms at the end of its run.
- 110 ohms
  With this set, the DARS Reference input is terminated in a resistance of 110 ohms. This is the correct setting for terminating a balanced digital audio connection for AES3 / AES-EBU / IEC60958-4 professional formats.

Jitter Generator

Jitter is the variation in time of an event, such as a clock signal, from nominal. In APx analyzers, jitter generation is only available for I/O modules that support jitter, and only when the Advanced Master Clock is installed. An APx digital or sync output signal that is impaired by jitter can be used to evaluated jitter tolerance in downstream or synchronized external devices.

A “jitter clock” is provided as a component of the Advanced Master Clock system. This clock can be modulated in time (jittered), and applied as a reference to either a digital output or to the signals provided to the Reference Out and Sync Out connections. The modulating signal can be one of three waveforms, across a range of frequencies and levels.

Apply To

Select the signal to be impaired by jitter.

- Digital Output
  This selection will apply jitter to the output of a digital module that supports output jitter, such as the Advanced Digital I/O module.
- Ref/Sync
  This selection will apply jitter to the Reference Out and Sync Out signals available on the APx rear panel.

On/Off

Turn the jitter On or Off.

Waveform

Select a waveform for the jitter generator.

- Sine
- Square
- Noise

Frequency

For the sine and square waveforms, select a frequency. The range is 2 Hz to 200 kHz.

Peak Level

Set the peak level of the jitter here. Since jitter is a time modulation, the level units are seconds. For the noise signal, the maximum peak level is 1.592 µs. For the sine and square waves, the maximum peak level varies with frequency, providing 1.591 µs (9.775 UI at 48 kHz) peak level up to 20 kHz. Above 20 kHz, the maximum level derates at 20 dB/decade, falling to 159.1 ns by 200 kHz.

Units

Since jitter is a time modulation, the level units are seconds.

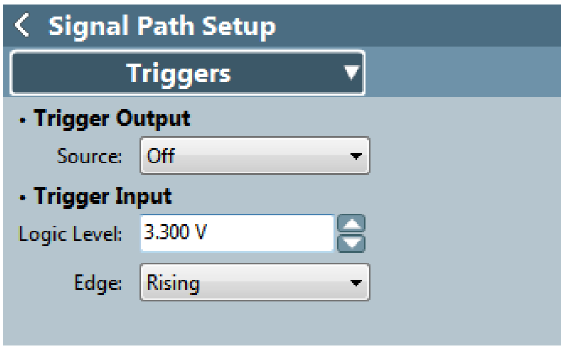
- (sec)
  Choose this to show jitter in seconds.
- (UI)
  Choose this to show jitter in Unit Intervals (UI), a unit that scales to the sample rate.

EQ

When the jitter waveform is sine, an EQ curve can be applied to the jitter generator. You can use this function to apply an EQ curve that reflects the AES3 jitter tolerance template for the current sample rate, or for any other EQ curve. The EQ curve affects the jitter generator level for both sweep and static measurements.

- None
- Create New
  You can create or edit the generator EQ curve in Editing the Generator EQ table.
- Browse for File
  Open an existing jitter generator EQ table (*.csv or *.xls file).

Signal Path Setup: Triggers



Overview

External trigger inputs and outputs are available when the Advanced Master Clock (AMC) is installed, and are available on connectors on the APx rear panel.

Trigger Output

Trigger Output sends a logic-level pulse when a certain condition is satisfied in APx. You can choose one of three trigger sources.

Source

Choose Audio Generator, High Performance Sine Generator, Jitter Generator or Off.

Level

When a Source is selected, the Level field is available. Set the trigger out level here. Minimum is 0.8 V; maximum is 3.6 V; default is 3.3 V.

Output triggering points by trigger source:

Trigger Source	*Trigger goes high at*	*Trigger goes low at*
High Performance Sine Generator	positive sine peak	negative sine peak
Burst sine	At start of high-level burst	At end of high-level burst
Jitter Generator (sine, square)	positive-going zero crossing	negative-going zero crossing
Jitter Generator (noise)	---	---
Sine (DAC)	positive-going zero crossing	negative-going zero crossing
Sine plus offset (DAC)	positive-going zero crossing of sine	negative-going zero crossing of sine
Sine split frequency (DAC)	positive-going zero crossing of freq A	negative-going zero crossing of freq A
Sine split phase (DAC)	positive-going zero crossing of freq A	negative-going zero crossing of freq A
Square	positive-going zero crossing	negative-going zero crossing
IMD (DAC)	positive-going zero crossing of freq 1	negative-going zero crossing of freq 1
DIM	positive-going zero crossing of square	negative-going zero crossing of square
Arbitrary/coded waveform	First sample	Second sample
Noise, all types	---	---
Bit test	First sample of sequence	Second sample of sequence
Constant value	---	---
Walking ones/zeros	At LSB/MSB transition	At next bit transition

Trigger Input

The Sequence Mode Signal Analyzer and the Bench Mode FFT measurement each have a Trigger choice called External Trigger. When this is selected, an external pulse at the rear-panel Trigger In connector satisfying the Logic Level and Edge requirements (below) will start the measurement.

Logic Level

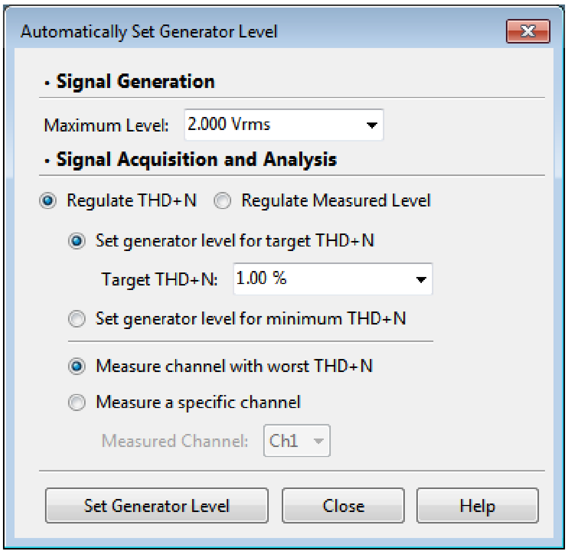
Set the expected logic level (logic family) here. The trigger threshold is 1/2 the value entered here. Minimum is 0.8 V; maximum is 3.6 V; default is 3.3 V.

Edge

Choose Rising edge or Falling edge for trigger input sensitivity.

References Dialogs Detail

Automatically Set Generator Level... dialog



Auto Gen Level uses generator regulation to automatically find the optimum testing level, regulating to your selected Target THD+N or Target Measured Level. Read more about regulation in beginning on page 433.

Note: The Auto Gen Level feature is not available when APx500 is configured for External Source.

From the References panel, in the Output References area, click Auto Gen Level... to open the Automatically Set Generator Level dialog. Your DUT should be connected and set up testing.

Signal Generation

Set the **Maximum Level** in the Signal Generation area. This limits the APx generator output to the selected value to protect your DUT.

Signal Acquisition and Analysis

Choose a regulation method:

- **Target THD+N** or
- Target Measured Level

Target THD+N

Target THD+N varies the generator level in search of a specified degree of distortion. You can specify a value, or have the regulation search for a minimum. If you have chosen Target THD+N, choose

- **Set generator level for target THD+N** or
- Set generator level for minimum THD+N

If you have chosen to search for a target THD+N, enter a value in the Target THD+N field.

You must also specify which channel to measure for regulation. Choose

- **Measure channel with worst THD+N** or
- Measure a specific channel.

If you have chosen to measure a specific channel, select the channel from the **Measured Channel** list.

Note: Distortion measurements such as THD+N benefit from filtering. Use the filters in Signal Path Setup > Input/Output to filter the signal used for regulation. Suggested filters are High-pass: 20 Hz, Low-pass: 22 kHz. Remember to reset the filters when you continue to other measurements.

Target Measured Level

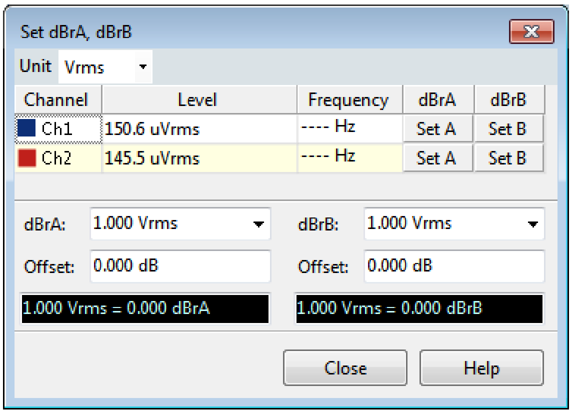
Target Measured Level varies the generator level in search of a specified level. If you have chosen Target Measured Level, enter a value in the **Target Level** field. Select the channel to be measured from the **Measured Channel** list.

Set Generator Level

Finally, click **Set Generator** Level. The generator is automatically adjusted while the analyzer observes the Measured Channel to find the Target THD+N or the Target Level. When this procedure is successfully completed, the generator level is copied to the dBrG field.

Set dBrA, B dialog

The **Set dBr A, B** dialog allows you to interactively set dBrA or B from input levels.



You can save two analog input reference levels and two digital input reference levels for each signal path. These saved input level references are called **dBrA** and **dBrB**, and are independently set for analog and digital inputs. They can be referenced throughout the signal path so that results can be evaluated relative to references. You can set dBrA and dBrB arbitrarily, or you can set them in interaction with your DUT, while observing measured level or distortion.

If you are making acoustic measurements, you may want to use **dBSPL** references (see page 64). For acoustic measurements using multiple microphones, it may be more useful to set sound pressure level references specific to each microphone and input channel. See Microphone Calibration / Set dBSPL1, dBSPL2 on page 65.

Setting dBrA and dBrB

If you have not yet set up your test, first go to Signal Path Setup to configure your inputs and outputs. See page 52.

Using the generator

In Sequence Mode, use the generator in Verify Connections and the RMS Level or THD+N ratio graphs attached to Signal Path Setup. In Bench Mode, use the Generator and the RMS Level or THD+N Ratio monitor meters.

With your DUT connected and adjusted for testing,

- Click the **Set dBr...** button in the References panel. This opens the **Set dBrA, dBrB** dialog, shown here.
- Turn the Generator **ON**, using a sine waveform.
- Set the Generator and the DUT level controls to produce a DUT output level appropriate to your test. This may involve first using **Auto Gen Level** (see page 62) to set an optimal generator output level, or it may be that you need to make your mea-

surements relative to some standard DUT output level, such as 1 volt rms.

- Use **RMS Level** or **THD+N Ratio** as the graph display (Bench Mode: monitor meter), depending upon which result you want to observe. If you are setting input levels in relation to a specific distortion figure, see the Distortion Target paragraph below.
- Choose a channel that exhibits the level or distortion you want as a reference, and click **Set A** or **Set B** to enter that value as a reference level.
- By default, the level you select will be referenced to the value of 0.000 dBrA (or dBrB). You can redefine how you reference the selected level by entering a value other than 0.000 in the **Offset** field. The display shows the result of your choices.

Using External Source

Setting input reference levels from an external source is slightly different from the operation described above, which uses the internal generator. First, be sure you are configured for External Source. Go to Signal Path Setup and select **None (External)** for the output connector in Output Configuration.

When using an external source, instead of using the generator you will play an audio signal from the DUT.

For setting references, you can use any mid-band audio sine wave signal.

Once the external signal is ON, setting an input reference is the same as using the generator: the RMS Level is shown on the meter bar display, and you can click the Set A or Set B buttons in the Set dBrA, dBrB dialog when your target level or THD+N reading is reached.

Read more about External Source in Chapter 19.

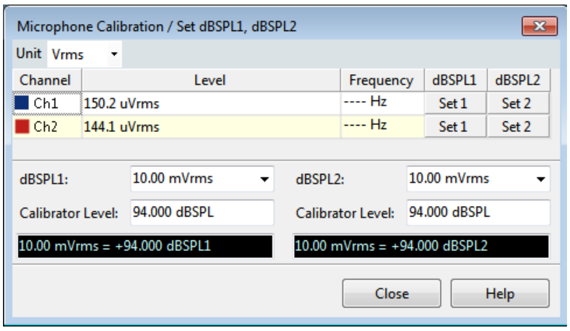
Using a Distortion Target to help set a reference level

If you would like to define your level reference by specifying a maximum acceptable distortion rather than a particular level, select the THD+N Ratio graph (Bench Mode: meter) to read the measured THD+N for the chosen channel.

To help you determine if the distortion exceeds your specification, you can set a limit on the THD+N display. In Bench Mode, set a limit in the usual way. In Sequence Mode, you can enter a value in the **Target THD+N:** field on the graph.

When the measured distortion for a channel exceeds this value the result is flagged in red. The distortion measurements can be filtered using the filters available in the **Signal Path Setup > Input / Output**. See **Signal Path Setup > Filters** on page 545 for more information.

Microphone Calibration /Set dBSPL1, dBSPL2 dialog



Typically, dBSPL is referenced to a measurement microphone and a microphone calibrator, in APx External Source configuration, without using the APx generator.

With your measurement microphone (and preamplifier, if required) connected and adjusted for testing,

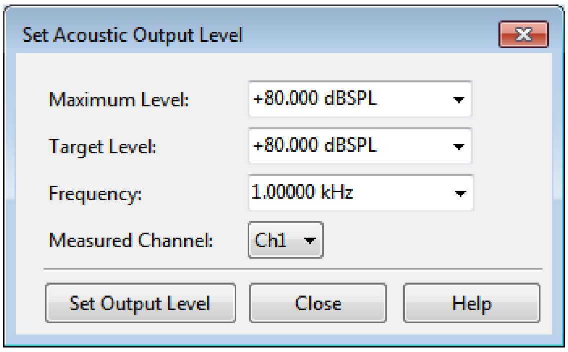
1. Set up the measurement microphone and microphone calibrator following the directions included with the microphone and calibrator. Turn on the microphone and connect it to the analyzer input.
2. Note the calibrator acoustic output level. Depending upon the calibrator, this might be fixed at one level, or switchable. 94 dBSPL and 114 dBSPL are two common calibrator acoustic output levels.
3. Navigate to the References panel.
4. Click the **Set dBSPL...** button on the References panel. This opens the **Set dBSPL1, dBSPL2 / Microphone Calibration** dialog.
5. Enter the nominal calibrator acoustic output level into the Calibrator Level field. By default, this is 94 dBSPL.
6. Mount the calibrator on the microphone connected to Channel 1 and turn it on.
7. Click the **Set 1** or **Set 2** button for Channel 1. You can set either reference, dBSPL1 or dBSPL2 from this value.
8. If you have a second measurement microphone you would like to reference, move the calibrator to the second microphone.
9. Click the **Set 1** or **Set 2** button for Channel 1. You can set either reference, dBSPL1 or dBSPL2 from this value.

Setting dBSPL without a calibrator

If a calibrator is not available, you can enter nominal references in the dBSPL1 and dBSPL2 fields.

The calibrator frequency is measured and displayed in the **Frequency** field. The mic signal is bandpass filtered at the measured calibrator frequency to reduce extraneous low-frequency noise.

Set Acoustic Output Level dialog.



This topic discusses setting acoustic output references when in Acoustic Output configuration. See page 56 for more information about Acoustic Output.

Since the analyzer outputs are voltage outputs, an acoustic output signal path must have scaling factor associated with it. This is called the **Voltage Ratio**, and it is expressed in Pa/V. The Voltage Ratio is set interactively here in the **Set Acoustic Output Level** dialog, and is displayed in the References (acoustics) dialog.

Alternatively, you can enter a known voltage ratio and reference frequency directly in the Voltage Ratio and Reference Freq fields in the References (acoustics) dialog.

Setting the Acoustic Output Level and obtaining Voltage Ratio

For this topic to be relevant, you must go to Signal Path Setup > Output Configuration and set a check in the Acoustic checkbox to enter Acoustic Output configuration.

To make this setting, you must have an audio power amplifier and a loudspeaker facing into a calibrated measurement microphone at a standard distance. To set a target level in dBSPL or Pa units, the Signal Path must also be in Acoustic input configuration. For an accurate dBSPL or Pa reading, you must first calibrate the measurement microphone.

Go to Signal Path Setup > Input/Output and check the Acoustic checkbox in Output Configuration. You must be set to an analog output to see the Acoustic checkbox. In Signal Path Setup > Output Configuration or References > Output References, click **Set Acoustic Output Level...** to open the dialog shown above:

You must have a power amplifier and loudspeaker connected to an APx output, and a measurement microphone facing the loudspeaker connected to an APx input. Click Set Output Level. APx uses generator regulation to automatically find the optimum testing level, regulating to your selected Target Level.

Maximum Level:

Maximum Level limits the APx generator output to the selected value to protect your amplifier and loudspeaker.

Target Level:

Target Level is the sound pressure level you would like to attain. For this to be valid, your microphone must be calibrated and set as an APx Acoustic input reference.

Frequency:

Frequency sets the frequency of the stimulus signal. By default, it is 1 kHz. Lower or higher frequencies might be more appropriate for various driver or loudspeaker types, or when Output Equalization is applied.

Measured Channel:

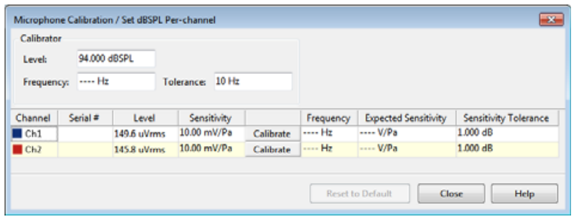
Set the channel to which the measurement microphone is connected here.

Units

Units available for Acoustic Output and Acoustic Input configuration are

- Pa
- dBSPL

Microphone Calibration / Set dBSPL Per-channel dialog



This topic discusses microphone calibration and setting dBSPL input references when in Acoustic Input configuration.

Since the analyzer inputs are voltage inputs, an acoustic input signal path must have scaling factor associated with it. This is the **Sensitivity** input reference, expressed in V/Pa. Sensitivity is set interactively (or entered as a known value from the microphone specification) here in the **Mic Cal / dBSPL** dialog, and is displayed in the References (acoustics) dialog.

Microphone Calibration / Set dBSPL Per-channel

For this topic to be relevant, you must go to Signal Path Setup > Input Configuration and set a check in the Acoustic checkbox to enter Acoustic Input configuration.

In Acoustic Input configuration, you can set input reference levels in acoustic units (dBSPL or Pa) independently for each input channel. This allows you to use multiple microphones in an acoustic measurement, where each microphone has been independently referenced to a microphone calibrator.

Setting dBSPL without a calibrator (entering a known value)

If performing an acoustic calibration of the measurement microphone is not possible, you can enter the microphone's known sensitivity (from a specification or from a previous calibration) in the Sensitivity field for a reference setting.

Setting dBSPL with a microphone calibrator

Calibration of a measurement microphone in an acoustic signal path requires a measurement microphone and a microphone calibrator.

The calibrator frequency is measured and displayed in the Frequency field. The mic signal is tightly bandpass filtered at the measured calibrator frequency to reduce extraneous noise.

1 Set up the measurement microphone(s), preamplifiers and acoustic calibrator following the directions included with the microphones and the calibrator. Turn on the microphones and preamps. Connect the preamp outputs to APx inputs.
2. Note the calibrator acoustic output level. Depending upon the calibrator hardware, this might be fixed at one level, or switchable. 94 dBSPL and 114 dBSPL are two common calibrator acoustic output levels.
3. In APx, go to Signal Path Setup > Input/Output and check the Acoustic checkbox in Input Configuration. You must be set to an analog input to see the Acoustic checkbox.
4. Go to Signal Path Setup > References and click the Mic Cal / dBSPL... button in Input References. This opens the Microphone Calibration / Set dBSPL Per-channel dialog.
5. Enter the nominal calibrator acoustic output level (check the calibrator instructions) into the Calibrator Level field. By default, this is 94 dBSPL.
6. Optional: Enter the nominal calibrator output frequency into the Calibrator Frequency field. Typical calibrator frequencies are 250 Hz or 1 kHz.
7. Optional: Set an acceptable tolerance for frequency in the Tolerance field.
8. Optional: Set the nominal microphone sensitivity in the Expected Sensitivity field.
9. Optional: Set an acceptable tolerance for sensitivity in the Sensitivity Tolerance field.
10. Optional: Enter a serial number or other identifying information in the Serial Number field.
11. Mount the calibrator on the first microphone and turn it on.
12. Click the Calibrate button for Channel 1.
13. Move the calibrator to the second microphone.
14. Click the Calibrate button for Channel 2.
15. Repeat this procedure until a reference level has been set for each microphone.

If you have entered nominal frequency and expected sensitivity values, calibration results that fall within the tolerances you have set will be accepted without comment. Values outside those ranges will be marked in red.

Reset to Default

This button sets the Calibrator Frequency, Calibrator Tolerance, Expected Sensitivity and Sensitivity Tolerance fields to their default settings.

Units

Units available for setting Reference Levels in Acoustic input are

- dBSPL
- Pa

7

Verify Connections
(Sequence Mode)

Verify Connections

It's easy to accidentally cross cables when preparing a test setup. Verify Connections allows you to check your connections by sending and receiving tones on the channels you are using.

The **Verify Connections** panel is at the bottom of the **Signal Path Setup** column, providing diagnosis features for signal interconnection and format in the associated displays at the right side of the Workspace. The **Verify Connections** panel can be hidden.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 6 for information about Signal Path Setup.

Generator Off/On

With all your connections made, click Generator On. APx500 will apply the waveform to the selected **Test Channel** (by default, all channels are selected).

Waveform

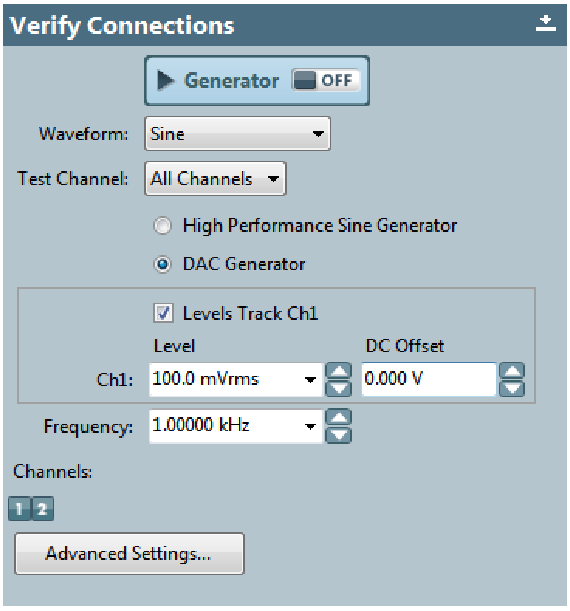
See Chapter 5 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

Test Channel

From the dropdown menu, use **All Channels** or select an individual channel to check for cross-connection.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 49 for a comparison of features.



Channels

Assign the generator to output channels. By default, all active output channels are selected.

Verifying your Connections

With all your connections made, click **Generator On.** APx500 will apply the waveform to the selected channel (by default, **All Channels** are selected).

Use the Monitors

The Monitors are always available, giving you a simultaneous view of Scope, Spectrum, Meter or Metadata displays of the input signal(s).

Check RMS Level

View the RMS Level results. From the dropdown menu, use **All Channels** or select an individual channel to check for cross-connection.

Check Gain

View the Gain result to evaluate the DUT gain.

Check THD+N Ratio

Click the THD+N Ratio result to view the signal distortion.

Bits and Error Rate: Diagnostics for Digital Signal Paths

For signal paths with both a digital output and input, you can apply special waveforms and measurements as diagnostic tests for digital signal paths. These tools can be helpful in diagnosing digital connection problems such as "stuck bits." When the Output Configuration is digital, the Verify Connections Generator will show:

- Bit test
- Walking Zeros
- Walking Ones

  and
- Constant Value

as generator waveform choices. (You could also use a properly formed digital file using these signals as a generator waveform or external source).

Read more about these special waveforms beginning on page 48.

Choose the Bits or Error Rate results.

Bits Result (digital only)

This view shows a real-time display of the audio bitstream. The bits are numbered from the MSB (most significant bit, bit 24) to the LSB (least significant bit, bit 1). It can be used with any waveform, but is most useful with these special waveforms:

- Walking Zeros
- Walking Ones
- Constant Value

Active Bits

When Active Bits is set, any bit that changes state during the diagnostic period (about 1/4 second) is shown as highlighted with color. Inactive bits are shown as white.

Data Bits

When Data Bits is set, the logical state (binary 1 or binary 0) of each bit in the audio word is shown. This is updated at the end of each diagnostic period (about 1/4 second). Binary 1 is shown in highlighted color; binary 0 is shown as white.

Note: The audio word may be any length between 32 bits and 8 bits, depending upon the Bit Depth setting in Signal Path Setup.

Error Rate Result (digital only)

This view recognizes Audio Precision bit-accurate waveforms, including

- Bit Test
  (This is usually the best choice for digital error rate measurements.)
- Walking Zeros
- Walking Ones
- Constant Value

When one of these bit-accurate waveforms is received with every sample at precisely the same value as was generated, zero errors are reported and an error rate of 0.000% is displayed.

When one of these waveforms is received with sample values different from the values in the original waveform, each differing sample is reported as one error, and the rate is calculated.

Read more about bit-accurate measurements beginning on page 251.

8

SPS: APx515 analog I/O

Introduction

This chapter discusses the Signal Path Setup analog I/O configurations for the APx515 analyzer. For other APx515 I/O configurations, go to

- **DIO** (digital I/O) configurations, Chapter 12.
- **ASIO** configurations, Chapter 18. (This is a software option for the APx515. See Software Options on page 166.)
- **External Source** configuration (open-loop, using no generator output), Chapter 5.
- **File Input** configurations, Chapter 19.

Common Signal Path Setup I/O Settings

- **Acoustic** (output) mode and output **EQ** are common to all analyzer analog output configurations,
- **Acoustic** (input) mode is common to all analyzer analog input configurations.
- **Loopback**, **Channel Labels**, **Filters** and **Device Under Test: Delay** settings are common to all analyzer input configurations.

Each of these common features is discussed in the **Input/Output** topics in Chapter 6, Signal Path Setup.

Bench Mode

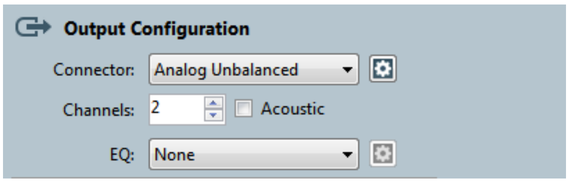
Bench Mode is an alternative operating interface for APx500, and is available as a software option for the APx515. See Software Options on page 166.

Output Configuration

The **Output Configuration** panel allows you to select the number of output channels operating for your test, and to choose the output format and connectors you will be using.

The **Settings** button opens an output **Settings** dialog for each configuration, offering more detailed control.

Output: Analog Unbalanced



This selects the unbalanced analog outputs available on BNC connectors on the instrument front panel.

Channels

The Channels setting allows you to set the number of output channels (1–2) to be tested for this signal path.

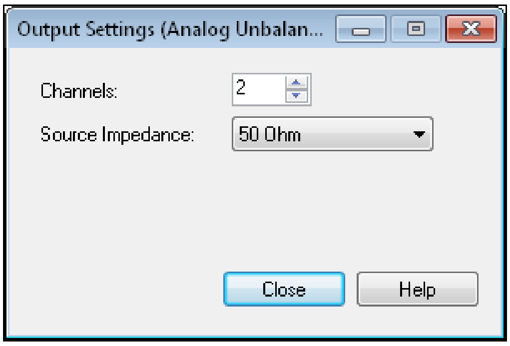
Acoustic

Read more about output **Acoustic** mode in the **Input/Output** topics in Chapter 6, Signal Path Setup.

EQ

Read more about output **EQ** in the **Input/Output** topics in Chapter 6, Signal Path Setup.

Settings for Analog Unbalanced Outputs



Channels

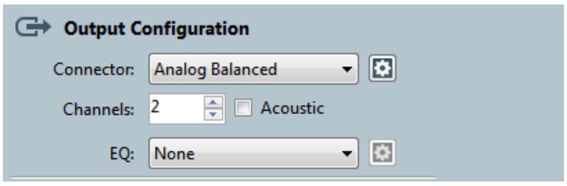
The **Channels** setting appears in both Signal Path Setup and here in the Settings dialog.

Source Impedance

Select the output **Source Impedance** here. For unbalanced outputs, the choices are

- 50 Ω (default)
- 600 Ω

Output: Analog Balanced



This selection enables the balanced analog outputs available on XLR3 male connectors.

Channels

The Channels setting allows you to set the number of output channels (1–2) to be tested for this signal path.

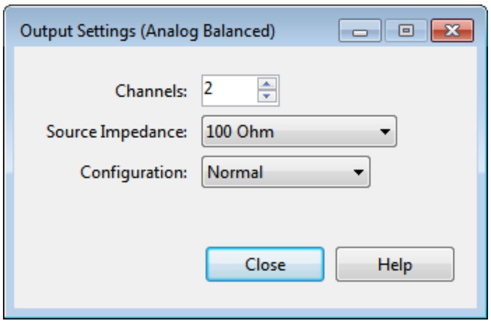
Acoustic

Read more about output **Acoustic** mode in the **Input/ Output** topics in Chapter 6, Signal Path Setup.

EQ

Read more about output **EQ** in the **Input/Output** topics in Chapter 6, Signal Path Setup.

Settings for Analog Balanced Outputs



Channels

The Channels setting appears in both Signal Path Setup and here in the Settings dialog.

Source Impedance

Select the output source impedance here. For balanced outputs, the choices are

- 100 Ω
- 600 Ω

Configuration

The Configuration control sets the **Normal Mode / Common Mode Test (CMTST)** configuration for the balanced analog outputs, for all measurements.

Configuration choices are:

- **Normal**
  The default, differentially balanced output configuration. Signal plus (+) is connected to pin 2; signal minus (–) is connected to pin 3. This is the recommended setting for all measurements.
- **CMTST**
  A fixed common-mode output configuration. Signal plus (+) is connected to both pin 2 and pin 3, signal minus (–) is connected to pin 1 (ground). This configuration is recommended only for common-mode testing.

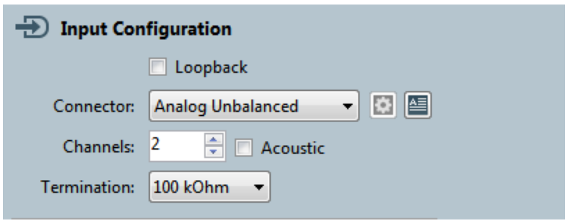
Setting a common mode configuration for all measurements can produce unexpected results. For Sequence Mode common mode rejection ratio tests, use the CMRR measurement instead.

Input Configuration

The Input Configuration panel allows you to select the number of input channels operating for your test, and to choose the input format and connectors you will be using.

The **Settings** button is unavailable for analog inputs.

Input: Analog Unbalanced



Connector

Analog Unbalanced sets both channels to the unbalanced analog inputs, available on BNC connectors on the instrument front panel.

Channels

The Channels setting appears in both Signal Path Setup and here in the Settings dialog.

Acoustic

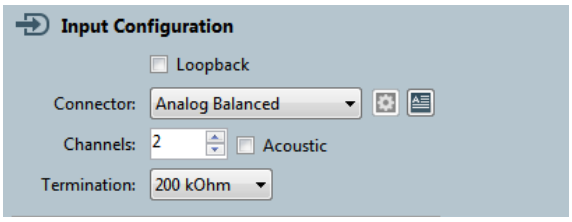
Read more about input **Acoustic** mode in the **Input/ Output** topics in Chapter 6, Signal Path Setup.

Termination

This control sets the input termination for both unbalanced input channels. Choices are:

- 100 kΩ or
- 600 Ω

Input: Analog Balanced



Connector

Analog Balanced sets both channels to the balanced analog inputs, available on XLR female connectors on the instrument front panel.

Channels

The Channels setting appears in both Signal Path Setup and here in the Settings dialog.

Acoustic

Read more about input **Acoustic** mode in the **Input/ Output** topics in Chapter 6, Signal Path Setup.

Termination

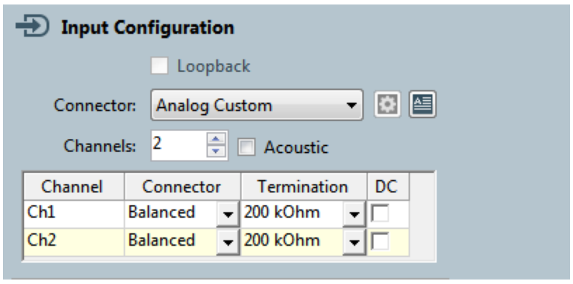
This control sets the input termination for both balanced input channels. Choices are:

- 200 kΩ or
- 600 Ω

Input: Analog Custom

Connector

Analog Custom allows you to set **Connector, Termination** and **Coupling** settings independently for each input channel. A channel set to **Unbalanced** is connected to a front panel BNC connector; a channel set to **Balanced** is connected to a front panel XLR female connector.



Channels

The Channels setting appears in both Signal Path Setup and here in the Settings dialog.

Acoustic

Read more about input **Acoustic** mode in the **Input/ Output** topics in Chapter 6, Signal Path Setup.

The Custom configuration grid

Channel

Each channel has a row for custom selections.

Connector

Select the Connector (Balanced or Unbalanced) for a particular channel here.

Termination

Select the Termination for a particular channel here. The Termination options will depend upon the Connector selection.

DC

Select the AC or DC coupling for a particular channel here. Note that when Input Configuration is set to **Custom**, the **High-pass Filter** selections are only AC-coupled. Selecting **DC** in the **Custom** grid overrides the **High-pass Filter** selection for the channel, and sets it to DC-coupled. See page 53 and also Chapter 91 for a detailed discussion of Signal Path Setup Filters.

Filters and Device Under Test: Delay

Filters

See page 53 and also Chapter 91 for a detailed discussion of Signal Path Setup **Filters.** Note that analog system bandwidth (DAC sample rate) is set using the **Low-Pass Filter** menu, and that AC/DC coupling is set using the **High-pass Filter** menu.

Device Under Test

Delay

See the Input/Output topics in Chapter 6, Signal Path Setup, for a detailed discussions of **DUT Delay.**

9

SPS: APx52x analog I/O and APx582 analog out

Introduction

This chapter discusses the Signal Path Setup analog I/O configurations for the APx520, 521, 525 and 526 analyzers. Additionally, the Output Configuration discussed here applies to the analog outputs of the APx582. For other I/O configurations for these analyzers, go to

- DIO (digital I/O) configurations, Chapter 12.
- ADIO (advanced digital I/O) configurations, Chapter 13. This an option that can replace the standard DIO.
- DSIO (digital serial I/O) configurations, Chapter 14.
- HDMI+ARC I/O configuration, Chapter 15.
- Bluetooth configurations, Chapter 16.
- PDM configuration, Chapter 17.
- ASIO configurations, Chapter 18.
- External Source configuration, Chapter 5.
- File Input configurations, Chapter 19.

Common Signal Path Setup I/O Settings

- Acoustic (output) mode and output EQ are common to all analyzer analog output configurations,
- Acoustic (input) mode is common to all analyzer analog input configurations.
- Loopback, Channel Labels, Filters and Device Under Test: Delay settings are common to all analyzer input configurations.

Each of these common features is discussed in the Input/Output topics in Chapter 6, Signal Path Setup.

Output Configuration

The Output Configuration panel allows you to select the number of output channels operating for your test, and to choose the output format and connectors you will be using.

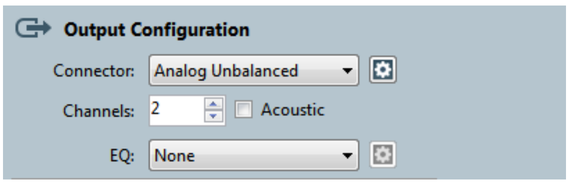
The Settings button opens an output Settings dialog for each configuration, offering more detailed control.

AG52 Analog Generator option

The AG52 Analog Generator option is available for APx525 family instruments. AG52 provides square wave and DIM waveforms and improved level and noise performance, but has no effect on the available analog output configuration selections.

The AG52 is a standard component for APx582 analyzers.

Output: Analog Unbalanced



This selects the unbalanced analog outputs available on BNC connectors on the instrument front panel.

Channels

The Channels setting allows you to set the number of output channels (1–2) to be tested for this signal path.

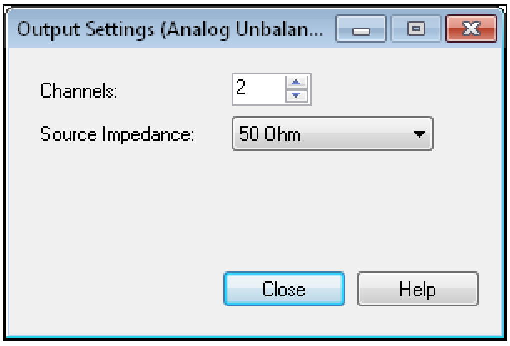
Acoustic

Read more about output Acoustic mode in the Input/Output topics in Chapter 6, Signal Path Setup.

EQ

Read more about output EQ in the Input/Output topics in Chapter 6, Signal Path Setup.

Settings for Analog Unbalanced Outputs



Channels

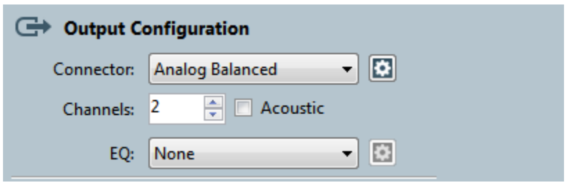
The **Channels** setting appears in both Signal Path Setup and here in the Settings dialog.

Source Impedance

Select the output **Source Impedance** here. For unbalanced outputs, the choices are

- 20 Ω (default)
- 50 Ω
- 75 Ω
- 100 Ω
- 600 Ω

Output: Analog Balanced



This selection enables the balanced analog outputs available on XLR3 male connectors and double banana connectors. The XLRs and the double bananas are wired in parallel.

Channels

The Channels setting allows you to set the number of output channels (1–2) to be tested for this signal path.

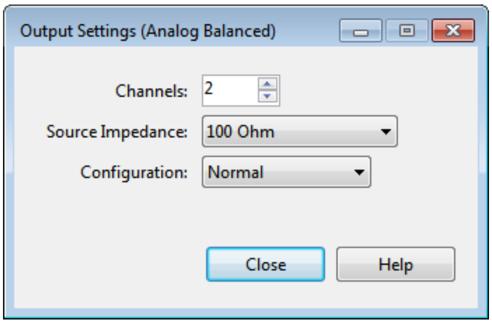
Acoustic

Read more about output **Acoustic** mode in the **Input/ Output** topics in Chapter 6, Signal Path Setup.

EQ

Read more about output **EQ** in the **Input/Output** topics in Chapter 6, Signal Path Setup.

Settings for Analog Balanced Outputs



Channels

The Channels setting appears in both Signal Path Setup and here in the Settings dialog.

Source Impedance

Select the output source impedance here. For balanced outputs, the choices are

- 40 Ω
- 100 Ω
- 150 Ω
- 200 Ω
- 600 Ω

Configuration

The Configuration control sets the **Normal Mode / Common Mode Test (CMTST)** configuration for the balanced analog outputs, for all measurements.

Configuration choices are:

- **Normal**
  The default, differentially balanced output configuration. Signal plus (+) is connected to pin 2; signal minus (–) is connected to pin 3. This is the recommended setting for all measurements.
- **CMTST**
  A fixed common-mode output configuration. Signal plus (+) is connected to both pin 2 and pin 3, signal minus (–) is connected to pin 1 (ground). This configuration is recommended only for common-mode testing.\CMTST (IEC pin 2)
  A fixed common-mode output configuration, using the IEC unbalanced leg configuration. Source impedance is fixed at 40 Ω. Signal plus (+) is connect to pin 2 and pin 3, with an additional 10 Ω resistor inserted in the pin 2 leg. Signal minus (–) is connected to pin 1 (ground).
- **CMTST (IEC pin 2)**
  A fixed common-mode output configuration, using the IEC unbalanced leg configuration. Source impedance is fixed at 40 Ω. Signal plus (+) is connect to pin 2 and pin 3, with an additional 10 Ω

resistor inserted in the pin 2 leg. Signal minus (–) is connected to pin 1 (ground).

- CMTST (IEC pin 3)
  A fixed common-mode output configuration, using the IEC unbalanced leg configuration. Source impedance is fixed at 40 Ω. Signal plus (+) is connect to pin 2 and pin 3, with an additional 10 Ω resistor inserted in the pin 3 leg. Signal minus (–) is connected to pin 1 (ground).

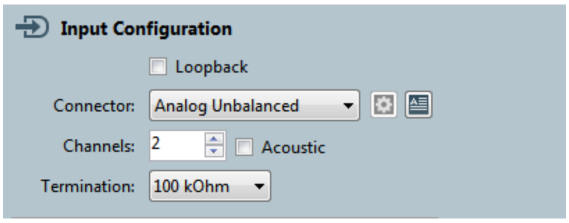
Setting a common mode configuration for all measurements can produce unexpected results. For Sequence Mode common mode rejection ratio tests, use the CMRR measurement instead.

Input Configuration

The Input Configuration panel allows you to select the number of input channels operating for your test, and to choose the input format and connectors you will be using.

The **Settings** button is unavailable for analog inputs.

Input: Analog Unbalanced



Connector

Analog Unbalanced sets both channels to the unbalanced analog inputs, available on BNC connectors on the instrument front panel.

Channels

The Channels setting appears in both Signal Path Setup and here in the Settings dialog.

Acoustic

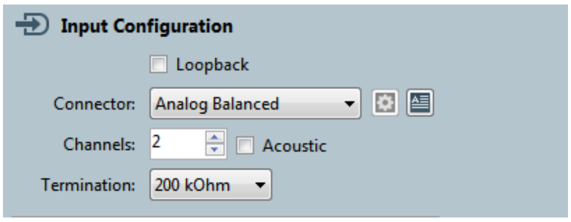
Read more about input **Acoustic** mode in the **Input/ Output** topics in Chapter 6, Signal Path Setup.

Termination

This control sets the input termination for both unbalanced input channels. Choices are:

- 100 kΩ
- 300 Ω
- 600 Ω

Input: Analog Balanced



Connector

Analog Balanced sets both channels to the balanced analog inputs, available on XLR female connectors on the instrument front panel.

Channels

The Channels setting appears in both Signal Path Setup and here in the Settings dialog.

Acoustic

Read more about input **Acoustic** mode in the **Input/ Output** topics in Chapter 6, Signal Path Setup.

Termination

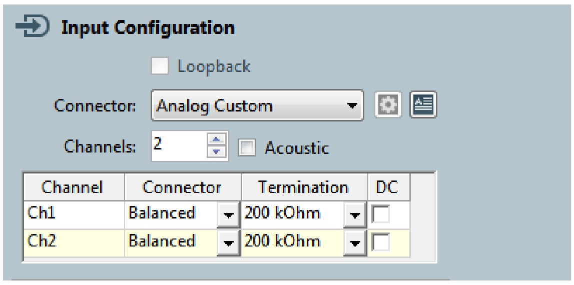
This control sets the input termination for both balanced input channels. Choices are:

- 200 kΩ
- 300 Ω
- 600 Ω

Input: Analog Custom

Connector

Analog Custom allows you to set **Connector, Termination** and **Coupling** settings independently for each input channel. A channel set to **Unbalanced** is connected to a front panel BNC connector; a channel set to **Balanced** is connected to a front panel XLR female connector.



Channels

The Channels setting appears in both Signal Path Setup and here in the Settings dialog.

Acoustic

Read more about input **Acoustic** mode in the **Input/ Output** topics in Chapter 6, Signal Path Setup.

The Custom configuration grid

Channel

Each channel has a row for custom selections.

Connector

Select the Connector (Balanced or Unbalanced) for a particular channel here.

Termination

Select the Termination for a particular channel here. The Termination options will depend upon the Connector selection.

DC

Select the AC or DC coupling for a particular channel here. Note that when Input Configuration is set to Custom, the High-pass Filter selections are only AC-coupled. Selecting DC in the Custom grid overrides the High-pass Filter selection for the channel, and sets it to DC-coupled. See page 53 and also Chapter 91 for a detailed discussion of Signal Path Setup Filters.

Filters and Device Under Test: Delay

See page 53 and also Chapter 91 for a detailed discussion of Signal Path Setup Filters. Note that analog system bandwidth (DAC sample rate) is set using the Low-Pass Filter menu, and that AC/DC coupling is set using the High-pass Filter menu.

Device Under Test

Delay

See the Input/Output topics in Chapter 6, Signal Path Setup, for a detailed discussions of DUT Delay.

10

SPS: APx555 analog I/O

Introduction

For analog signal generation, the APx555 uses either DSP/DAC signal generation, or the High Performance Sine Generator. See Chapter 5 for more information about these generation techniques.

Likewise, the APx555 has dedicated analog notch filters and an additional pair of ADCs for use with the High Performance Sine Analyzer.

However, these high performance features do not affect the configuration controls and options discussed in this chapter, which are much the same as the I/O Configuration controls for other 2-channel APx instruments.

This chapter discusses the Signal Path Setup analog I/O configurations for the APx555 analyzer. For other I/O configurations for the APx555, go to

- ADIO (advanced digital I/O) configurations, Chapter 13.
- DSIO (digital serial I/O) configurations, Chapter 14.
- HDMI+ARC I/O configuration, Chapter 15.
- Bluetooth configurations, Chapter 16.
- PDM configuration, Chapter 17.
- ASIO configurations, Chapter 18.
- External Source configuration, Chapter 5.
- File Input configurations, Chapter 19.

Common Signal Path Setup I/O Settings

- Acoustic (output) mode and output EQ are common to all analyzer analog output configurations,
- Acoustic (input) mode is common to all analyzer analog input configurations.
- Loopback, Channel Labels, Filters and Device Under Test: Delay settings are common to all analyzer input configurations.

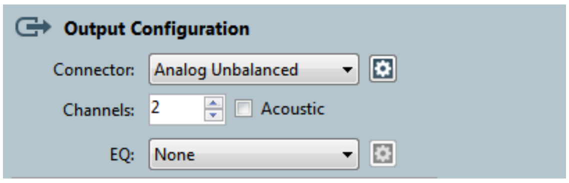
Each of these common features is discussed in the Input/Output topics in Chapter 6, Signal Path Setup.

Output Configuration

The Output Configuration panel allows you to select the number of output channels operating for your test, and to choose the output format and connectors you will be using.

The Settings button opens an output Settings dialog for each configuration, offering more detailed control.

Output: Analog Unbalanced



This selects the unbalanced analog outputs available on BNC connectors on the instrument front panel.

Channels

The Channels setting allows you to set the number of output channels (1–2) to be tested for this signal path.

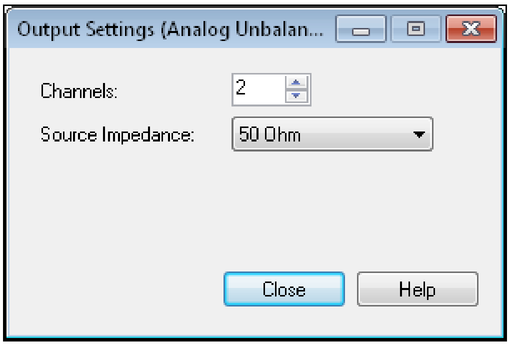
Acoustic

Read more about output Acoustic mode in the Input/Output topics in Chapter 6, Signal Path Setup.

EQ

Read more about output EQ in the Input/Output topics in Chapter 6, Signal Path Setup.

Settings for Analog Unbalanced Outputs



Channels

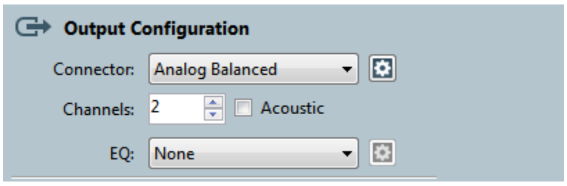
The **Channels** setting appears in both Signal Path Setup and here in the Settings dialog.

Source Impedance

Select the output **Source Impedance** here. For unbalanced outputs, the choices are

- 20 Ω (default)
- 50 Ω
- 75 Ω
- 100 Ω
- 600 Ω

Output: Analog Balanced



This selection enables the balanced analog outputs available on XLR3 male connectors and double banana connectors. The XLRs and the double bananas are wired in parallel.

Channels

The Channels setting allows you to set the number of output channels (1–2) to be tested for this signal path.

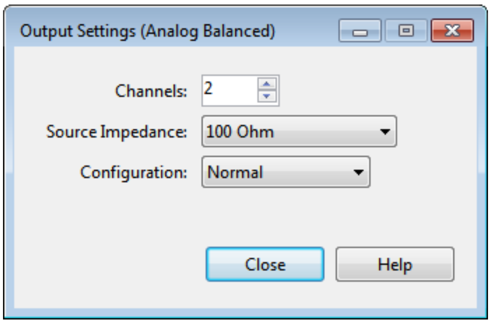
Acoustic

Read more about output **Acoustic** mode in the **Input/Output** topics in Chapter 6, Signal Path Setup.

EQ

Read more about output **EQ** in the **Input/Output** topics in Chapter 6, Signal Path Setup.

Settings for Analog Balanced Outputs



Channels

The Channels setting appears in both Signal Path Setup and here in the Settings dialog.

Source Impedance

Select the output source impedance here. For balanced outputs, the choices are

- 40 Ω
- 100 Ω
- 150 Ω
- 200 Ω
- 600 Ω

Configuration

The Configuration control sets the **Normal Mode / Common Mode Test (CMTST)** configuration for the balanced analog outputs, for all measurements.

Configuration choices are:

- **Normal**
  The default, differentially balanced output configuration. Signal plus (+) is connected to pin 2; signal minus (–) is connected to pin 3. This is the recommended setting for all measurements.
- **CMTST**
  A fixed common-mode output configuration. Signal plus (+) is connected to both pin 2 and pin 3, signal minus (–) is connected to pin 1 (ground). This configuration is recommended only for common-mode testing.\CMTST (IEC pin 2)
  A fixed common-mode output configuration, using the IEC unbalanced leg configuration. Source impedance is fixed at 40 Ω. Signal plus (+) is connect to pin 2 and pin 3, with an additional 10 Ω resistor inserted in the pin 2 leg. Signal minus (–) is connected to pin 1 (ground).
- **CMTST (IEC pin 2)**
  A fixed common-mode output configuration, using the IEC unbalanced leg configuration. Source impedance is fixed at 40 Ω. Signal plus (+) is connect to pin 2 and pin 3, with an additional 10 Ω

resistor inserted in the pin 2 leg. Signal minus (–) is connected to pin 1 (ground).

- CMTST (IEC pin 3)
  A fixed common-mode output configuration, using the IEC unbalanced leg configuration. Source impedance is fixed at 40 Ω. Signal plus (+) is connect to pin 2 and pin 3, with an additional 10 Ω resistor inserted in the pin 3 leg. Signal minus (–) is connected to pin 1 (ground).

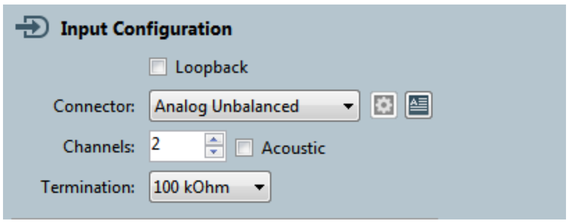
Setting a common mode configuration for all measurements can produce unexpected results. For Sequence Mode common mode rejection ratio tests, use the CMRR measurement instead.

Input Configuration

The Input Configuration panel allows you to select the number of input channels operating for your test, and to choose the input format and connectors you will be using.

The **Settings** button opens an input **Settings** dialog for each configuration, offering more detailed control.

Input: Analog Unbalanced



Connector

Analog Unbalanced sets both channels to the unbalanced analog inputs, available on BNC connectors on the instrument front panel.

Channels

The Channels setting appears in both Signal Path Setup and here in the Settings dialog.

Acoustic

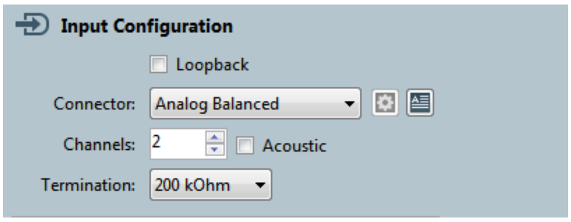
Read more about input **Acoustic** mode in the **Input/Output** topics in Chapter 6, Signal Path Setup.

Termination

This control sets the input termination for both unbalanced input channels. Choices are:

- 100 kΩ
- 300 Ω
- 600 Ω

Input: Analog Balanced



Connector

Analog Balanced sets both channels to the balanced analog inputs, available on XLR female connectors on the instrument front panel.

Channels

The Channels setting appears in both Signal Path Setup and here in the Settings dialog.

Acoustic

Read more about input **Acoustic** mode in the **Input/Output** topics in Chapter 6, Signal Path Setup.

Termination

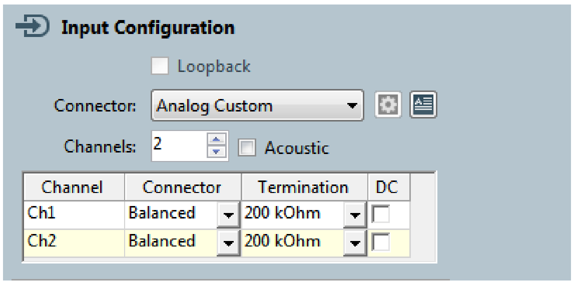
This control sets the input termination for both balanced input channels. Choices are:

- 200 kΩ
- 300 Ω
- 600 Ω

Input: Analog Custom

Connector

Analog Custom allows you to set **Connector, Termination** and **Coupling** settings independently for each input channel. A channel set to **Unbalanced** is connected to a front panel BNC connector; a channel set to **Balanced** is connected to a front panel XLR female connector.



Channels

The Channels setting appears in both Signal Path Setup and here in the Settings dialog.

Acoustic

Read more about input **Acoustic** mode in the **Input/Output** topics in Chapter 6, Signal Path Setup.

The Custom configuration grid

Channel

Each channel has a row for custom selections.

Connector

Select the Connector (Balanced or Unbalanced) for a particular channel here.

Termination

Select the Termination for a particular channel here. The Termination options will depend upon the Connector selection.

DC

Select the AC or DC coupling for a particular channel here. Note that when Input Configuration is set to Custom, the High-pass Filter selections are only AC-coupled. Selecting DC in the Custom grid overrides the High-pass Filter selection for the channel, and sets it to DC-coupled. See page 53 and also Chapter 91 for a detailed discussion of Signal Path Setup Filters.

Filters and Device Under Test: Delay

Filters

See page 53 and also Chapter 91 for a detailed discussion of Signal Path Setup Filters. Note that analog system bandwidth (DAC sample rate) is set using the Low-Pass Filter menu, and that AC/DC coupling is set using the High-pass Filter menu.

Device Under Test

Delay

See the Input/Output topics in Chapter 6, Signal Path Setup, for a detailed discussions of DUT Delay.

11

SPS: APx58x analog I/O and APx582 analog in

Introduction

This chapter discusses the Signal Path Setup analog I/O configurations for the APx585 and 586 analyzers. Additionally, the Input Configuration discussed here applies to the analog inputs of the APx582. For other APx585 or 586 I/O configurations, go to

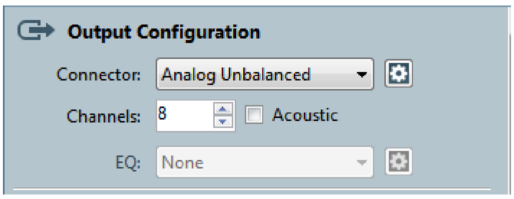
- DIO (digital I/O) configurations, Chapter 12.
- DSIO (digital serial I/O) configurations, Chapter 14.
- HDMI+ARC I/O configuration, Chapter 15.
- Bluetooth configurations, Chapter 16.
- PDM configuration, Chapter 17.
- External Source configuration, Chapter 5.
- File Input configurations, Chapter 19.

Output Configuration

The Output Configuration panel allows you to select the number of output channels operating for your test, and to choose the output format and connectors you will be using.

The **Settings** button opens an output Settings dialog for each configuration, offering more detailed control.

Output: Analog Unbalanced

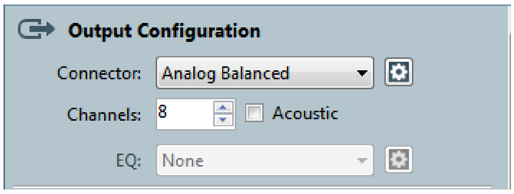


This selects the unbalanced analog outputs available on BNC connectors on the instrument front panel. The source impedance for the unbalanced analog outputs is 50 Ω.

Channels

The Channels setting allows you to set the number of output channels (1–8) to be tested for this signal path.

Output: Analog Balanced



This selection enables the balanced analog outputs available on a 25-pin female D-Sub connector (see page 11) on the instrument front panel. The source impedance for the balanced analog outputs is 100 Ω. When Analog Balanced is selected, the **Channels** selection field (duplicated in the Settings dialog) is available.

Channels

The Channels setting allows you to set the number of output channels (1–8) to be tested for this signal path.

Acoustic

Read more about output **Acoustic** mode in the **Input/Output** topics in Chapter 6, Signal Path Setup.

EQ

Read more about output **EQ** in the **Input/Output** topics in Chapter 6, Signal Path Setup.

For the APx585 and 586, **Output EQ** is unavailable when output Channels is set to more than 2.

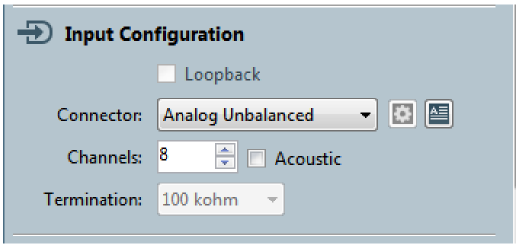
Input Configuration

The Input Configuration panel allows you to select the number of input channels operating for your test, and to choose the input format and connectors you will be using. The **Settings...** button opens an input Settings dialog for each configuration, offering more detailed control. You can also choose identifying names and colors for your test input channels.

Loopback

Loopback is not available for APx582, 585 or 586 analog inputs.

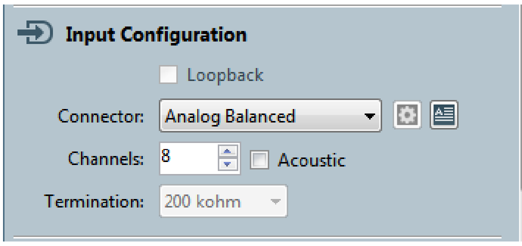
Input: Analog Unbalanced



Connector

Analog Unbalanced sets all channels to the unbalanced analog inputs, available on BNC connectors on the instrument front panel. The APx586 is fitted with 8 additional BNC connectors to provide 16 unbalanced inputs. The input termination for the unbalanced analog inputs is fixed at 100 kΩ.

Input: Analog Balanced



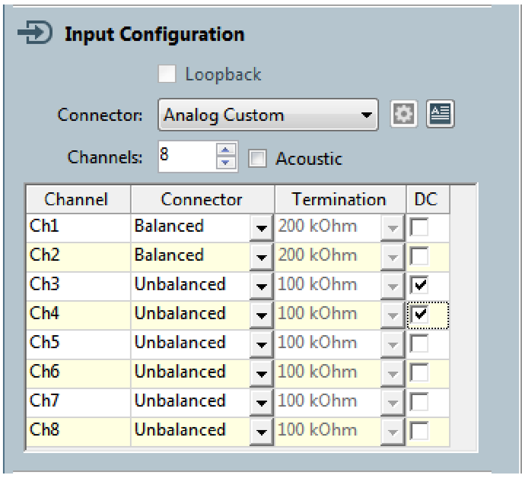
Connector

Analog Balanced sets all channels to the balanced analog inputs, available on 25-pin D-Sub connectors on the instrument front panel. The APx586 is fitted with a second D-Sub connector to provide 16 unbalanced inputs. The input termination for the balanced analog inputs is fixed at 200 kΩ.

Input: Analog Custom

Connector

Analog Custom allows you to choose **Unbalanced** or **Balanced** connections and **Coupling** settings independently for each input channel. Channels set to **Unbalanced** are connected to the front panel BNC connectors, with a fixed input termination of 100 kΩ; channels set to **Balanced** are connected to the front panel D-Sub connector, with a fixed input termination of 200 kΩ;



Channels

The **Channels** setting allows you to choose the number of input channels to be tested for this signal path. For the APx582 or 585, the maximum number of input channels is 8. For the APx586, the maximum number of input channels is 16.

Acoustic

Read more about input **Acoustic** mode in the **Input/ Output** topics in Chapter 6, Signal Path Setup.

The Custom configuration grid

Channel

Each channel has a row for custom selections.

Connector

Select the Connector (Balanced or Unbalanced) for a particular channel here.

Termination

Select the Termination for a particular channel here. The Termination options will depend upon the Connector selection.

DC

Select the AC or DC coupling for a particular channel here. Note that when Input Configuration is set to **Custom**, the **High-pass Filter** selections are only AC-coupled. Selecting **DC** in the **Custom** grid overrides the **High-pass Filter** selection for the channel, and sets it to DC-coupled. See page 53 and also Chapter 91 for a detailed discussion of Signal Path Setup Filters.

Filters and Device Under Test: Delay

Filters

See page 53 and also Chapter 91 for a detailed discussion of Signal Path Setup **Filters.** Note that analog system bandwidth (DAC sample rate) is set using the

Low-Pass Filter menu, and that AC/DC coupling is set using the High-pass Filter menu.

Device Under Test

Delay

See the Input/Output topics in Chapter 6, Signal Path Setup, for a detailed discussions of DUT Delay.

12

SPS: Digital I/O

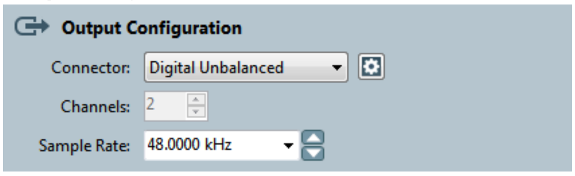
Introduction

Most APx Series audio analyzers have a digital input/output interface (DIO). The software control for the DIO is the same for all APx Series analyzers. For information on the other supported digital interfaces, see Chapter 14 (DSIO), Chapter 15 (HDMI), Chapter 16 (Bluetooth), and Chapter 17, (PDM).

An advanced digital input/output (ADIO) interface is also available, standard on an APx555 and an option for the APx525/585 family. See Chapter 13 for more information about the advanced digital input/output.

Output Configuration

Output: Digital Unbalanced



This selection enables the unbalanced digital electrical output, available on a BNC connector on the instrument front panel.

The **Settings** button opens an output **Settings** dialog for each configuration, offering more detailed control.

Channels

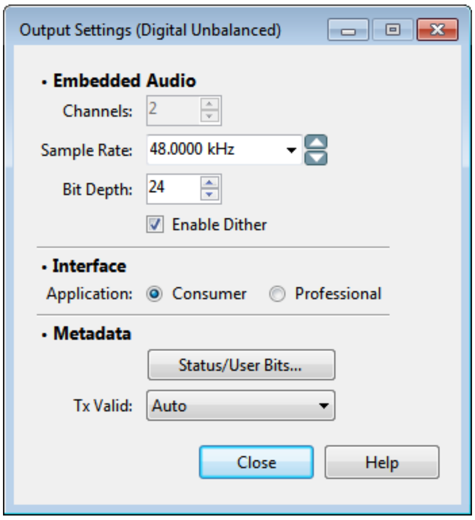
Audio channels are fixed at two.

Sample Rate

To set the output sample rate (Fs), click the up/down arrows to select a standard rate, or enter an arbitrary value in the **Sample Rate** field. Minimum setting is 8 kHz; maximum is 216 kHz. Default is 48 kHz. **Sample Rate** is also available in the Settings dialog.

The APx515 has a minimum output sample rate of 22 kHz.

Settings for Digital Unbalanced Output



Embedded Audio

The **Channels** and **Sample Rate** fields in the Settings dialog duplicate the fields in the Output Configuration panel.

Bit Depth

The Bit Depth control allows you to set the output bit depth (also called word length). The default setting is 24 bits, which is also the maximum setting. Minimum setting is 8 bits.

Enable Dither

When the **Enable Dither** checkbox is checked, dither is **ON** (the default). When **ON**, dither is set according to the bit depth, using TPDF (triangular probability density function) dither at ±1 LSB.

Uncheck the checkbox to turn dither **OFF**.

Interface

Application: Consumer

Consumer application (the default) sets two aspects of the unbalanced digital output signal:

1. It sets the nominal interface signal voltage at the unbalanced output connector to 0.5 Vpp. This is the defined signal voltage for consumer applications such as SPDIF and IEC60958-3.
2. It sets the status bits for both subframes A and B to the consumer application. The status bits can be customized or even set to professional application in the Set Status Bits / User Bits panel.

Application: Professional

Professional application sets two aspects of the unbalanced digital output signal:

3. 1.It sets the nominal interface signal voltage at the unbalanced output connector to 1.0 Vpp. This is the defined signal voltage for unbalanced professional applications such as AES3id and SMPTE 276M.
4. 2.It sets the status bits for both subframes A and B to the professional application. The status bits can be customized or even set to consumer application in the Set Status Bits / User Bits panel.

Metadata

Status/User Bits

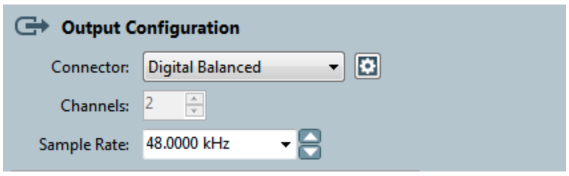
This button opens the Set Status Bits / Set User Bits dialog. See page 539 for more information about setting status bits and user bits.

Tx Valid

The last four time slots of each subframe contain a validity bit, a user data bit, a status bit, and a parity bit. By default, APx automatically transmits the bit as Valid for linear PCM embedded audio, and Invalid for coded audio.

- Auto
  APx automatically transmits the validity bit as **Valid** for linear PCM embedded audio, and **Invalid** for coded audio.
- Valid
  The validity bit is transmitted as **Valid** without regard to the nature of the embedded audio.
- Invalid
  The validity bit is transmitted as **Invalid** without regard to the nature of the embedded audio.

Output: Digital Balanced



This selection enables the balanced digital electrical output, available on a male XLR connector on the instrument front panel.

Note: Earlier APx585/586 Series analyzers may be fitted with a Model 109 Digital I/O module, which does not support balanced DIO.

Channels

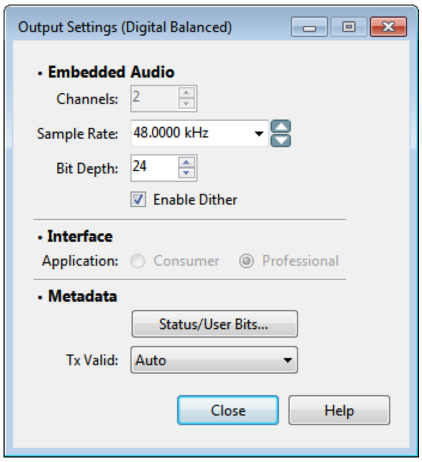
Audio channels are fixed at two.

Sample Rate

To set the output sample rate (Fs), click the up/down arrows to select a standard rate, or enter an arbitrary value in the **Sample Rate** field. Minimum setting is 8 kHz; maximum is 216 kHz. Default is 48 kHz. **Sample Rate** is also available in the Settings dialog.

The APx515 has a minimum output sample rate of 22 kHz.

Settings for Digital Balanced Output



Embedded Audio

The **Channels** and **Sample Rate** fields in the Settings dialog duplicate the fields in the Output Configuration panel.

Bit Depth

The Bit Depth control allows you to set the output bit depth (also called digital word length). The default setting is 24 bits, which is also the maximum setting. Minimum setting is 8 bits.

Enable Dither

When the **Enable Dither** checkbox is checked, dither is **ON** (the default). When **ON**, dither is set according to the bit depth, using TPDF (triangular probability density function) dither at ±1 LSB.

Uncheck the checkbox to turn dither **OFF**.

Interface

Application

The Application buttons are not available in Digital Balanced output; instead, these aspects of the balanced digital output signal are set:

5. 1.The nominal interface signal voltage at the balanced output connector is always set to 5.0 Vpp. This is within the range of defined signal voltages for balanced professional applications such as AES3, AES/EBU and IEC60958-4.
6. 2.By default, the status bits for both subframes A and B are set to the professional application. The status bits can be customized or even set to consumer application in the Set Status Bits / User Bits panel.

Metadata

Status/User Bits

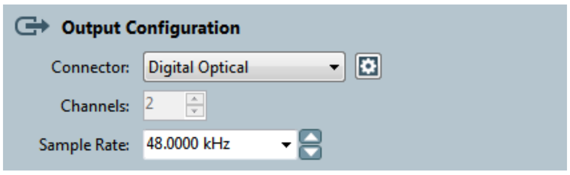
This button opens the Set Status Bits / Set User Bits dialog. See page 539 for more information about setting status bits and user bits.

Tx Valid

The last four time slots of each subframe contain a validity bit, a user data bit, a status bit, and a parity bit. By default, APx automatically transmits the bit as Valid for linear PCM embedded audio, and Invalid for coded audio.

- Auto
  APx automatically transmits the validity bit as **Valid** for linear PCM embedded audio, and **Invalid** for coded audio.
- Valid
  The validity bit is transmitted as **Valid** without regard to the nature of the embedded audio.
- Invalid
  The validity bit is transmitted as **Invalid** without regard to the nature of the embedded audio.

Output: Digital Optical



This selection enables the digital optical output, available on a Toslink connector on the instrument front panel.

Channels

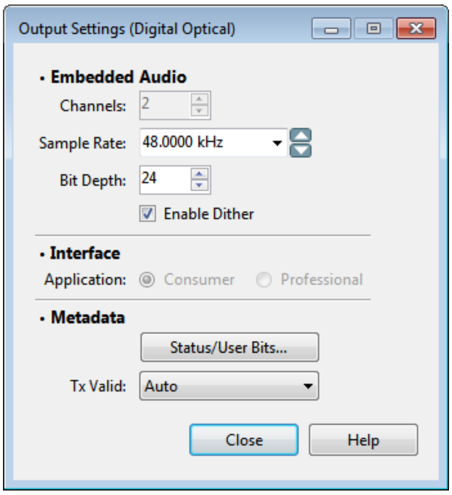
Audio channels are fixed at two.

Sample Rate

To set the output sample rate (Fs), click the up/down arrows to select a standard rate, or enter an arbitrary value in the **Sample Rate** field. Minimum setting is 8 kHz; maximum is 216 kHz. Default is 48 kHz. **Sample Rate** is also available in the Settings dialog.

The APx515 has a minimum output sample rate of 22 kHz.

Settings for Digital Optical Output



Embedded Audio

The **Channels** and **Sample Rate** fields in the Settings dialog duplicate the fields in the Output Configuration panel.

Bit Depth

The Bit Depth control allows you to set the output bit depth (also called digital word length). The default setting is 24 bits, which is also the maximum setting. Minimum setting is 8 bits.

Enable Dither

When the **Enable Dither** checkbox is checked, dither is **ON** (the default). When **ON**, dither is set according to the bit depth, using TPDF (triangular probability density function) dither at ±1 LSB.

Uncheck the checkbox to turn dither **OFF**.

Interface

Application

The Application buttons are not available in Digital Optical output; instead, the status bits for both subframes A and B are set to the consumer application. The status bits can be customized or even set to professional application in the Set Status Bits / User Bits panel.

Metadata

Status/User Bits

This button opens the Set Status Bits / Set User Bits dialog. See page 539 for more information about setting status bits and user bits.

Tx Valid

The last four time slots of each subframe contain a validity bit, a user data bit, a status bit, and a parity bit. By default, APx automatically transmits the bit as Valid for linear PCM embedded audio, and Invalid for coded audio.

- **Auto**
  APx automatically transmits the validity bit as **Valid** for linear PCM embedded audio, and **Invalid** for coded audio.
- **Valid**
  The validity bit is transmitted as **Valid** without regard to the nature of the embedded audio.
- **Invalid**
  The validity bit is transmitted as **Invalid** without regard to the nature of the embedded audio.

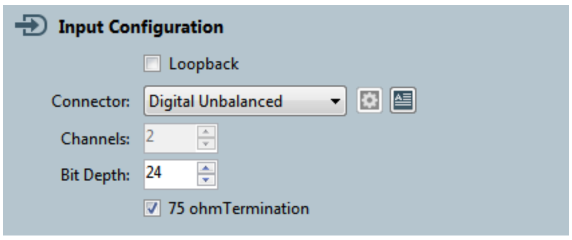
Input Configuration

Operation with absent or corrupt interface signal

When Input Configuration is set to a digital input and no signal is present, or if the signal is corrupt or out of range, the input receiver cannot lock (synchronize) to the interface signal, and no valid audio can be recovered. The sample rate indicator in the Status Bar will display an “unlocked” warning, and any measurement result referencing audio from the unlocked input will display an “----” (invalid) result. See page 560 for more about invalid results.

The **Settings** button is unavailable for DIO inputs.

Input: Digital Unbalanced



This selection enables the unbalanced digital electrical input, available on a BNC connector on the instrument front panel.

Channels

Audio channels are fixed at two.

Bit Depth

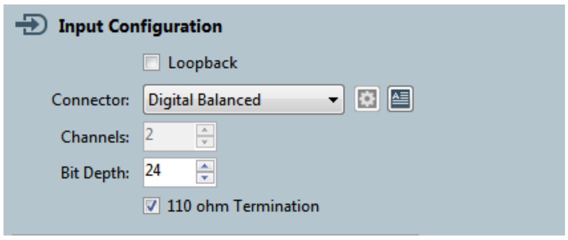
The Bit Depth control allows you to set the input bit depth (digital word length). The default setting is 24 bits, which is also the maximum setting. Minimum setting is 8 bits. The input signal is truncated (not dithered down) if Bit Depth is set to a lower value than Bit Depth of the signal.

We recommend that you use the default setting of 24 bits, unless you have a special application that requires that you mask out low-order bits.

75 ohm Termination checkbox

When Digital Unbalanced is selected, the **75 ohm Termination** checkbox is available on the panel. When this checkbox is set (the default), the digital unbalanced input is terminated in a resistance of 75 Ω. This is the correct setting for terminating an unbalanced digital audio connection for both the SPDIF / IEC60958-3 (consumer) and AES-3id / SMPTE276M (professional) formats. If you uncheck the checkbox, the digital unbalanced input is unterminated. An unterminated setting is useful when bridging a digital connection (as with a “T” connector). An unbalanced digital connection should be properly terminated in 75 Ω at the end of its run.

Input: Digital Balanced



Note: Earlier APx585/586 Series analyzers may be fitted with a Model 109 Digital I/O module, which does not support balanced DIO.

Digital Balanced enables the balanced digital electrical input, available on an XLR female connector on the instrument front panel. This input is optimized to accept signal at the professional signal level for balanced signals compatible with AES3 and IEC60958-4 (5 Vpp).

Channels

Audio channels are fixed at two.

Bit Depth

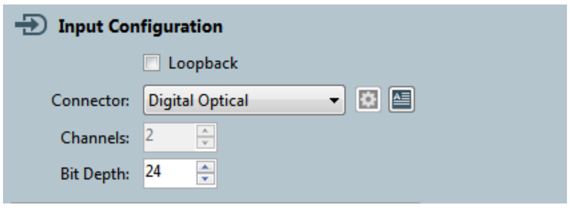
The Bit Depth control allows you to set the input bit depth (digital word length). The default setting is 24 bits, which is also the maximum setting. Minimum setting is 8 bits. The input signal is truncated (not dithered down) if Bit Depth is set to a lower value than Bit Depth of the signal.

We recommend that you use the default setting of 24 bits, unless you have a special application that requires that you mask out low-order bits.

110 Ohm Termination checkbox

When Digital Balanced is selected, the 110 Ohm Termination checkbox is available on the panel. When this checkbox is set (the default), the digital balanced input is terminated in a resistance of 110 Ω. This is the correct setting for terminating a balanced digital audio connection for AES3 / AES-EBU / IEC60958-4 professional formats. If you uncheck the checkbox, the digital balanced input is unterminated. This setting is useful when bridging a digital connection (as with a “Y” cable). A balanced digital connection should be properly terminated in 110 Ω at the end of its run. More settings are available in the Settings dialog.

Input: Digital Optical



Digital Optical enables the digital optical input, available on a Toslink connector on the instrument front panel.

Channels

Audio channels are fixed at two.

Bit Depth

The Bit Depth control allows you to set the input bit depth (digital word length). The default setting is 24 bits, which is also the maximum setting. Minimum setting is 8 bits. The input signal is truncated (not dithered down) if Bit Depth is set to a lower value than Bit Depth of the signal.

We recommend that you use the default setting of 24 bits, unless you have a special application that requires that you mask out low-order bits.

13

SPS: Advanced Digital I/O

Introduction

The advanced digital input/output (ADIO) interface is the standard digital interface on an APx555, and an option for the APx52x/58x family. The ADIO is similar to the standard DIO described in Chapter 12, but differs in that it offers impairments to the transmitted digital interface signal, and the ability to measure certain characteristics of the received interface signal.

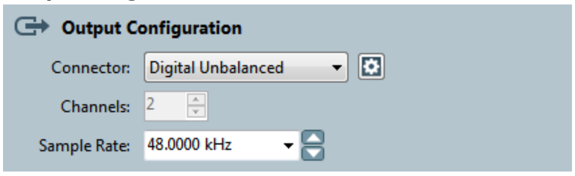
The transmitted interface signal can be impaired with false metadata, validity and parity errors, jitter, noise, common mode and risetime impairments (including cable simulation), to test the tolerance of a downstream device.

The level and sample rate of the received interface signal can be measured, and the jitter can be analyzed in great detail.

For information on the other supported digital interfaces, see Chapter 14 (DSIO), Chapter 15 (HDMI), Chapter 16 (Bluetooth), and Chapter 17, (PDM).

Output Configuration

Output: Digital Unbalanced



This selection enables the unbalanced digital electrical output, available on a BNC connector on the instrument front panel.

The **Settings** button opens an output **Settings** dialog for each configuration, offering more detailed control.

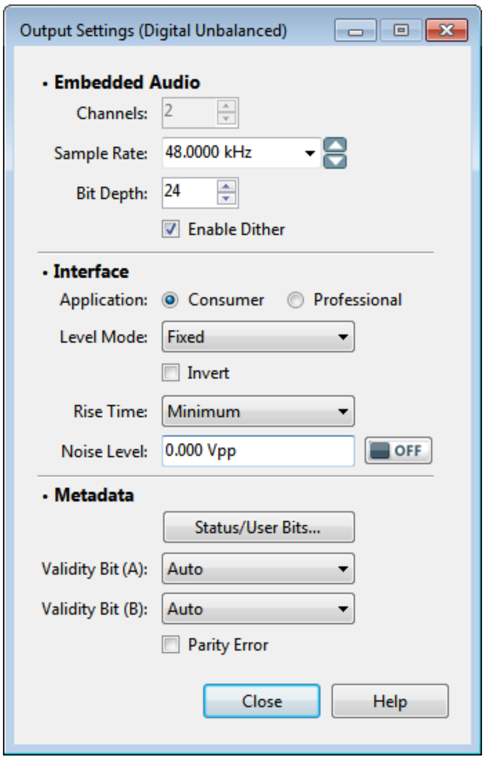
Channels

Audio channels are fixed at two.

Sample Rate

To set the output sample rate (Fs), click the up/down arrows to select a standard rate, or enter an arbitrary value in the **Sample Rate** field. Minimum setting is 8 kHz; maximum is 216 kHz. Default is 48 kHz. **Sample Rate** is also available in the Settings dialog.

Settings for Digital Unbalanced Output



Embedded Audio

The **Channels** and **Sample Rate** fields in the Settings dialog duplicate the fields in the Output Configuration panel.

Bit Depth

The Bit Depth control allows you to set the output bit depth (also called word length). The default setting is 24 bits, which is also the maximum setting. Minimum setting is 8 bits.

Enable Dither

When the Enable Dither checkbox is checked, dither is ON (the default). When ON, dither is set according to the bit depth, using TPDF (triangular probability density function) dither at ±1 LSB.

Uncheck the checkbox to turn dither OFF.

Interface

Application: Consumer

Consumer application (the default) sets two aspects of the unbalanced digital output signal:

1. It sets the nominal interface signal voltage at the unbalanced output connector to 0.5 Vpp. This is the defined signal voltage for consumer applications such as SPDIF and IEC60958-3.
2. It sets the status bits for both subframes A and B to the consumer application. The status bits can be customized or even set to professional application in the Set Status Bits / User Bits panel.

Application: Professional

Professional application sets two aspects of the unbalanced digital output signal:

3. 1.It sets the nominal interface signal voltage at the unbalanced output connector to 1.0 Vpp. This is the defined signal voltage for unbalanced professional applications such as AES3id and SMPTE 276M.
4. 2.It sets the status bits for both subframes A and B to the professional application. The status bits can be customized or even set to consumer application in the Set Status Bits / User Bits panel.

Level Mode

- Fixed
  The standard voltage for the unbalanced consumer application is 0.5 Vpp; for the unbalanced professional application it is 1.0 Vpp. The Fixed selection sets the interface voltage to the standard, according to the Application settings made above.
- Custom
  You can enter a custom value for the interface voltage in the Level field, as an impairment to test a downstream receiver. The acceptable range is 0 Vpp to 2.5 Vpp.

Invert

This checkbox inverts the interface signal waveform to test the tolerance of a downstream receiver.

Rise Time

The rise time of the pulses in the interface waveform should be short for error-free reception. However, cable capacitance and other environmental factors can degrade the signal by lengthening pulse rise time. Rise time impairments can be imposed on the interface signal to test the tolerance of a downstream receiver.

- Minimum
  The Minimum setting uses no artificial impairments and is the fastest rise time available from the digital transmitter.
- Cable Simulation
  The Cable Simulation setting lengthens the pulse rise time by simulating the capacitance of a long cable.
- Custom
  Enter a custom rise time impairment in the field provided. The acceptable range is 12.00 ns to 100.0 ns.

Noise Level

You can add noise to the interface signal as an impairment to test the tolerance of a downstream receiver. Enter the Noise Level in the field provided, and click ON. The acceptable range is 0.0 Vpp to 625.0 mVpp.

Metadata

Status/User Bits

This button opens the Set Status Bits / Set User Bits dialog. See page 539 for more information about setting status bits and user bits.

Validity Bit (A) / Validity Bit (B)

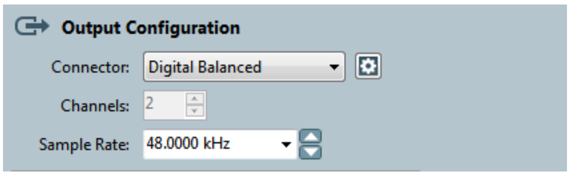
The last four time slots of each subframe contain a validity bit, a user data bit, a status bit, and a parity bit. By default, APx automatically transmits the bit as Valid for linear PCM embedded audio, and Invalid for coded audio.

- Auto
  APx automatically transmits the validity bit as Valid for linear PCM embedded audio, and Invalid for coded audio.
- Valid
  The validity bit is transmitted as Valid without regard to the nature of the embedded audio.
- Invalid
  The validity bit is transmitted as Invalid without regard to the nature of the embedded audio.

Parity Error

One of the administrative bits in the interface is the parity (P) bit. In normal operation, even parity is maintained in each subframe. By setting a checkmark in the Parity Error checkbox, the parity bit can be set to produce odd parity, as an impairment to test a downstream receiver.

Output: Digital Balanced



This selection enables the balanced digital electrical output, available on a male XLR connector on the instrument front panel.

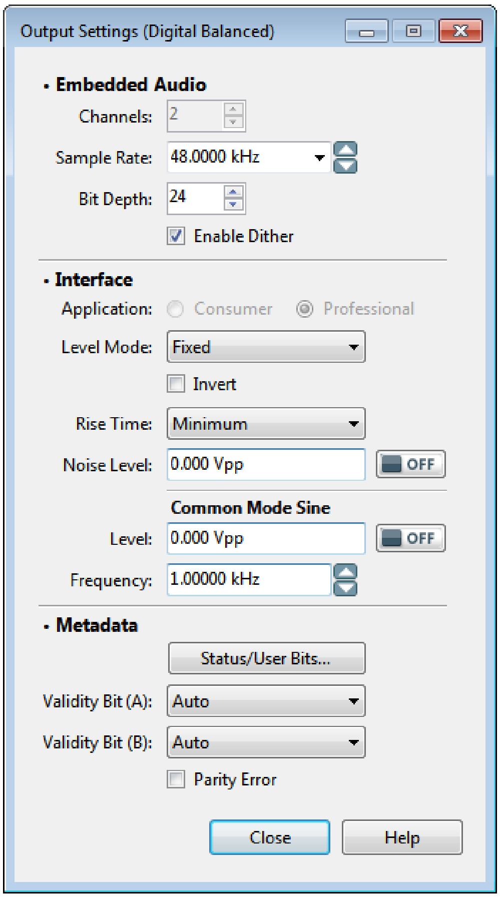
Channels

Audio channels are fixed at two.

Sample Rate

To set the output sample rate (Fs), click the up/down arrows to select a standard rate, or enter an arbitrary value in the **Sample Rate** field. Minimum setting is 8 kHz; maximum is 216 kHz. Default is 48 kHz. **Sample Rate** is also available in the Settings dialog.

Settings for Digital Balanced Output



Embedded Audio

The **Channels** and **Sample Rate** fields in the Settings dialog duplicate the fields in the Output Configuration panel.

Bit Depth

The Bit Depth control allows you to set the output bit depth (also called digital word length). The default setting is 24 bits, which is also the maximum setting. Minimum setting is 8 bits.

Enable Dither

When the **Enable Dither** checkbox is checked, dither is **ON** (the default). When **ON**, dither is set according to the bit depth, using TPDF (triangular probability density function) dither at ±1 LSB.

Uncheck the checkbox to turn dither **OFF**.

Interface

Application

The Application buttons are not available in Digital Balanced output; instead, these aspects of the balanced digital output signal are set:

5. 1.The nominal interface signal voltage at the balanced output connector is always set to 5.0 Vpp. This is within the range of defined signal voltages for balanced professional applications such as AES3, AES/EBU and IEC60958-4.
6. 2.By default, the status bits for both subframes A and B are set to the professional application. The status bits can be customized or even set to consumer application in the Set Status Bits / User Bits panel.

Level Mode

- Fixed
  The standard voltage for the unbalanced consumer application is 0.5 Vpp; for the unbalanced professional application it is 1.0 Vpp. The Fixed selection sets the interface voltage to the standard, according to the Application settings made above.
- Custom
  You can enter a custom value for the interface voltage in the Level field, as an impairment to test a downstream receiver. The acceptable range is 0 Vpp to 2.5 Vpp.

Invert

This checkbox inverts the interface signal waveform to test the tolerance of a downstream receiver.

Rise Time

The rise time of the pulses in the interface waveform should be short for error-free reception. However, cable capacitance and other environmental factors can degrade the signal by lengthening pulse rise time. Rise time impairments can be imposed on the interface signal to test the tolerance of a downstream receiver.

- Minimum
  The **Minimum** setting uses no artificial impair-

ments and is the fastest rise time available from the digital transmitter.

- Cable Simulation
  The **Cable Simulation** setting lengthens the pulse rise time by simulating the capacitance of a long cable.
- Custom
  Enter a custom rise time impairment in the field provided. The acceptable range is 12.00 ns to 100.0 ns.

Noise Level

You can add noise to the interface signal as an impairment to test the tolerance of a downstream receiver. Enter the **Noise Level** in the field provided, and click **ON**. The acceptable range is 0.0 Vpp to 2.0.

Common Mode Sine

A sine signal can be imposed on the digital interface signal as a common mode impairment, to test the tolerance of a downstream receiver.

Level
Set the common mode sine level here. The acceptable range is 0.0 Vpp to 20.0 Vpp.

Frequency
Set the common mode sine frequency here. The acceptable range is 20 Hz to 100 kHz.

Metadata

Status/User Bits

This button opens the Set Status Bits / Set User Bits dialog. See page 539 for more information about setting status bits and user bits.

Validity Bit (A) / Validity Bit (B)

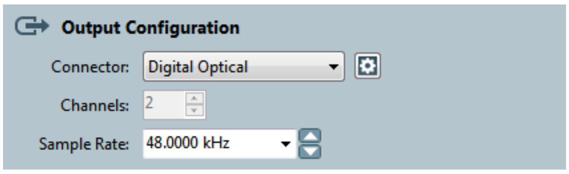
The last four time slots of each subframe contain a validity bit, a user data bit, a status bit, and a parity bit. By default, APx automatically transmits the bit as Valid for linear PCM embedded audio, and Invalid for coded audio.

- Auto
  APx automatically transmits the validity bit as **Valid** for linear PCM embedded audio, and **Invalid** for coded audio.
- Valid
  The validity bit is transmitted as **Valid** without regard to the nature of the embedded audio.
- Invalid
  The validity bit is transmitted as **Invalid** without regard to the nature of the embedded audio.

Parity Error

One of the administrative bits in the interface is the parity (P) bit. In normal operation, even parity is maintained in each subframe. By setting a checkmark in the **Parity Error** checkbox, the parity bit can be set to produce odd parity, as an impairment to test a downstream receiver.

Output: Digital Optical



This selection enables the digital optical output, available on a Toslink connector on the instrument front panel.

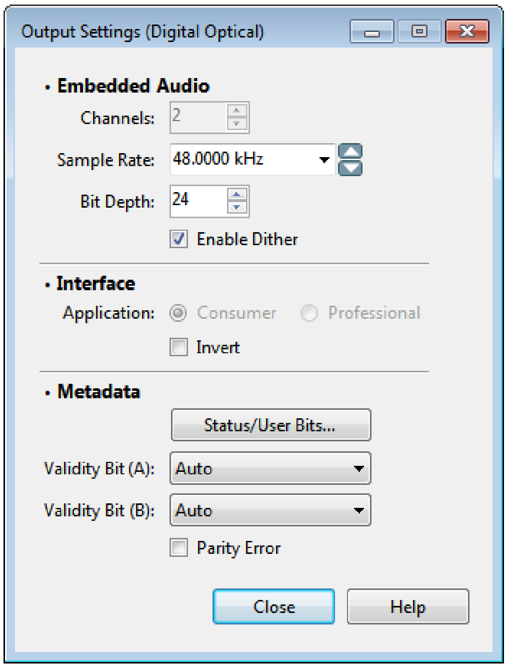
Channels

Audio channels are fixed at two.

Sample Rate

To set the output sample rate (Fs), click the up/down arrows to select a standard rate, or enter an arbitrary value in the **Sample Rate** field. Minimum setting is 8 kHz; maximum is 216 kHz. Default is 48 kHz. **Sample Rate** is also available in the Settings dialog.

Settings for Digital Optical Output



Embedded Audio

The **Channels** and **Sample Rate** fields in the Settings dialog duplicate the fields in the Output Configuration panel.

Bit Depth

The Bit Depth control allows you to set the output bit depth (also called digital word length). The default setting is 24 bits, which is also the maximum setting. Minimum setting is 8 bits.

Enable Dither

When the **Enable Dither** checkbox is checked, dither is **ON** (the default). When **ON**, dither is set according to the bit depth, using TPDF (triangular probability density function) dither at ±1 LSB.

Uncheck the checkbox to turn dither **OFF**.

Interface

Application

The Application buttons are not available in Digital Optical output; instead, the status bits for both subframes A and B are set to the consumer application. The status bits can be customized or even set to professional application in the Set Status Bits / User Bits panel.

Invert

This checkbox inverts the interface signal waveform to test the tolerance of a downstream receiver.

Metadata

Status/User Bits

This button opens the Set Status Bits / Set User Bits dialog. See page 539 for more information about setting status bits and user bits.

Validity Bit (A) / Validity Bit (B)

The last four time slots of each subframe contain a validity bit, a user data bit, a status bit, and a parity bit. By default, APx automatically transmits the bit as Valid for linear PCM embedded audio, and Invalid for coded audio.

- **Auto**
  APx automatically transmits the validity bit as **Valid** for linear PCM embedded audio, and **Invalid** for coded audio.
- **Valid**
  The validity bit is transmitted as **Valid** without regard to the nature of the embedded audio.
- **Invalid**
  The validity bit is transmitted as **Invalid** without regard to the nature of the embedded audio.

Parity Error

One of the administrative bits in the interface is the parity (**P**) bit. In normal operation, even parity is maintained in each subframe. By setting a checkmark in the **Parity Error** checkbox, the parity bit can be set to produce odd parity, as an impairment to test a downstream receiver.

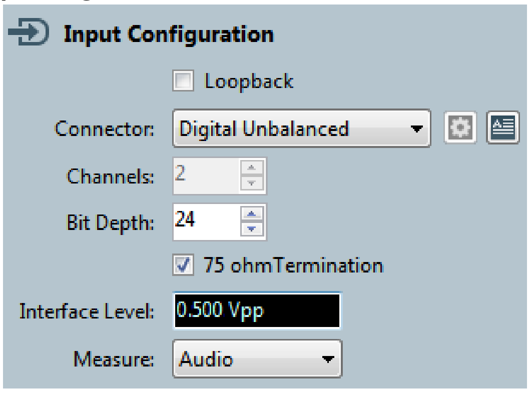
Input Configuration

Operation with absent or corrupt interface signal

When Input Configuration is set to a digital input and no signal is present, or if the signal is corrupt or out of range, the input receiver cannot lock (synchronize) to the interface signal, and no valid audio can be recovered. The sample rate indicator in the Status Bar will display an “unlocked” warning, and any measurement result referencing audio from the unlocked input will display an “----” (invalid) result. See page 560 for more about invalid results.

The **Settings** button is unavailable for ADIO inputs.

Input: Digital Unbalanced



This selection enables the unbalanced digital electrical input, available on a BNC connector on the instrument front panel.

Channels

Audio channels are fixed at two.

Bit Depth

The Bit Depth control allows you to set the input bit depth (digital word length). The default setting is 24 bits, which is also the maximum setting. Minimum setting is 8 bits. The input signal is truncated (not dithered down) if Bit Depth is set to a lower value than Bit Depth of the signal.

We recommend that you use the default setting of 24 bits, unless you have a special application that requires that you mask out low-order bits.

75 ohm Termination checkbox

When Digital Unbalanced is selected, the **75 ohm Termination** checkbox is available on the panel. When this checkbox is set (the default), the digital unbalanced input is terminated in a resistance of 75 Ω. This is the correct setting for terminating an unbalanced digital audio connection for both the SPDIF / IEC60958-3 (consumer) and AES-3id / SMPTE276M (professional) formats. If you uncheck the checkbox, the digital unbalanced input is unterminated. An unterminated setting is useful when bridging a digital connection (as with a “T” connector). An unbalanced digital connection should be properly terminated in 75 Ω at the end of its run.

Interface Level (Sequence Mode)

This field is only available in Sequence Mode when the Advanced Digital Input/Output module is fitted, and

Input Configuration is set to Digital Unbalanced or Digital Balanced. It displays the peak-to-peak voltage level of the incoming digital interface waveform.

In Bench Mode, the same information is available from the Digital Interface Level meter.

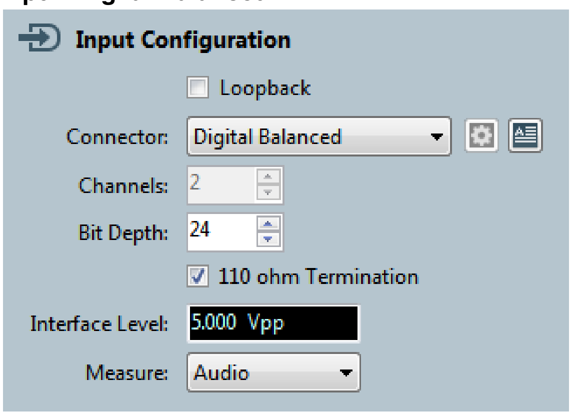
Measure

This menu is only available when the Advanced Master Clock and a digital I/O module that supports jitter are both fitted. The Measure menu allows you to select whether the audio signal or the jitter is routed to the analyzer.

- Audio
  This is the normal setting. The embedded audio is routed to all measurements and meter readings.
- Jitter (UI)
  This is the jitter setting. The jitter signal is extracted from the carrier and routed to all measurements and meter readings. Results are shown in Unit Interval (UI) units.
- Jitter (sec)
  This is an alternative Jitter setting. The jitter is measured as above, but the results are shown in seconds (s) units.

Read more about jitter on page 60.

Input: Digital Balanced



Digital Balanced enables the balanced digital electrical input, available on an XLR female connector on the instrument front panel. This input is optimized to accept signal at the professional signal level for balanced signals compatible with AES3 and IEC60958-4 (5 Vpp).

Channels

Audio channels are fixed at two.

Bit Depth

The Bit Depth control allows you to set the input bit depth (digital word length). The default setting is 24 bits, which is also the maximum setting. Minimum setting is 8 bits. The input signal is truncated (not dithered down) if Bit Depth is set to a lower value than Bit Depth of the signal.

We recommend that you use the default setting of 24 bits, unless you have a special application that requires that you mask out low-order bits.

110 Ohm Termination checkbox

When Digital Balanced is selected, the 110 Ohm Termination checkbox is available on the panel. When this checkbox is set (the default), the digital balanced input is terminated in a resistance of 110 Ω. This is the correct setting for terminating a balanced digital audio connection for AES3 / AES-EBU / IEC60958-4 professional formats. If you uncheck the checkbox, the digital balanced input is unterminated. This setting is useful when bridging a digital connection (as with a “Y” cable). A balanced digital connection should be properly terminated in 110 Ω at the end of its run. More settings are available in the Settings dialog.

Interface Level (Sequence Mode)

This field is only available in Sequence Mode when the Advanced Digital Input/Output module is fitted, and Input Configuration is set to Digital Unbalanced or Digital Balanced. It displays the peak-to-peak voltage level of the incoming digital interface waveform.

In Bench Mode, the same information is available from the Digital Interface Level meter.

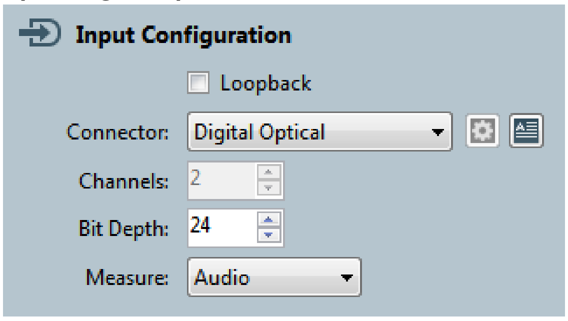
Measure

This menu is only available when the Advanced Master Clock and a digital I/O module that supports jitter are both fitted. The Measure menu allows you to select whether the audio signal or the jitter is routed to the analyzer.

- Audio
  This is the normal setting. The embedded audio is routed to all measurements and meter readings.
- Jitter (UI)
  This is the jitter setting. The jitter signal is extracted from the carrier and routed to all measurements and meter readings. Results are shown in Unit Interval (UI) units.
- Jitter (sec)
  This is an alternative Jitter setting. The jitter is measured as above, but the results are shown in seconds (s) units.

Read more about jitter on page 60.

Input: Digital Optical



Digital Optical enables the digital optical input, available on a Toslink connector on the instrument front panel.

Channels

Audio channels are fixed at two.

Bit Depth

The Bit Depth control allows you to set the input bit depth (digital word length). The default setting is 24 bits, which is also the maximum setting. Minimum setting is 8 bits. The input signal is truncated (not dithered down) if Bit Depth is set to a lower value than Bit Depth of the signal.

We recommend that you use the default setting of 24 bits, unless you have a special application that requires that you mask out low-order bits.

Measure

This menu is only available when the Advanced Master Clock and a digital I/O module that supports jitter are both fitted. The Measure menu allows you to select whether the audio signal or the jitter is routed to the analyzer.

- Audio
  This is the normal setting. The embedded audio is routed to all measurements and meter readings.
- Jitter (UI)
  This is the jitter setting. The jitter signal is extracted from the carrier and routed to all measurements and meter readings. Results are shown in Unit Interval (UI) units.
- Jitter (sec)
  This is an alternative Jitter setting. The jitter is measured as above, but the results are shown in seconds (s) units.

Read more about jitter on page 60.

14

DSIO

Introduction

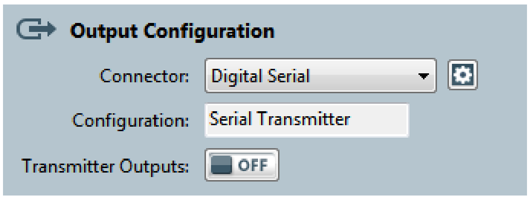
The Digital Serial Interface (or DSIO) option provides a flexible chip- or board-level serial input and output interface. DSIO is a hardware option available for the APx52x, 555 and 58x families of analyzers. The DSIO transmitter can be jittered when the Advanced Master Clock is fitted in the analyzer. Read more about jitter on page 60.

The DSIO allows you to transmit or receive Digital Serial data in a number of different formats. The DSIO physical layer provides connections for Master Clock, Frame Clock, Bit Clock, Channel Clock and 4 Data lines.

The DSIO Receiver Frame Clock is available for arbitrary use as a reference clock, when Input Configuration is not DSIO. See page 36.

Output Configuration

Digital Serial (when installed)



This selection enables the Digital Serial output available on the Transmitter HD-15 connector on the instrument front panel; also, the parallel buffered transmitter Monitor output is enabled.

Transmitter Outputs

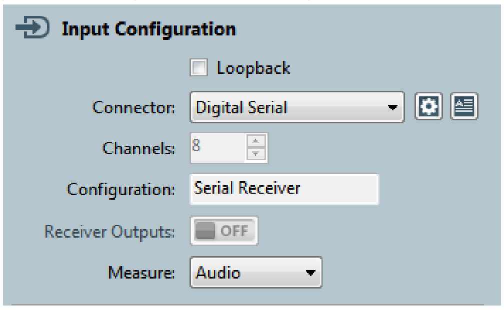
When Output Configuration: Digital Serial is selected, the Transmitter Outputs ON/OFF switch (duplicated in the Settings dialog) is available. This is a “kill switch” that, when OFF, removes all output voltages from the DSIO transmitter output connector pins.

Settings

The **Settings** button opens the DSIO output **Settings** dialog. Settings for DSIO are quite extensive. See Settings for Digital Serial output on page 100.

Input Configuration

Digital Serial (when installed)



This Connector selection enables the Digital Serial input available on the Receiver HD-15 connector on the instrument front panel; also, the parallel buffered receiver Monitor output is enabled.

Loopback

When the Loopback checkbox is set, the digital input is disconnected from the front panel input connector and is instead routed to the digital output circuits. This function is called GenMon in other Audio Precision products. Loopback allows you to view the signal present at the digital output on the analyzer meters or graphs.

Channels

The Channels setting is a read-only display on the Signal Path Setup panel. Go to the Input Settings (Digital Serial Receiver) dialog to set channels.

Settings

The **Settings** button opens the DSIO input **Settings** dialog. Settings for DSIO are quite extensive. See Settings for Digital Serial input on page 104.

Receiver Outputs

When Input Configuration: Digital Serial is selected, the Receiver Outputs ON/OFF switch (duplicated in the Settings dialog) is available. This is a “kill switch” that, when OFF, removes all output voltages from the DSIO receiver input connector pins.

Measure

The Measure menu is only available for a jitter-enabled DSIO module when the Advanced Master Clock is installed in the analyzer.

The Measure menu allows you to select whether the receiver audio signal or the bit clock jitter signal is routed to the analyzer.

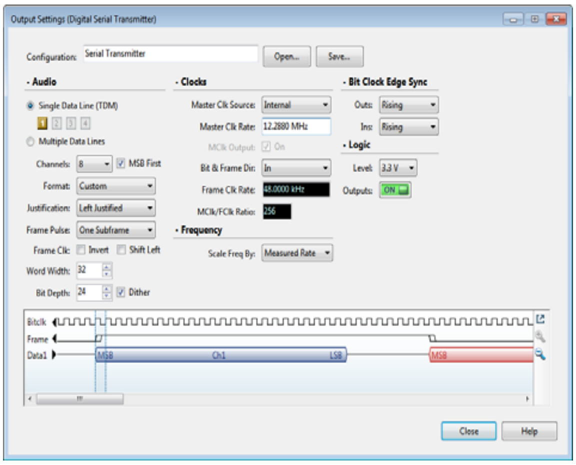
- Audio
  This is the normal setting. The audio signal is routed to all measurements and meter readings.
- Jitter (UI)
  This is the jitter setting. The jitter signal is extracted from receiver bit clock line and routed to all measurements and meter readings. Results are shown in Unit Interval (UI) units.
- Jitter (sec)
  This is an alternative Jitter setting. The jitter is measured as above, but the results are shown in seconds (s) units.

Read more about jitter on page 60.

Operation with absent or corrupt interface signal

When Input Configuration is set to a digital input and no signal is present, or if the signal is corrupt or out of range, the input receiver cannot lock (synchronize) to the interface signal, and no valid audio can be recovered. The sample rate indicator in the Status Bar will display an “unlocked” warning, and any measurement result referencing audio from the unlocked input will display an “----” (invalid) result. See page 560 for more about invalid results.

Output Settings for the Digital Serial Transmitter



This dialog allows you to set, save or load digital serial transmitter configurations. A dynamic timing diagram provides an on-screen display of your configuration settings. Transmitter connections are provided on an HD-15 connector, with four sets of Master Clock, Bit Clock, Channel Clock and Data connections. A monitor connector provides buffered outputs of these signals for oscilloscope monitoring.

The channel clock is provided as an aid for users monitoring on an oscilloscope. The channel clock signal is aligned with the data and has a rising edge with each channel change.

See illustrations of typical Digital Serial Input/Output configurations beginning on page 108.

Common Settings for Digital Serial Output

Most of the settings on this panel are common to all Digital Serial Output configurations, and are discussed first.

The settings in the transmitter Clocks area vary with clock configuration, and the four possible cases are discussed following these common settings.

Timing Diagram

The timing diagram at the bottom of the panel is dynamic, changing to illustrate your current settings. Use the zoom controls to contract or expand the horizontal axis of the display, or click the **Undock** button to display a large view.

Configuration

Type a name for your input configuration here.

Open

Click Open to view the Open Digital Serial Transmitter Settings dialog, where you can load saved *.stx configuration files.

Save

Click Save to view the Save Digital Serial Receiver Settings dialog, where you can save the current configuration as an *.stx file.

Audio

Data Lines

- Single (TDM) uses time division multiplexing to embed the audio data in a single data stream. Channels are embedded in the data stream sequentially, beginning with channel 1, followed by a channel 2 followed by channel 3, and so on. Although only a single, multichannel data stream can be configured, this stream can be sent on any combination (or none) of the four data line conductors in the physical layer. Use the Data Line buttons to select the output data line(s).
- Multiple uses 1, 2 or 4 separate data lines, each outputted on individual data line conductors in the physical layer. Each of the data lines carries 2 channels of audio data.

Channels

- When configured for a single data line (TDM), the Digital Serial Output can transmit 1, 2, 4, 8 or 16 audio data channels. Select the number of channels from the drop-down menu.
- When configured for multiple data lines, the Digital Serial Output can transmit 1, 2, 4, or 8 audio data channels. Select the number of channels from the drop-down menu.

MSB First

The active audio data is a string of bits, whose number is set by Bit Depth. When MSB First is set, the MSB (Most Significant Bit) leads the string. When MSB First is not set, the LSB (Least Significant Bit) leads the string.

Format

Set the transmitter format here.

- I2S (spoken “I-squared-S”) is an acronym for the Philips “inter IC sound” bus. I^2S is a common audio serial interface format. I2S sets the Frame Clock low for the first channel (left or “1”) and high for the second channel (right or “2”). If there are more channels, the Frame Clock stays high until the beginning of the next frame.

  I^2S data is offset by one bit clock period from the beginning of each word.
- DSP

  In the DSP (digital signal processing) format, the frame clock pulse is the same length as one bit clock period.

  Like I^2S, DSP offsets the data by one bit clock period from the beginning of each word.
- Custom

  The Custom selection allows you to set Justification and Frame Pulse in configurations not defined by I^2S or DSP formats.

Justification

- Left Justified

  Left Justified sets the active bits fully to the left (leading) edge of the data word. Padding bits, if any, trail the data.
- Right Justified

  Right Justified sets the active bits fully to the right (trailing) edge of the data word. Padding bits, if any, lead the data.

Frame Pulse

When Format is Custom, you can set the width of the Frame Clock Pulse here.

- One Bit Clock

  The Frame Clock pulse is one Bit Clock bit wide.
- One Subframe

  The Frame Clock pulse is one subframe wide. The subframe is the width of one data word. A subframe is a frame divided by the number of channels on that data line.
- 50% Duty Cycle

  The Frame Clock pulse is set to a 50% duty cycle, which makes the width of the Frame Clock pulse 1/2 the width of the frame.

Frame Clk: Invert

Check this box to invert the polarity of the Frame Clock.

Frame Clk: Shift Left

Check this box to shift the Frame Clock one Bit Clock pulse to the left, relative to the data.

Word Width

The word width is the width, in bits, of one data word, or subframe. A subframe is a frame divided by the number of channels. Set the word width here. Maximum is 128 bits; minimum is 8 bits. Word width cannot be less than bit depth.

Bit Depth

Set the bit depth (audio data length) here. Maximum is 32 bits; minimum is 8 bits.

When the **Bit Depth** is set to the same value of the **Word Width**, the entire bit allocation for the channel carries audio data. When **Bit Depth** is less than **Word Width**, only some of the bits carry data. The remaining bits are called pads or padding bits. The digital serial transmitter sets the logic state of padding bits to logical low.

Enable Dither

When the **Enable Dither** checkbox is checked, dither is **ON** (the default). When **ON**, dither is set according to the bit depth, using TPDF (triangular probability density function) dither at ±1 LSB.

Uncheck the checkbox to turn dither **OFF**.

Frequency

These fields only appear when the DSIO transmitter is slaved to the DUT. See Cases 2, 3 and 4, following.

Scale Frequency By:

Select a sample rate as the scaling factor for the embedded audio here. Typically, this is set to **Measured Rate**, which tracks the incoming **Frame Clock**, and scales the generator to give the requested signal frequency.

In some applications (for instance, when jitter is present), you may want to enter a **Fixed Rate**, so that changes in the measured rate don't affect the signal frequency and cause distortion.

Fixed Rate

When **Scale Freq. By** is set to **Fixed Rate**, enter the reference sample rate here.

Bit Clock Edge Sync

These selects whether the leading edge of the frame (and data) are synchronized with the rising edge or the falling edge of the bit clock, for DSIO receiver inputs and outputs.

Outs

- Rising Edge
  For clocks that are set to OUT, synchronization is at the rising edge of the clock signal.
- Falling Edge
  For clocks that are set to **OUT**, synchronization is at the falling edge of the clock signal.

Ins

- Rising Edge
  For clocks that are set to **IN**, synchronization is at the rising edge of the clock signal.
- Falling Edge
  For clocks that are set to IN, synchronization is at the falling edge of the clock signal.

Logic Level

This sets the nominal voltage level for the DSIO clock and data signals at the HD-15 receiver connections. Choose the level compatible with your DUT. Choices are 1.8 V, 2.5 V and 3.3 V. The logic level for the monitor connections is fixed at 3.3 V.

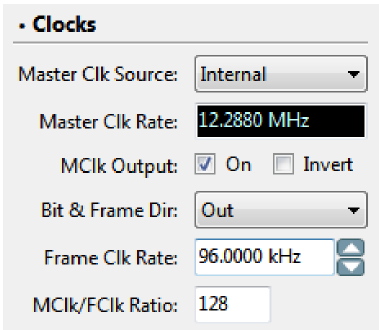
Transmitter Outputs

This is a "kill switch" that, when **OFF**, removes all output voltages from the DSIO output connector pins. This switch is duplicated on the Signal Path Setup dialog.

Transmitter Clocks

The **visibility** and availability of controls and readings in the transmitter Clocks area depend upon the settings of **Master Clock Source** and **Bit & Frame Clock Direction**. The discussions here are separated into the four possible configurations.

Case 1: Master Clock Internal, Bit & Frame Out



Transmitter Settings: Master Clk Internal, Bit & Frame Out

Master Clock Source: Internal

The bit and frame clocks are generated internally from the APx master clock. In this configuration, the APx DSIO transmitter is the master, and the DUT is slaved to the APx.

Master Clk Rate

This field displays the rate of the Master Clock signal. Master Clock rate is the product of the Sample Rate multiplied by the Ratio value. For 1, 2, or 4 channels, maximum Master Clock rate is 55.296 MHz; for 8 or 16 channels, maximum Master Clock rate is 49.152 MHz.

MClk Output

When this box is checked, the master clock signal appears on the DSIO transmitter output HD-15 connector, pin 1.

Invert

This checkbox is shown if **MClk Output** is checked. Checking the **Invert** checkbox inverts the master clock output signal. When not inverted, the rising edge of the bit clock is synchronized with the rising edge of the master clock. When inverted, the rising edge of the bit clock is synchronized with the falling edge of the master clock.

Bit & Frame Clock Dir: Out

The bit and frame clocks are divided internally from the APx master clock. In this configuration, the DUT is slaved to the APx DSIO transmitter.

Frame Clk Rate

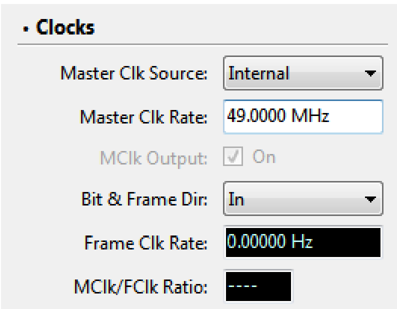
Set the frame clock rate (audio data sample rate) here. Minimum is 4 kHz. For 1, 2, or 4 channels, maximum frame clock rate is 216 kHz; for 8 or 16 channels, maximum frame clock rate is 192 kHz. Actual rate is dependent upon other settings on this panel.

MClk/FClk Ratio

The ratio is a factor in determining the rate of the master clock output signal. Your DUT will accept a specified range of Master Clock frequencies. Once you have chosen your sample rate, enter a Ratio here to set the Master Clock Rate, displayed below, to an acceptable frequency.

Note: Word width, channel count, sample rate, master clock minimum and maximum frequencies and other synchronization requirements constrain Ratio values. Common values for Ratio are 128, 256, or 512.

Case 2: Master Clock Internal, Bit & Frame In



Transmitter Settings: Master Clk Internal, Bit & Frame In

Master Clock Source: Internal

The Master Clock is output to the DUT.

Master Clk Rate

Set the Master Clock rate here. For 1, 2, or 4 channels, maximum Master Clock rate is 55.296 MHz; for 8 or 16 channels, maximum Master Clock rate is 49.152 MHz.

MClk Output

MClk Output is forced On in this configuration. The master clock signal appears on the DSIO transmitter output HD-15 connector, pin 1.

Invert

This checkbox is not available in this mode.

Bit & Frame Clock Dir: In

The bit and frame clocks are divided from the APx Master Clock externally in the DUT. In this configuration, the APx transmitter is slaved to the DUT.

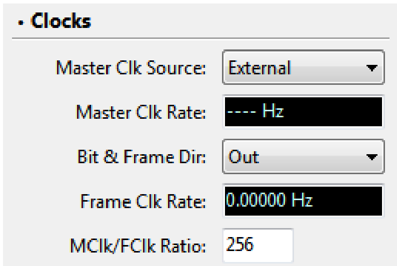
Frame Clk Rate

This field displays the rate of the Frame Clock signal.

MClk/FClk Ratio

This field displays the MClk/FClk Ratio.

Case 3: Master Clock External, Bit & Frame Out



Transmitter Settings: Master Clk External, Bit & Frame Out

Master Clock Source: External

The APx DSIO Master Clock is not used.

Master Clk Rate

This field is not available in this mode.

MClk Output

This checkbox is not available in this mode.

Invert

This checkbox is not available in this mode.

Bit & Frame Clock Dir: In

The bit and frame clocks are provided from the DUT. In this configuration, the APx transmitter is slaved to the DUT.

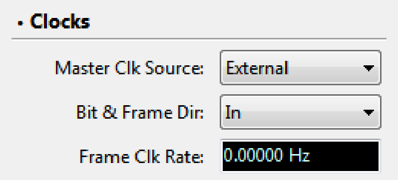
Frame Clk Rate

This field displays the rate of the Frame Clock signal.

MClk/FClk Ratio

This field is not available in this mode.

Case 4: Master Clock External, Bit & Frame In



Transmitter Settings: Master Clk External, Bit & Frame In

Master Clock Source: External

The APx DSIO Master Clock is not used.

Master Clk Rate

This field displays the rate of the Master Clock signal.

MClk Output

This checkbox is not available in this mode.

Invert

This checkbox is not available in this mode.

Bit & Frame Clock Dir: Out

The bit and frame clocks are divided internally from the external master clock. In this configuration, the APx DSIO transmitter is slaved to the DUT.

Frame Clk Rate

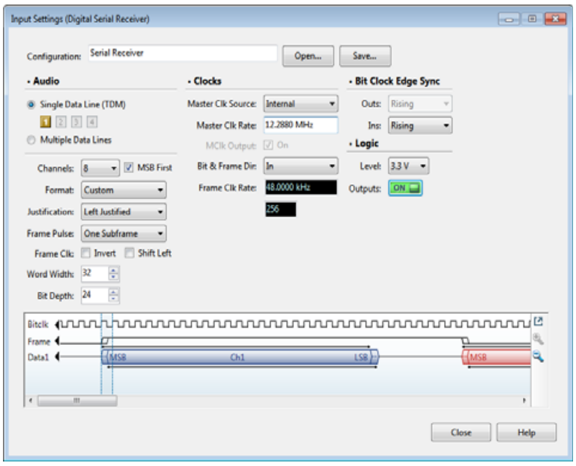
This field displays the rate of the Frame Clock signal.

MClk/FClk Ratio

The ratio is a factor in determining the rate of the Frame Clock signal. Enter a Ratio here to divide the external master clock to an acceptable Frame Clock Rate.

Note: Word width, channel count, sample rate, master clock minimum and maximum frequencies and other synchronization requirements constrain Ratio values. Common values for Ratio are 128, 256, or 512.

Input Settings for the Digital Serial Receiver



This dialog allows you to set, save or load digital serial receiver configurations. A dynamic timing diagram provides an on-screen display of your configuration settings. Receiver connections are provided on an HD-15 connector, with Master Clock, Frame Clock, Bit Clock, Channel Clock and four Data line connections. A monitor connector provides buffered outputs of these signals for oscilloscope monitoring.

The channel clock is provided as an aid for users monitoring on an oscilloscope. The channel clock signal is aligned with the data and has a rising edge with each channel change.

The DSIO Receiver Frame Clock is available for arbitrary use as a reference clock, when Input Configuration is not DSIO. See page 36.

See illustrations of typical Digital Serial Input/Output configurations beginning on page 108.

Common Settings for Digital Serial Input

Most of the settings on this panel are common to all Digital Serial Input configurations, and are discussed first.

The settings in the receiver Clocks area vary with clock configuration, and the four possible cases are discussed following these common settings.

Timing Diagram

The timing diagram at the bottom of the panel is dynamic, changing to illustrate your current settings. Use the zoom controls to contract or expand the horizontal axis of the display, or click the **Undock** button to display a large view.

Configuration

Type a name for your input configuration here.

Open

Click Open to view the Open Digital Serial Receiver Settings dialog, where you can load saved *.srx configuration files.

Save

Click Save to view the Save Digital Serial Receiver Settings dialog, where you can save the current configuration as an *.srx file.

Audio

Data Lines

- Single (TDM) uses time division multiplexing to read the audio data in a single data stream. Channels are read from the data stream sequentially, beginning with channel 1, followed by a channel 2 followed by channel 3, and so on. Use the Data Line buttons to select the input data line.
- Multiple uses 1, 2 or 4 separate data lines, each received on individual data line conductors in the physical layer. Each of the multiple data lines carries 2 channels of audio data.

Channels

- When configured for a single data line (TDM), the Digital Serial Input can receive 1, 2, 4, 8 or 16 audio data channels. Select the number of channels from the drop-down menu.
- When configured for multiple data lines, the Digital Serial Input can receive 1, 2, 4, or 8 audio data channels. Select the number of channels from the drop-down menu.

MSB First

The active audio data is a string of bits, whose number is the Bit Depth. When MSB First is set in the receiver, the MSB (Most Significant Bit) is assumed to lead the string. When MSB First is not set, the LSB (Least Significant Bit) is assumed to lead the string.

Format

Set the receiver format here.

- I2S (spoken “I-squared-S”) is an acronym for the Philips “inter IC sound” bus. I^2S is a common audio serial interface format. I2S sets the input to receive I^2S formatted signals, where the Frame Clock low for the first channel (left or “1”) and high for the second channel (right or “2”). If there are more channels, the Frame Clock stays high until the beginning of the next frame. I^2S data is offset by one bit clock period from the beginning of each word.
- DSP
  In the DSP (digital signal processing) format, the frame clock pulse is the same length as one bit clock period. Like I^2S, DSP offsets the data by one bit clock period from the beginning of each word. Choose this setting to configure the receiver for the DSP format.
- Custom
  The Custom selection allows you to configure the receiver for Justification and Frame Pulse in configurations not defined by I^2S or DSP formats.

Justification

- Left Justified
  Left Justified configures the receiver for active bits fully to the left (leading) edge of the data word.
- Right Justified
  Right Justified configures the receiver for active bits fully to the right (trailing) edge of the data word.

Frame Pulse

When Format is Custom, you can set the receiver for the width of the Frame Clock Pulse here.

- One Bit Clock
  The Frame Clock pulse is one Bit Clock bit wide.
- One Subframe
  The Frame Clock pulse is one subframe wide. The subframe is the width of one data word. A subframe is a frame divided by the number of channels on that data line.
- 50% Duty Cycle
  The Frame Clock pulse is set to a 50% duty cycle, which makes the width of the Frame Clock pulse 1/2 the width of the frame.

Invert Frame Clk

Check this box to set the receiver for inverted polarity of the Frame Clock.

Word Width

The word width is the width, in bits, of one data word, or subframe. A subframe is a frame divided by the number of channels. Set the received word width here. Maximum is 128 bits; minimum is 8 bits. Word width cannot be less than bit depth.

Bit Depth

Set the received bit depth (audio data length) here. Maximum is 32 bits; minimum is 8 bits. When the Bit Depth is set to the same value of the Word Width, the entire bit allocation for the channel carries audio data. When Bit Depth is less than Word Width, only some of the bits carry data. The remaining bits are called pads or padding bits.

Bit Clock Edge Sync

These selects whether the leading edge of the frame (and data) are synchronized with the rising edge or the falling edge of the bit clock, for DSIO receiver inputs and outputs.

Outs

- Rising Edge
  For clocks that are set to OUT, synchronization is at the rising edge of the clock signal.
- Falling Edge
  For clocks that are set to **OUT**, synchronization is at the falling edge of the clock signal.

Ins

- Rising Edge
  For clocks that are set to **IN**, synchronization is at the rising edge of the clock signal.
- Falling Edge
  For clocks that are set to IN, synchronization is at the falling edge of the clock signal.

Logic Level

This sets the nominal voltage level for the DSIO clock and data signals at the HD-15 receiver connections. Choose the level compatible with your DUT. Choices are 1.8 V, 2.5 V and 3.3 V. The logic level for the monitor connections is fixed at 3.3 V.

Receiver Outputs

This is a “kill switch” that, when **OFF**, removes all output voltages from the DSIO input connector pins. This switch is duplicated on the Signal Path Setup dialog.

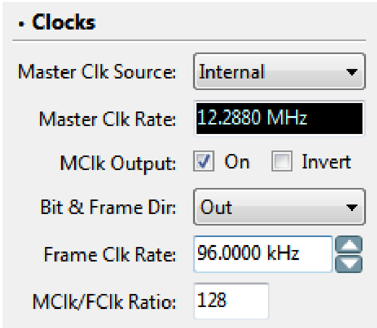
Audio Coupling

Sets the digital AC or DC coupling for the received embedded audio data.

Receiver Clocks:

The visibility and availability of controls and readings in the receiver **Clocks** area depend upon the settings of **Master Clock Source** and **Bit & Frame Clock Direction**. The explanations here are segmented into the four possible combinations of these settings.

Case 1: Master Clock Internal, Bit & Frame Out



Receiver Settings: Master Clk Internal, Bit & Frame Out

Master Clock Source: Internal

The bit and frame clocks are generated internally from the APx master clock. In this configuration, the APx DSIO transmitter is the master, and the DUT is slaved to the APx.

Master Clk Rate

This field displays the rate of the Master Clock signal. Master Clock rate is the product of the Sample Rate multiplied by the Ratio value. For 1, 2, or 4 channels, maximum Master Clock rate is 55.296 MHz; for 8 or 16 channels, maximum Master Clock rate is 49.152 MHz.

MClk Output

When this box is checked, the master clock signal appears on the DSIO transmitter output HD-15 connector, pin 1.

Invert

This checkbox is shown if **MClk Output** is checked. Checking the **Invert** checkbox inverts the master clock output signal. When not inverted, the rising edge of the bit clock is synchronized with the rising edge of the master clock. When inverted, the rising edge of the bit clock is synchronized with the falling edge of the master clock.

Bit & Frame Clock Dir: Out

The bit and frame clocks are divided internally from the APx master clock. In this configuration, the DUT is slaved to the APx DSIO transmitter.

Frame Clk Rate

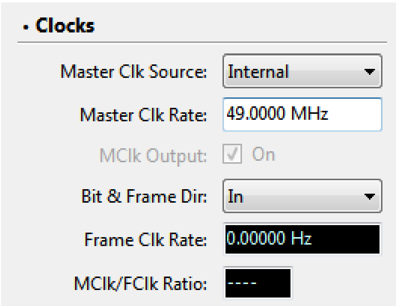
Set the frame clock rate (audio data sample rate) here. Minimum is 4 kHz. For 1, 2, or 4 channels, maximum frame clock rate is 216 kHz; for 8 or 16 channels, maximum frame clock rate is 192 kHz. Actual rate is dependent upon other settings on this panel.

MClk/FClk Ratio

The ratio is a factor in determining the rate of the master clock output signal. Your DUT will accept a specified range of Master Clock frequencies. Once you have chosen your sample rate, enter a Ratio here to set the Master Clock Rate, displayed below, to an acceptable frequency.

> *Note: Word width, channel count, sample rate, master clock minimum and maximum frequencies and other synchronization requirements constrain Ratio values. Common values for Ratio are 128, 256, or 512.*

Case 2: Master Clock Internal, Bit & Frame In



Receiver Settings: Master Clk Internal, Bit & Frame In

Master Clock Source: Internal

The Master Clock is output to the DUT.

Master Clk Rate

Set the Master Clock rate here. For 1, 2, or 4 channels, maximum Master Clock rate is 55.296 MHz; for 8 or 16 channels, maximum Master Clock rate is 49.152 MHz.

MClk Output

MClk Output is forced On in this configuration. The master clock signal appears on the DSIO transmitter output HD-15 connector, pin 1.

Invert

This checkbox is not available in this mode.

Bit & Frame Clock Dir: In

The bit and frame clocks are divided from the APx Master Clock externally in the DUT. In this configuration, the APx transmitter is slaved to the DUT.

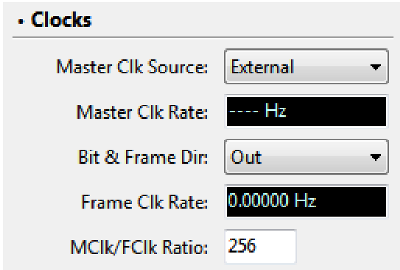
Frame Clk Rate

This field displays the rate of the Frame Clock signal.

MClk/FClk Ratio

This field displays the MClk/FClk Ratio.

Case 3: Master Clock External, Bit & Frame Out



Receiver Settings: Master Clk External, Bit & Frame Out

Master Clock Source: External

The APx DSIO Master Clock is not used.

Master Clk Rate

This field displays the rate of the Master Clock signal.

MClk Output

This checkbox is not available in this mode.

Invert

This checkbox is not available in this mode.

Bit & Frame Clock Dir: Out

The bit and frame clocks are divided internally from the external master clock. In this configuration, the DUT is slaved to the APx DSIO transmitter.

Frame Clk Rate

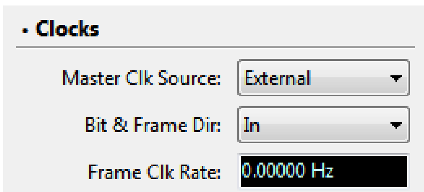
This field displays the rate of the Frame Clock signal.

MClk/FClk Ratio

The ratio is a factor in determining the rate of the Frame Clock signal. Enter a Ratio here to divide the external master clock to an acceptable Frame Clock Rate.

> *Note: Word width, channel count, sample rate, master clock minimum and maximum frequencies and other synchronization requirements constrain Ratio values. Common values for Ratio are 128, 256, or 512.*

Case 4: Master Clock External, Bit & Frame In



Receiver Settings: Master Clk External, Bit & Frame In

Master Clock Source: External

The APx DSIO Master Clock is not used.

Master Clk Rate

This field is not available in this mode.

MClk Output

This checkbox is not available in this mode.

Invert

This checkbox is not available in this mode.

Bit & Frame Clock Dir: In

The bit and frame clocks are provided from the DUT. In this configuration, the APx transmitter is slaved to the DUT.

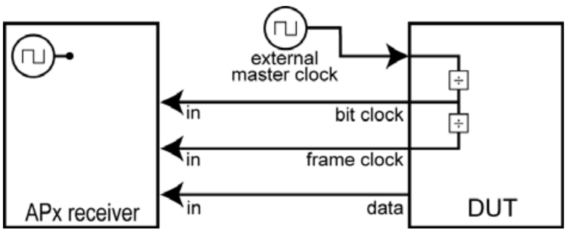
Frame Clk Rate

This field displays the rate of the Frame Clock signal.

MClk/FClk Ratio

This field is not available in this mode.

Typical DSIO configurations



DSIO Receiver configuration 1

This is a receiver configuration.

The DUT provides the master clock and the bit and frame clock dividers.

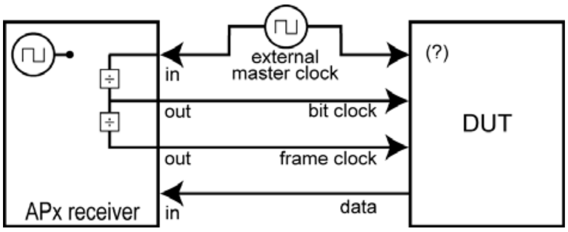
DUT sends bit clock, frame clock and data.

Receiver Settings

Clocks

Set Master Clk Source to External.

Set Bit & Frame Dir to In.



DSIO Receiver Configuration 2

This is a receiver configuration.

The DUT provides the master clock, and uses the bit and frame clock dividers in the APx. The DUT may or may not use the master clock (indicated by "?"), which does not affect the APx configuration.

The DUT sends master clock and data; the APx sends bit clock and frame clock.

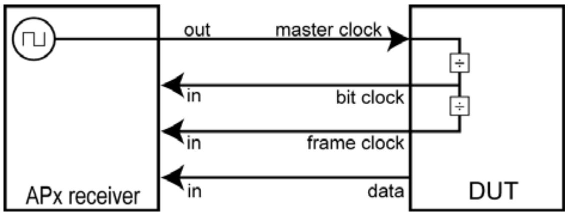
Receiver Settings

Clocks

Set Master Clk Source to External.

Set Bit & Frame Dir to Out.

Enter a value in MClk/FClk Ratio to determine the sample rate.



DSIO Receiver Configuration 3

This is a receiver configuration.

The DUT provides the bit and frame clock dividers, but has no master clock.

APx sends master clock; the DUT sends bit clock, frame clock and data.

Receiver Settings

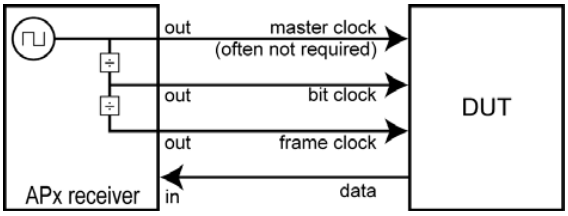
Clocks

Set Master Clk Source to Internal.

Set Bit & Frame Dir to In.

Set Frame Rate to the desired value.

Enter a value in MClk/FClk Ratio to determine the sample rate.



DSIO Receiver Configuration 4

This is a receiver configuration.

The APx provides the master clock and the bit and frame clock dividers.

The DUT sends data; the APx send master clock, bit clock and frame clock. In some DUT configurations, only the bit clock and the frame clock are required.

Receiver Settings

Clocks

Set **Master Clk Source** to **Internal**.

If the master clock is to be used, put a check in the **MClk Output On** checkbox.

If the master clock is not to be used, clear the check from the **On** checkbox.

Set **Bit & Frame Dir** to **Out**.

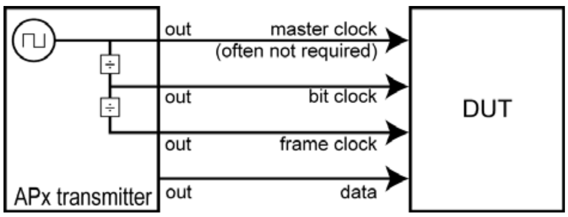
Set **Frame Rate** to the desired value.

The **Frame Rate** multiplied by the number of **Channels** multiplied by the **Word Width** determines the bit clock rate.

If master clock is on, enter a value in **MClk/FClk Ratio** to determine the sample rate.

Check the **Invert** checkbox to change the polarity of the master clock signal. This effectively toggles between rising edge/ falling edge synchronization.

Note: this configuration is identical to DSIO Transmitter Configuration 1, except for data direction.



DSIO Transmitter Configuration 1

The APx provides the master clock and the bit and frame clock dividers.

The APx sends master clock, bit clock, frame clock and data. In some DUT configurations, the master clock is not required.

Transmitter Settings

Clocks

If the master clock is to be used, put a check in the **MClk Output On** checkbox.

If the master clock is not to be used, clear the check from the **On** checkbox.

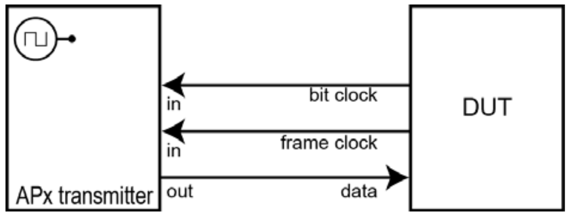
Set **Frame Rate** to the desired value.

The **Frame Rate** multiplied by the number of **Channels** multiplied by the **Word Width** determines the bit clock rate.

If master clock is on, enter a value in **MClk/FClk Ratio** to determine the sample rate.

Check the **Invert** checkbox to change the polarity of the master clock signal. This effectively toggles between rising edge/ falling edge synchronization.

Note: this configuration is identical to DSIO Receiver Configuration 4, except for data direction.



DSIO Transmitter Configuration 2

The DUT provides the bit and frame clocks.

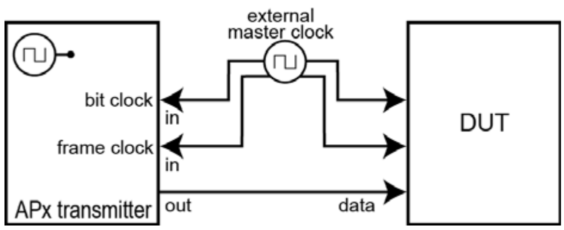
The APx sends data.

Transmitter Settings

Clocks

If the master clock is to be used, put a check in the **MClk Output On** checkbox.

Bit/Frame Clks: In



DSIO Transmitter Configuration 3

An external clock provides the bit and frame clocks to both the APx and the DUT

The APx sends data.

Transmitter Settings

Clocks

Bit/Frame Clks: In

Note: this configuration is identical to DSIO Transmitter Configuration 2, but using an external master clock source.

15

HDMI+ARC

Introduction

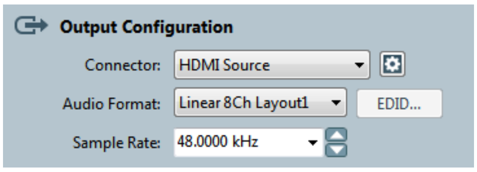
HDMI (High Definition Multimedia Interface) is an interface designed to carry high-bandwidth digital streams providing an audio/video interface that includes content protection and a bi-directional channel for interaction with connected electronic devices.

ARC (Audio Return Channel) provides an additional digital audio channel to the HDMI interconnection that can simplify interface cabling in typical applications, for user convenience. HDMI+ARC is a current hardware option available for APx525, 582 and 585 families of analyzers.

APx Output and Input configurations and settings for HDMI and ARC are independent selections. Configuration and settings documentation for audio over HDMI begins here; documentation for ARC begins on page 117.

HDMI Source Output Configuration

Output: HDMI Source



The HDMI Source selection is only available for an APx analyzer fitted with the HDMI option (model 112) or HDMI+ARC (model 114 or 214) option.

This selection embeds the output audio in the HDMI stream, available on the “SOURCE” type A HDMI connector on the instrument front panel. When HDMI Source is selected, the Audio Format and Sample Rate controls are available. The **EDID** button is available when EDID data from a downstream device is present at the AUX OUT HDMI connector.

Settings

The **Settings** button opens the HDMI output **Settings** dialog. Settings for HDMI are quite extensive. See Output Settings for HDMI Source on page 112.

Audio Format

Choose the HDMI audio format here.

- **Layout 0 LPCM 2ch**
  Layout 0 is only 2 channel, but can output any supported sample rate in any supported video format.
- **Layout 1 LPCM 8ch**
  Layout 1 is 8 channel. Supported sample rates are constrained by selected video format. See Settings to choose video format and for information about supported sample rates.

EDID

This button is only available when there is EDID information at the HDMI SOURCE connector. It opens the EDID Viewer, which displays a property grid that shows the EDID data from the downstream device at the SOURCE HDMI connector. The EDID data can be viewed, saved to a file and reloaded using the EDID Editor (see page 116), for use as the APx instrument EDID sent to an upstream device at the SINK HDMI connector.

Sample Rate (duplicated on the Settings dialog)

To set the output sample rate (Fs) for the embedded digital audio, click the up/down arrows to select a standard rate, or enter an arbitrary value in the **Sample Rate** field. Minimum setting is 30.7 kHz; maximum is 192 kHz. Default is 48 kHz. Supported sample rates are constrained to certain ranges by the selected video format. See **Settings** to choose video format and for information about supported sample rates.

Settings for HDMI are quite extensive. See Output Settings for HDMI Source on page 112.

HDMI Sink Input Configuration

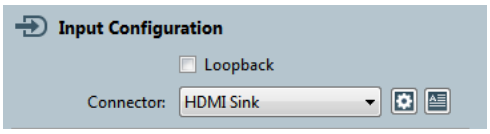
In the Input Configuration panel, you select the number of input channels operating for your test and the connectors you will be using. You can choose references and names for your test input channels.

Loopback

Loopback is only available when Output Configuration is set to a digital output.

When the Loopback checkbox is set, the digital input is disconnected from the front panel input connector and is instead routed to the digital output circuits. This function is called GenMon in other Audio Precision products. Loopback allows you to view the signal present at the digital output on the analyzer meters or graphs.

Input: HDMI Sink



The HDMI Sink selection is only available for an APx analyzer fitted with the HDMI option (model 112) or HDMI+ARC (model 114) option.

HDMI Sink de-embeds the input audio in the HDMI stream presented at the SINK connector.

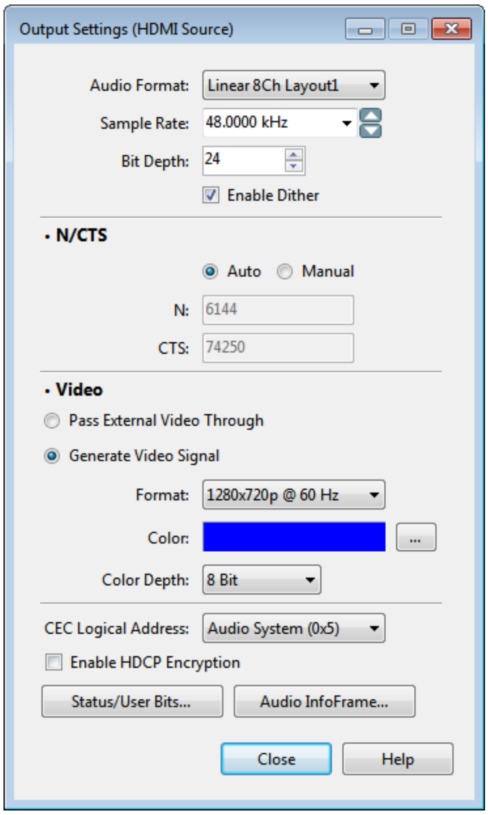
Settings

The **Settings** button opens the HDMI input **Settings** dialog. Settings for HDMI are quite extensive. See Input Settings for HDMI Sink on page 114.

Operation with absent or corrupt interface signal

When Input Configuration is set to a digital input and no signal is present, or if the signal is corrupt or out of range, the input receiver cannot lock (synchronize) to the interface signal, and no valid audio can be recovered. The sample rate indicator in the Status Bar will display an “unlocked” warning, and any measurement result referencing audio from the unlocked input will display an “----” (invalid) result. See page 560 for more about invalid results.

Output Settings for HDMI Source



Audio Format

Audio Format appears on both Signal Path Setup and here in the Settings dialog. Choose the HDMI audio format here.

Layout 0 LPCM 2ch

Layout 0 is only 2 channel, but can output any supported sample rate in any supported video format.

Layout 1 LPCM 8ch

Layout 1 is 8 channel. Supported sample rates are constrained by selected video format. See Video Settings > Format, below, to choose video format and for information about supported sample rates.

Sample Rate

To set the output sample rate (Fs) for the embedded digital audio, click the up/down arrows to select a standard rate, or enter an arbitrary value in the **Sample Rate** field. Minimum setting is 30.7 kHz; maximum is 192 kHz. Default is 48 kHz. Supported sample rates are constrained to certain ranges by the selected video format. See Settings to choose video format and for information about supported sample rates. See **Video Settings: Format** below.

Bit Depth

Bit Depth has the same function for HDMI signal generation as it does in the other digital output modes. Return to Advanced Settings for Signal Path Setup to see this information.

Enable Dither

Dither is enabled by default, and should remain enabled for all ordinary audio signals. Dither is disabled for signals that must be bit-accurate, such as Walking Zeros, Walking Ones, Constant Value and Bit test Random.

N/CTS

At the HDMI transmitter, the audio sample is rate embodied in the relationship between the values of N (the video clock) and CTS (the cycle time stamp, sent as metadata). The downstream HDMI receiver uses N/CTS to re-create the audio clock.

Auto

In Auto, N/CTS follows the HDMI standard for all supported video resolutions and audio rates.

Manual

In Manual, you are allowed to enter non-standard values as impairments to test the robustness of downstream devices.

Video Settings

HDMI is a video interface with embedded audio. The video content for the APx HDMI generator output can be from either an external video signal brought in at the HDMI AUX IN connector, or from an internal video generator (the default). Select the video source from these choices:

Pass External Video Through

When Pass External Video Through is selected, the video signal at the HDMI AUX IN connector is passed to the HDMI SOURCE connector. Any audio on the external HDMI signal is stripped off and replaced by audio generated in APx500. The APx generator audio format is constrained by the video format of the external signal. The HDCP encryption state is passed through unchanged.

Generate Video Signal (the default)

When Generate Video Signal is selected, the video content for the HDMI SOURCE is generated within the Audio Precision HDMI module. Format, color, color depth and HDCP encryption of this signal are set in the next controls.

- Format:
  One of 14 video formats can be set here, ranging from 640x480p @ 60 Hz to 1920x1080p @ 24 Hz. Note that only high-definition video formats support multi-channel high-sample-rate embedded audio. Audio Format and Video Format selections will constrain HDMI channel count and sample rate.

Refer to the chart of HDMI channel count and sample rates by audio and video formats on page 116.

- Color:
  Select video content color here. Default color is a blue screen. Click the browser button to select other colors.
- Color Depth:
  Select video content color depth here. Choices are 8 bit, or Deep Color modes 10 bit or 12 bit.

HDCP Encryption

For internally-generated video, HDCP encryption may be enabled or disabled by clicking the checkbox.

For externally-provided video, the HDCP encryption state is passed through to the downstream device unchanged. The HDCP Encryption checkbox is unavailable in this mode.

About HDCP encryption

- HDCP (High-bandwidth Digital Content Protection) can be used to protect content transmitted on an HDMI interface.
- When enabled, non-HDCP-compliant downstream devices will not be able to establish a connection. A blank screen or error message may be displayed.

Status/User Bits

This button opens a dialog where you can set the Status Bits and User Bits embedded in the audio carried in the HDMI stream. See More About Metadata on page 354.

Audio InfoFrame

This button opens a dialog where you can set the Audio InfoFrame metadata carried in the HDMI stream. For information about reading Audio InfoFrame data, see page 33.

Audio streams in HDMI contain metadata in the form of status bits, user bits, and for coded audio, burst information. The HDMI Audio InfoFrame contains additional information about the audio stream.

Setting Audio InfoFrame

When Output Configuration is set to HDMI Source, click Settings and then Audio InfoFrame.

Auto

When Auto is selected, APx500 sets the HDMI Audio InfoFrame to default values.

When Auto is not selected, Audio InfoFrame fields can be set to any available value.

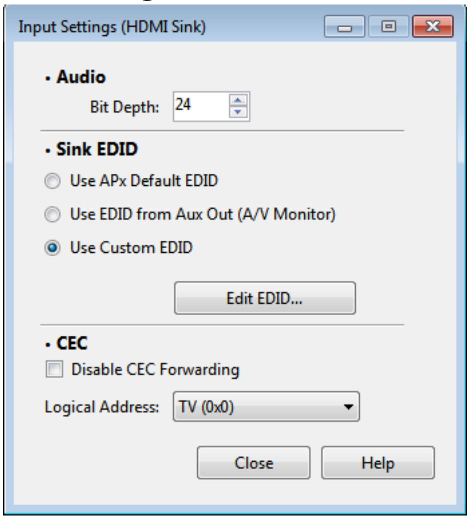
Plain Text display and Hex display

Audio InfoFrame fields can be set from the grid display, reading the plain text interpretations and choosing settings from drop-down menus in the grid. Audio InfoFrame data can also be set by entering hex values for selected bytes in the Audio InfoFrame hex display. These two displays are coupled, and a change in one will be reflected by the corresponding change in the other.

Refer to HDMI.org and the standards document CEA-861-DB for detailed information about Audio InfoFrame and the HDMI transport stream.

Read more about Audio InfoFrame data on page 356.

Input Settings for HDMI Sink



Disable CEC Forwarding

By default, CEC commands from devices at the HDMI Sink or AUX OUT connectors are forwarded between these devices. Select Disable CEC Forwarding to prevent this communication.

About CEC

- CEC (Consumer Electronics Control) is an optional HDMI protocol that uses a dedicated bidirectional bus to communicate consumer control commands (such as volume or channel change) between HDMI-connected devices. Read more about CEC on page 120.

SINK EDID

About EDID and DDC

- APx500 supports the Enhanced Display Data Channel (E-DDC, or simply DDC), which is used by the HDMI source device to read the E-EDID data from the HDMI sink device to learn what audio/video formats it supports. APx supports EDID and E-EDID for HDMI.
- EDID (Extended Display Identification Data) and E-EDID (Enhanced Extended Display Identification Data) are data stored in downstream (sync) devices and communicated to upstream (source) devices.
- EDID contains information about the video and audio capabilities of the downstream device, so that the source device can format the content properly. This means, for example, that a downstream 720x480p video monitor with only two audio channels will cause an upstream surround decoder or Blu-ray player to provide a 720x480p video signal with a stereo downmix, when connected to this device.

EDID 1.4

- With APx500 v 3.1, EDID for HDMI 1.4 is supported, in addition to the previously supported EDID 1.3. Note that in EDID 1.4, some of the EDID bytes are interpreted differently than in EDID 1.3. For more information, see the following standard, available at vesa.org.
- VESA ENHANCED EXTENDED DISPLAY IDENTIFICATION DATA STANDARD (Defines EDID Structure Version 1, Revision 4).

Use APx Default EDID

When this button is set, the APx default EDID is sent to the upstream device connected at the HDMI SINK connector. The APx instrument is configured with default EDID data that will produce expected results for many DUT / instrument / monitor setups. You can view these default settings by using the Edit EDID button, below.

Use EDID from AUX OUT (monitor)

By default, EDID read from the downstream display device on AUX OUT is NOT forwarded to the upstream device connected to SINK. This configuration allows you to analyze the full high-definition audio embedded in the SINK signal without constraining it to the limits of the display device.

To constrain the upstream device on SINK to the capabilities of the monitor connected to AUX OUT, check the Copy EDID box.

Use Custom EDID

In some cases you may want to send custom EDID data to the upstream device. Open the EDID editor by clicking the Edit EDID button. Read more about the EDID Editor starting at page 116.

Embedded Audio

Bit Depth

The Bit Depth control allows you to set the input bit depth (digital word length). The default setting is 24 bits, which is also the maximum setting. Minimum setting is 8 bits. The input signal is truncated (not dithered down) if Bit Depth is set to a lower value than Bit Depth of the signal.

We recommend that you use the default setting of 24 bits, unless you have a special application that requires that you mask out low-order bits.

Coupling

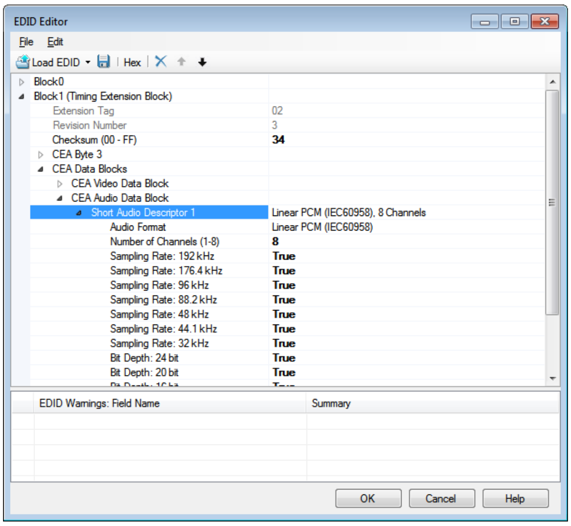
Input signals can be analyzed and displayed in one of two coupling modes:

- AC
  AC coupling blocks any dc signal component from analysis, displaying results for only the ac components. AC coupling is the default setting.
- DC
  DC coupling includes the dc signal component in analysis and display.

Audio Format	*Video Format*	*Channel Count*	*Sample Rates*
Linear 2Ch Layout 0	any	2	30.7 kHz to 33.3 kHz 42.3 kHz to 45.9 kHz 46.0 kHz to 50.0 kHz 84.6 kHz to 91.8 kHz 92.0 kHz to 100.0 kHz 169.2 kHz to 183.6 kHz 184.0 kHz to 192.0 kHz
Linear 8Ch Layout 1	640x720p @ 60 Hz 720x480p @ 60 Hz 720x576p @ 50 Hz	8	30.7 kHz to 33.3 kHz 42.3 kHz to 45.9 kHz 46.0 kHz to 50.0 kHz
Linear 8Ch Layout 1	720x240p @ 60 Hz 720x288p @ 50 Hz 720x480i @ 60 Hz 720x576i @ 50 Hz	8	30.7 kHz to 33.3 kHz 42.3 kHz to 45.9 kHz 46.0 kHz to 50.0 kHz 84.6 kHz to 91.8 kHz
Linear 8Ch Layout 1	All 1280x720 and 1920x1080 formats	8 (Note: Although the HDMI interface can carry 8 channels at all these sample rates, Blu-ray Discs are limited to 6 channels at 192 kHz sample rate.)	30.7 kHz to 33.3 kHz 42.3 kHz to 45.9 kHz 46.0 kHz to 50.0 kHz 84.6 kHz to 91.8 kHz 92.0 kHz to 100.0 kHz 169.2 kHz to 183.6 kHz 184.0 kHz to 192.0 kHz

HDMI audio constraints by format

The EDID Viewer and the EDID Editor



The EDID Editor.
The EDID Viewer displays information in the same way.

The EDID Viewer

The APx500 EDID Viewer is a property grid that displays the EDID settings received from the downstream device connected to the instrument HDMI SOURCE connector. These properties cannot be edited, since they come from the downstream DUT, but they can be viewed and saved as a *.edid data file. The EDID Viewer is available from the Output Configuration panel when Digital HDMI is selected and a downstream device that transmits EDID is connected. Click **EDID**.

The EDID Editor

The EDID Editor is a property grid that displays the EDID settings to be sent to the upstream device connected to the instrument HDMI SINK connector. These properties can be viewed, edited and saved as a data file. The EDID Editor is available from the Input Settings (Digital HDMI) panel. Click **Edit EDID**.

Using the EDID Editor

Navigate through the grid to the setting of interest, and enter the value or select the option required. Click **OK** to accept the changes you have made.

EDID Audio settings are stored in **Block1 > CEA Data Blocks > CEA Audio Data Block > Short Audio Descriptor n**. By default, APx500 has 2 audio settings, one for 8 channel PCM and one for 2 channel PCM, listed in **Short Audio Descriptor 1** and **Short Audio Descriptor 2**. These settings inform the upstream device that the APx500 SYNC will only accept 8 or 2 channel PCM, at the sample rates and bit depths indicated in the descriptors.

Changing an audio setting

To change an audio setting, you can either add a new audio descriptor or edit one of the existing descriptors.

To add a descriptor, right-click on the **CEA Audio Data Block field. Choose Add CEA Extension Data > Short Audio Descriptor**. Edit the new descriptor to define the new format.

> *Note: the standard limits the amount of data that can be carried in the EDID. You can add new descriptors, but you will not be*

allowed to add many without deleting other information.

To edit a descriptor, in the left column select the field to be changed. Select or enter a new value in the right column for that field.

Note: When a defining field such as Audio Format is changed, other fields within the descriptor may be affected. Note that when selecting AC-3 (see below), the Bit Depth fields are replaced with a Maximum Bit Rate field.

Example: changing PCM to Dolby Digital 5.1

Under **Short Audio Descriptor 1**, select the **Audio Format** field in the left column. Click the arrow in the right column, and choose **AC-3** from the drop-down menu. Now change **Number of Channels** to **6**, change all the sample rates except 48kHz to **False**, and set **Maximum Bit Rate** to **2040**.

Hex

The Hex button enables a display of the EDID data in hexadecimal notation. You may have to enlarge the window to view all the data.

Saving an EDID file

Using the File menu commands or the Toolbar buttons, you can save your changes to a proprietary *.edid file, or to a Microsoft Excel file (*.xls) or a text file (*.txt).

Loading EDID

When you first open the EDID Editor in a New Project, the EDID data are set to the APx default.

Using the File menu commands or the Toolbar buttons, you can load EDID settings from an existing *.edid or *.xls file, or from the device at the HDMI **SOURCE** or **AUX OUT** connectors.

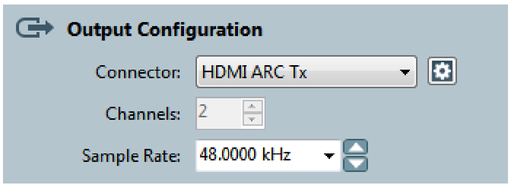
EDID Warnings

Some EDID settings will allow undefined values to be entered and passed with the EDID data. When an undefined value is entered, a warning will appear in the EDID Warning list.

ARC Output Configuration

See page 120 for more about ARC.

Output: HDMI ARC Tx



The ARC Tx selection is only available for an APx analyzer fitted with the HDMI+ARC (model 114) option.

Introduction

HDMI ARC Tx sends digital audio in the IEC 60958 / IEC 61937 (S/PDIF) interface format via the HDMI ARC Tx / AUX IN connector, in accordance with the HEAC ARC (audio return channel) definition in the HDMI 1.4a specification. In this configuration the APx represents a TV, returning audio down an HDMI cable to a home theater receiver Source.

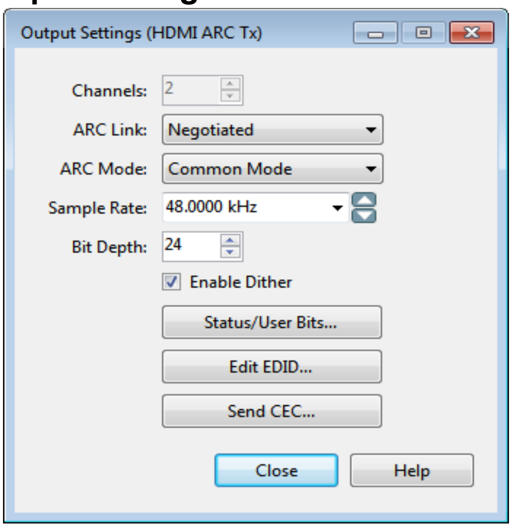
Sample Rate (duplicated on the Settings dialog)

To set the output sample rate (Fs), click the up/down arrows to select a standard rate, or enter an arbitrary value in the **Sample Rate** field. Minimum setting is 8 kHz; maximum is 216 kHz. Default is 48 kHz. **Sample Rate** is also available in the Settings dialog.

Settings

Click the **Settings** button to open the **Output Settings for ARC Tx** dialog.

Output Settings for ARC Tx



ARC Link

- Negotiated
  Typically, ARC is negotiated between two devices, using EDID identification and exchange of CEC messages. If this negotiation fails, an ARC link cannot be established. **Negotiated** is the APx default.
- Audio Forced On
  This mode ignores all negotiation, and simply streams digital audio data on the ARC connectors, regardless of EDID status or exchange of CEC messages.

ARC Mode

- Common Mode
  In Common Mode, the ARC signal is sent commonly on the HEAC+ and HEAC- lines, relative to ground.
- Single Mode
  In Single Mode, the ARC signal is sent on the HEAC+ line, relative to ground.

Sample Rate

To set the output sample rate (Fs), click the up/down arrows to select a standard rate, or enter an arbitrary value in the **Sample Rate** field. Minimum setting is 8 kHz; maximum is 216 kHz. Default is 48 kHz. **Sample Rate** is also available in the Settings dialog.

Bit Depth

The **Bit Depth** control allows you to set the output bit depth (also called digital word length). The default setting is 24 bits, which is also the maximum setting. Minimum setting is 8 bits.

Enable Dither

When the **Enable Dither** checkbox is checked, dither is **ON** (the default). When **ON**, dither is set according to the bit depth, using TPDF (triangular probability density function) dither at ±1 LSB.

Uncheck the checkbox to turn dither **OFF**.

Status/User Bits...

The ARC digital audio stream includes IEC 60958 Status Bits and User Bits. This button opens the Set Status Bits / User Bits panel for the ARC audio stream. See page 539.

Edit EDID...

EDID is Extended Display Identification Data, one of several metadata protocols in the HDMI transport stream.

Click the **EDID...** button to open the EDID Editor. See page 116.

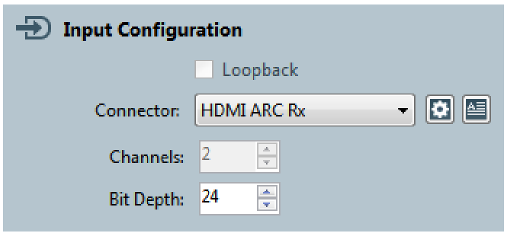
Send CEC...

Establishing, confirming or terminating an ARC link requires exchange of CEC messages. Click **Send CEC...** to open a dialog box. See page 120.

ARC Input Configuration

See page 120 for ore about ARC.

Input: HDMI ARC Rx



The ARC Rx selection is only available for an APx analyzer fitted with the HDMI+ARC (model 114) option.

Introduction

HDMI ARC Rx receives digital audio in the IEC 60958 / IEC 61937 (S/PDIF) interface format via the ARC Rx / HDMI AUX OUT connector, in accordance with the HEAC ARC (audio return channel) definition in the HDMI 1.4a specification. In this configuration the APx represents a home theater receiver, accepting audio sent down an HDMI cable from a TV Sink.

Labels/Colors

Click the **Labels/Colors...** button to open the Labels/Colors dialog, where you can name the input channels with names you choose and change the channel color assignments used in the display and reports. Names and color assignments are maintained independently for the each input configuration selection.

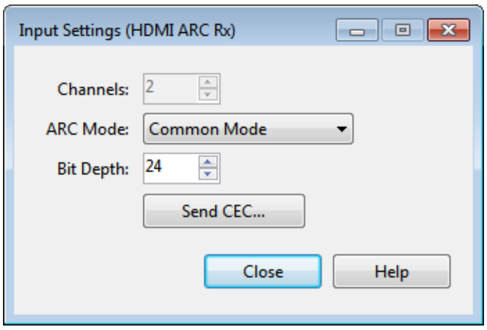
Operation with absent or corrupt interface signal

When Input Configuration is set to a digital input and no signal is present, or if the signal is corrupt or out of range, the input receiver cannot lock (synchronize) to the interface signal, and no valid audio can be recovered. The sample rate indicator in the Status Bar will display an “unlocked” warning, and any measurement result referencing audio from the unlocked input will display an “----” (invalid) result. See page 560 for more about invalid results.

Settings

Click the **Settings** button to open the **Input Settings for ARC Rx** dialog.

Input Settings for ARC Rx



ARC Mode

- Common Mode
  In Common Mode, the ARC signal is sent commonly on the HEAC+ and HEAC- lines, relative to ground. This setting configures the receiver for Common Mode.
- Single Mode
  In Single Mode, the ARC signal is sent on the HEAC+ line, relative to ground. This setting configures the receiver for Single Mode.

See page 120 for more about the ARC physical layer.

Bit Depth

The Bit Depth control allows you to set the input bit depth (digital word length). The default setting is 24 bits, which is also the maximum setting. Minimum setting is 8 bits. The input signal is truncated (not dithered down) if Bit Depth is set to a lower value than Bit Depth of the signal.

We recommend that you use the default setting of 24 bits, unless you have a special application that requires that you mask out low-order bits.

Coupling

Input signals can be analyzed and displayed in one of two coupling modes:

- AC
  AC coupling inserts a digital filter to block any dc signal component from analysis, displaying results for only the ac components. As a consequence, AC coupling introduces a rolloff below 10 Hz. AC coupling is the default setting.
- DC
  DC coupling does not insert a blocking filter, and includes the dc signal component in analysis and display.

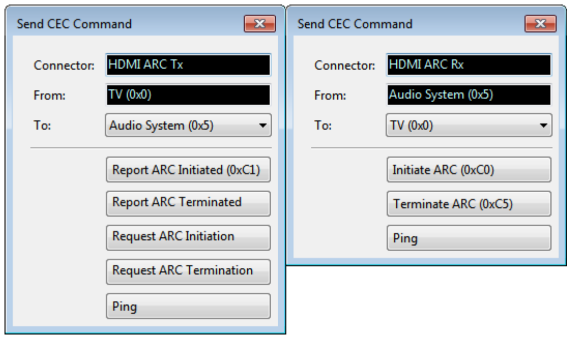
See More about DC in APx analyzers on page 240.

Send CEC...

Establishing, confirming or terminating an ARC link requires exchange of CEC messages.

Click **Send CEC...** to open a dialog box.

Send CEC dialog boxes



When in an HDMI ARC configuration, the Send CEC dialog allows you to send ARC-related CEC commands to the source or sink device connected.

In normal operation, EDID settings and CEC messages are exchanged in a link negotiation process, and use of the commands in this dialog is unnecessary. These are provided for diagnostic purposes.

Tx and Rx dialogs

If the **Send CEC Message** dialog has been opened from Output Settings (HDMI ARC Tx), the role of the APx is “TV,” from which CEC commands can be sent to the connected device, defined as “Audio System.”

Available CEC messages (shown with their hex equivalents) are:

- Report ARC Initiated (0xC1)
- Report ARC Terminated (0xC2)
- Request ARC Initiation (0xC3)
- Request ARC Termination (0xC4)

If the **Send CEC Message** dialog has been opened from Input Settings (HDMI ARC Rx), the role of the APx is “Audio System,” from which CEC commands can be sent to the connected device, defined as “TV.”

Available CEC messages (shown with their hex equivalents) are:

- Initiate ARC (0xC0)
- Terminate ARC (0xC5)

More About ARC

APx support of HDMI versions and features

APx HDMI modules fully support HDMI version 1.3a. The APx HDMI+ARC module adds ARC (Audio Return Channel) support, which is a subset of the HDMI 1.4 specification. Other HDMI features added in 1.4a are not supported. In the APx implementation, ARC Tx (transmit) and ARC Rx (receive) input and output selections are entirely independent of the standard

HDMI audio configuration and features. The ARC Tx signal appears on the connector also used for HDMI AUX IN, and the ARC Rx uses the connector also used for HDMI AUX OUT.

About ARC

ARC (Audio Return Channel) is a feature of HDMI 1.4a that enables digital audio (IEC 60958 / SPDIF) transmission and reception on existing but previously unused conductors in standard HDMI connectors and cables.

ARC was added to eliminate the need for a coaxial or optical digital cable to return audio from a device designated as an HDMI sink. The classic case is a television (typically connected as a sink to a home theater receiver acting as a video source) that has been configured to receive video directly from an RF antenna, rather than from the receiver. The audio from the on-air signal must be routed to the receiver; in the past this required an additional cable. The HDMI ARC function designates existing conductors in the HDMI cable to act as a digital audio return channel, presenting the audio to the source connector on the receiver.

The ARC carrier and embedded audio formats conform to IEC60958 (AES3-S/PDIF).

Physical Layer

The ARC digital audio signal is carried on a twisted pair of wires in an HDMI cable. These wires are designated HEAC+ and HEAC–. The ARC signal can be carried in two modes. In Single Mode, the digital audio signal is carried on the HEAC+ conductor, relative to the HDMI ground. In Common Mode, the ARC signal is carried commonly on the HEAC+ and HEAC– conductors, relative to ground. Common mode facilitates an HDMI 1.4a feature not supported by APx, carrying an ethernet signal simultaneously in differential mode on the HEAC– and HEAC+ conductors.

CEC

In APx, CEC commands are sent using the HD Monitor panel, CEC tab. *See page 33.*

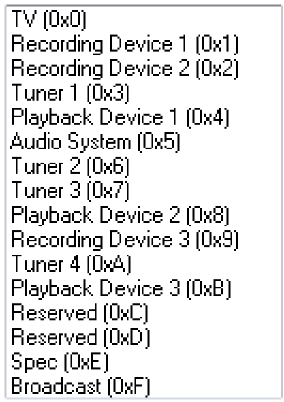
CEC is Consumer Electronics Control, a bus in the HDMI interface. CEC enables HDMI interconnected devices such as a television, a cable set-top box, an audio-video receiver, a disc player, etc. to identify each other and exchange control commands.

These commands are used to simplify operating an audio-video system for the consumer. Selecting the DVD input on a receiver, for example, may also turn the DVD player ON. One IR remote control may be configured to control several different devices.

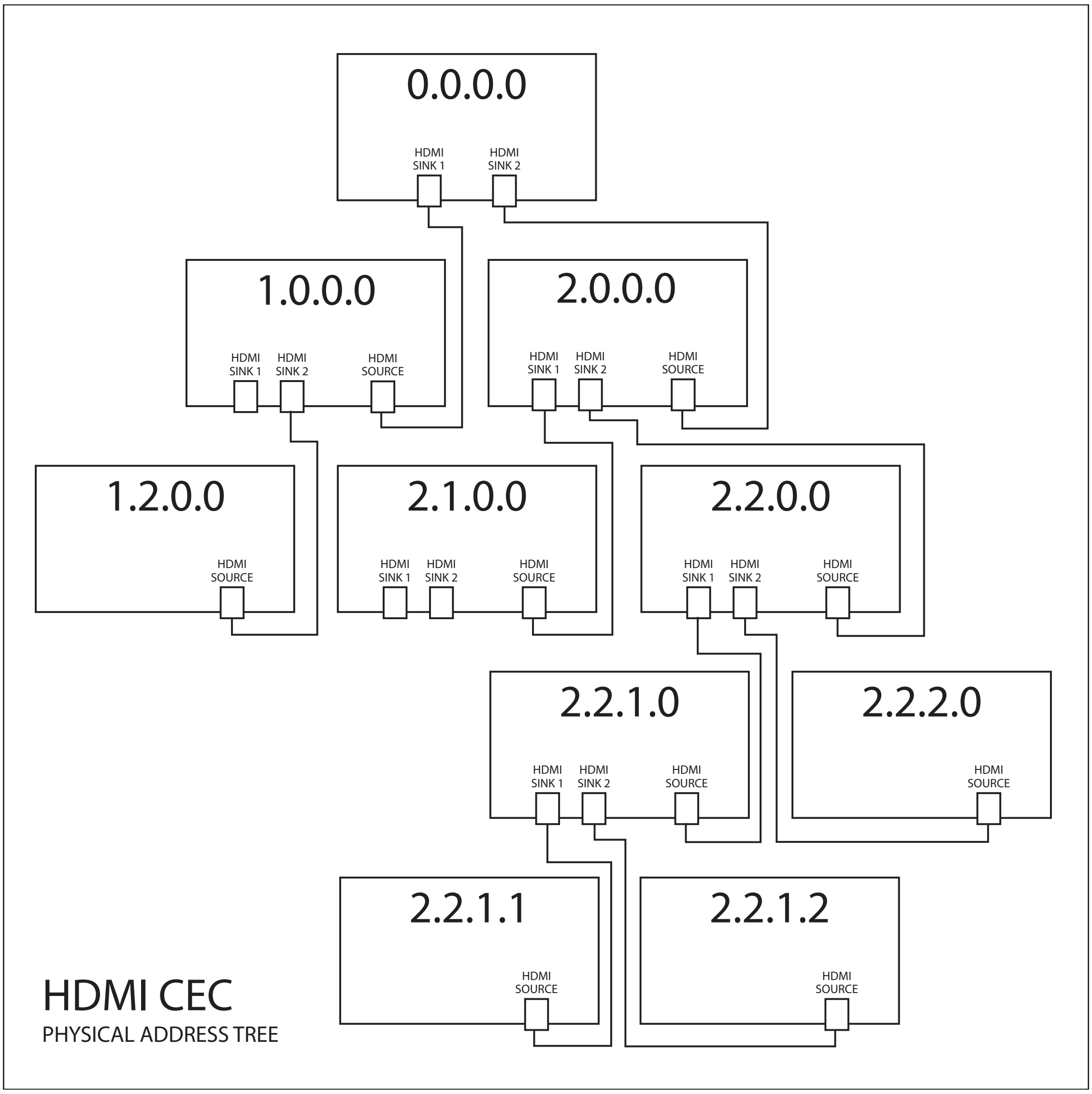
The CEC system assumes the video display or television is the "root" device, and the CEC network expands from this root in a tree pattern, with switchers (e.g. a receiver) as "branches" and source devices as "leaves." The tree can only be 5 layers deep. Up to 10 devices can be connected to a CEC bus.

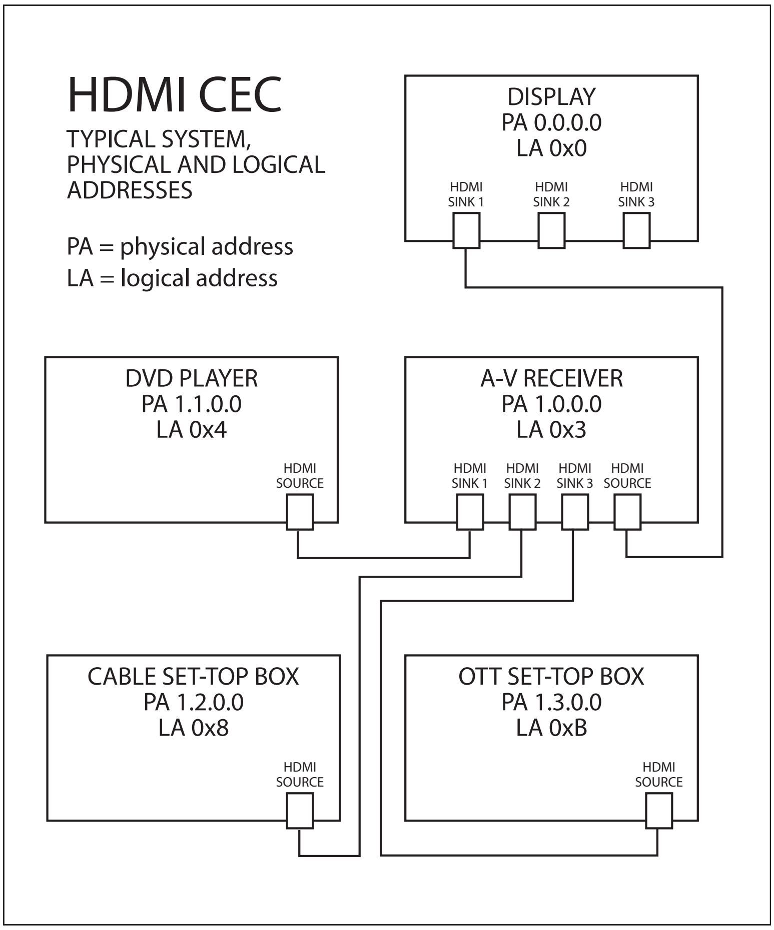
Each device on the tree has a physical address and a logical address. The physical address for the root device is 0.0.0.0. Other devices inherit physical addresses according to the connector and the device each is connected to.

Devices also have logical addresses, corresponding with their roles in the system. There are 16 CEC logical addresses defined. Each device negotiates for a logical address when it is connected to the bus.



CEC Device menu, showing logical addresses.


0.0.0.0
HDMI SINK 1
HDMI SINK 2
1.0.0.0
HDMI SINK 1
HDMI SINK 2
HDMI SOURCE
2.0.0.0
HDMI SINK 1
HDMI SINK 2
HDMI SOURCE
1.2.0.0
HDMI SOURCE
2.1.0.0
HDMI SINK 1
HDMI SINK 2
HDMI SOURCE
2.2.0.0
HDMI SINK 1
HDMI SINK 2
HDMI SOURCE
2.2.1.0
HDMI SINK 1
HDMI SINK 2
HDMI SOURCE
2.2.2.0
HDMI SOURCE
2.2.1.1
HDMI SOURCE
2.2.1.2
HDMI SOURCE
HDMI CEC
PHYSICAL ADDRESS TREE



Detailed information about CEC and HDMI is available at www.hdmi.org.

16

Bluetooth I/O

Introduction

The APx Bluetooth Option is a hardware module available for the APx525 and 585 families of analyzers, including the APx582. It provides a Bluetooth wireless technology source or sink interface.

Bluetooth profiles support audio in a lower-quality (voice) mode (HFP and HSP profiles, now with mSBC wideband speech) and in a higher-quality (music) mode (A2DP profile).

Bluetooth Discovery, Pairing and Connection

Bluetooth connections require successful radio frequency (RF) communication between the APx analyzer and the DUT, and successful device handshaking through device discoverability, pairing and connection protocols. Data transfer is encrypted and must be authorized by the exchange of link keys, and personal identification number (PIN) codes may be required.

Source / Sink exclusivity with APx Bluetooth

The Bluetooth A2DP profile is a simplex (unidirectional) audio channel. Since the APx Bluetooth option provides only one Bluetooth RF link, this means that for the A2DP profile, **Output Configuration: Bluetooth** and **Input Configuration: Bluetooth** are mutually exclusive. Only one of these configurations can be selected simultaneously in a project.

The Bluetooth HSP and HFP profiles are duplex (bidirectional) audio channels. This means that for these profiles, both **Output Configuration: Bluetooth** and **Input Configuration: Bluetooth** can be selected simultaneously in a project.

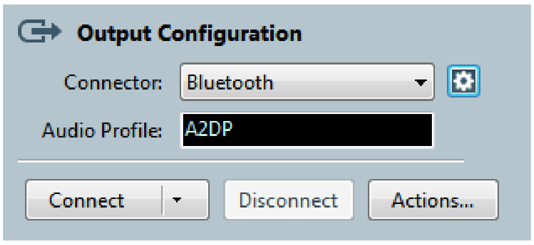
See APx Bluetooth Option on page 9, More About Bluetooth on page 131 and Using Bluetooth in a Sequence on page 129. See More About Supported Bluetooth Profiles on page 132.

Bluetooth Settings are Global

Bluetooth settings are unique in APx500, in that they are *global*. This means the settings affect the entire project: all measurements, all signal paths.

Output Configuration: Bluetooth

This selection enables output of audio from the APx generator to a Bluetooth device under test, using Bluetooth radio frequency (RF) transmission.



Note: The Bluetooth choice is only available for analyzers fitted with the APx Bluetooth Option module.

Profile

If the APx is already connected to a Bluetooth device, the audio profile is shown here.

Settings

Click the **Settings...** button to discover, select, pair, connect and configure a Bluetooth link. See page 125.

Connect

If the APx is already paired with one or more Bluetooth devices, you can open the Connect menu to select a device and one or more profiles for connection.

Disconnect

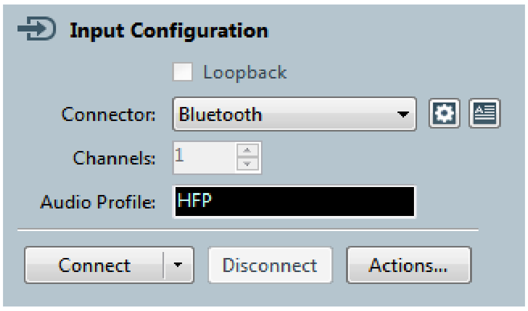
If the APx is already connected to a Bluetooth device, you can click the **Disconnect** button to terminate the connection.

Actions

Click **Actions** to open the **Bluetooth Actions** dialog. If the APx is already connected to a Bluetooth device, you can initiate Bluetooth Actions from this dialog. Only the actions appropriate to the current connection and profile are shown. See Actions on page 128.

Input Configuration: Bluetooth

This selection enables input of audio from a Bluetooth device under test, using Bluetooth radio frequency (RF) transmission.



Note: The Bluetooth choice is only available for analyzers fitted with the APx Bluetooth Option module.

Profile

If the APx is already connected to a Bluetooth device, the audio profile is shown here.

Settings

Click the **Settings...** button to discover, select, pair, connect and configure a Bluetooth link. See page 125.

Connect

If the APx is already paired with one or more Bluetooth devices, you can open the **Connect** menu to select a device and one or more profiles for connection.

Disconnect

If the APx is already connected to a Bluetooth device, you can click the **Disconnect** button to terminate the connection.

Actions

Click **Actions** to open the **Bluetooth Actions** dialog. If the APx is already connected to a Bluetooth device, you can initiate Bluetooth Actions from this dialog. Only the actions appropriate to the current connection and profile are shown. See Actions on page 128.

Loopback

Loopback is not available when either the Output or Input Configuration is set to Bluetooth.

Operation with absent or corrupt interface signal

When Input Configuration is set to a digital input and no signal is present, or if the signal is corrupt or out of range, the input receiver cannot lock (synchronize) to the interface signal, and no valid audio can be recovered. The sample rate indicator in the Status Bar will display an “unlocked” warning, and any measurement result referencing audio from the unlocked input will display an “----” (invalid) result. See page 560 for more about invalid results.

A2DP phase error with APX-BT-WB hardware

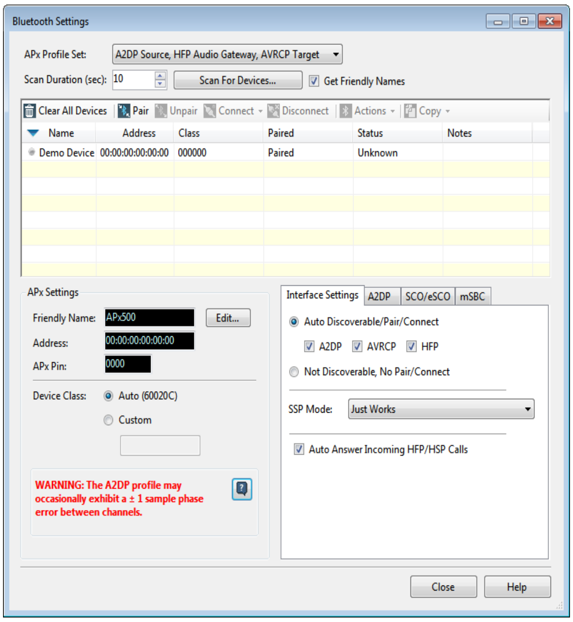
A flaw in the firmware supplied by our Bluetooth chipset vendor causes the APx Wideband Speech Bluetooth module (APX-BT-WB) to exhibit a +/-1 sample phase error between channels when using the A2DP profile (source and sink), in about 15% of trials. The error, when present, is introduced at the time the Bluetooth settings are committed to the module chipset, and is constant during the operation of the Bluetooth channel. When present, the error affects results sensitive to phase, such as the time domain signal monitor, the group delay result, and the interchannel phase result .

The error only occurs only when using the APX-BT-WB hardware, only in the A2DP profile. HSP and HFP profiles are unaffected and operate correctly.

A warning appears on the Bluetooth Settings panel when APX-BT-WB hardware is in use.

Although we are actively working to resolve this issue, we cannot guarantee if and when it will be corrected. The APX-BT module, which does not support wideband speech, does not exhibit this problem.

Bluetooth Settings Dialog



This dialog provides settings for Bluetooth discovery, pairing, connection and other configuration. Unlike most APx settings, Bluetooth profile selection, pairing and connection settings are global to the APx project. Settings made here affect both Bluetooth source and sink configurations across all signal paths in the project.

A flow chart depicting discovery, pairing and connection paths is shown on page 128.

Settings

APx Profile Set

APx has four sets of profiles that the Bluetooth Option module can assume. Each set lists the profiles available for connection and/or the profiles available for access by a remote device. Choose a Bluetooth Profile Set here.

- A2DP Source, HFP Audio Gateway, AVRCP Target
- A2DP Source, HSP Audio Gateway, AVRCP Target
- A2DP Sink, HFP Hands Free, AVRCP Controller
- A2DP Sink, HSP Headset, AVRCP Controller

In A2DP Source, the APx Bluetooth transmitter mutes the audio when the average signal level falls below –54 dBFS for more than 1 second. We recommend maintaining test levels above –54 dBFS for meaningful results.

Scan

Scan for Devices

This button initiates a scan to identify all the discoverable Bluetooth devices within range. Discovered devices are listed in the device grid.

Scan Duration

This setting limits the scan time to the value set. The maximum time is 48 seconds. The default is 10 seconds.

Get Friendly Names

By default, APx asks for a device's friendly name. In automated processes, this process adds extra time. Uncheck this checkbox to disable this feature.

Device grid

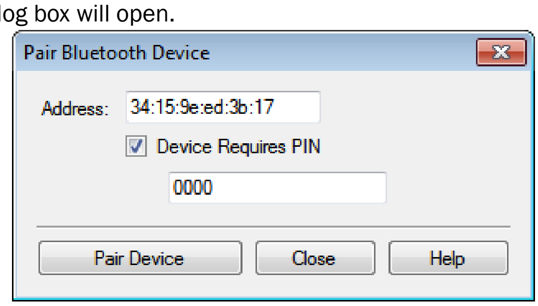
This grid shows the devices discovered in the scan, along with information fields. There are controls to pair, unpair, connect, disconnect and to copy the device address and link key.

Clear All Devices

Removes all devices from the list, including any pairing information for devices. Click **Scan** to re-populate the list.

Pair

Pairing provides a way for devices to exchange link keys. To enable pairing, you will typically be required to execute a command or series of commands at the device under test; refer to the documentation received with the device. When pairing is enabled, select the device in the Device Grid, and click **Pair**. A Pairing dialog box will open.



Some devices may require the exchange of a PIN code. The APx Bluetooth PIN code is set to 0000; a field in the pairing dialog allows input of a remote device's PIN code.

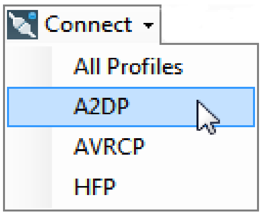
Pairing information is saved in the project, until the Device Grid is cleared, or the until device is Unpaired.

Unpair

Unpairs the selected, previously paired device.

Connect

Connects to the selected, previously paired device. A Connect menu opens. Select the profile you would like to offer to the device.



Disconnect

Disconnects the selected device if connected.

Actions

Opens various Actions dialogs. This duplicates the function of the Actions button shown on Signal Path Setup, without the necessity of leaving the Bluetooth Settings dialog.

Copy

Copies the device address and key link to the Windows Clipboard. This text can then be pasted into another application or tool.

APx Settings

Friendly Name

The Bluetooth Option module is programmed with its Friendly Name, set at the factory to the model and serial number of the host instrument. You can change this to a Friendly Name of your choice by clicking the Edit button.

Address

The APx Bluetooth address is shown here. This address is unique to each Bluetooth Option module, and cannot be changed.

APx PIN

Some devices require the exchange of a PIN (personal identification number) when pairing. The fixed PIN for an APx analyzer is 0000, as shown here. Enter this PIN in the device's pairing dialog if required.

Device Class

A Bluetooth device class is a code that identifies the type of device. Some Bluetooth devices will only recognize devices of a certain class.

Auto

When Device Class is Auto and APx is set to a source profile, the APx device type is set to 60020C, which represents "smart phone". When set to Auto and a sink profile, the APx device type is set to 240408, which represents "hands free device" (a stereo headset).

Custom

You can enter a different Device Class in the Custom field. See Class of Device Generator at http://bluetooth-pentest.narod.ru/software/bluetooth_class_of_device-service_generator.html.

Interface Settings Tab

Auto Discoverable/Pair/Connect

When this is selected, the APx Bluetooth Option is discoverable by remote devices. If the remote device initiates pairing, the APx will pair. If the remote device initiates connection, the APx is available to connect as a sink for the profiles checked here.

Not Discoverable, No Pair/Connect

When this is selected, the APx Bluetooth Option is not discoverable by remote devices, and it will not respond to pairing or connection initiated by remote device.

SSP Mode

APx provides four SSP (Secure Simple Pairing) options. The Just Works choice is used in Bluetooth device relationships that require little security. The Numeric Comparison choices are man-in-the-middle (MITM) interactions for device relationships that require higher security. Choose the SSP Mode that is appropriate for the Bluetooth device you are testing. See "More About Secure Simple Pairing" on page 133 for detailed information.

- Just Works
- Numeric Comparison, Display Only
- Numeric Comparison, Display+Buttons
- Numeric Comparison, Keyboard

Auto Answer Incoming HFP/HSP Calls

Check this checkbox if you would like your APx analyzer to automatically answer phone calls to a connected Bluetooth phone.

Audio Coupling

The digital audio embedded in the Bluetooth signal can be allowed to have a DC component (an offset), if present, by choosing DC Coupling; alternatively, the DC can be blocked using a digital filter by choosing AC Coupling (the default).

A2DP tab

Not all devices support all options within a given profile. The Profile settings here show the APx Bluetooth preferred settings for several options. If the preferred setting is available in the remote device, it will be used. If not, the default setting supported by the remote device will be used.

A2DP Codec

The SBC codec is mandatory for all A2DP devices. Other codecs can be used. If an alternative codec is selected in APx, and if it is supported in the remote device, it will be used instead of the SBC codec. The coded selections are

- SBC
- apt-X

A2DP Sample Rate

Choose the audio sample rate for the A2DP profile here. If the sample rate is supported in the remote device, it will be used; otherwise, the default sample rate supported by the remote device will be used. The selections are

- 16 kHz
- 32 kHz
- 44.1 kHz
- 48 kHz (This selection is only available if the mSBC codec is supported by your Bluetooth module hardware. See Bluetooth Option on page 9 for more information about Bluetooth codec support.)

SBC Channel Mode

Choose the channel mode for the SBC codec here. Selections are

- Joint Stereo
- Stereo
  Stereo sends two discrete channels of audio data, phase locked.
- Dual Channel
  Dual Channel sends two discrete channels of audio data.
- Mono
  Mono sends only one channel of audio data.

SCO/eSCO tab

This tab is only available if the mSBC codec is supported by your Bluetooth module hardware. See Bluetooth Option on page 9 for more information about Bluetooth codec support.

This tab provides management of SCO and eSCO channels and packet types. A SCO is a synchronous connection-oriented link, a radio link used for voice data. eSCO links are enhanced SCO, providing greater flexibility. For additional reading, see the Bluetooth protocols article in Wikipedia. For a SCO connection, you can specify any combination of supported packet types:

- HV1
- HV2
- HV3

For an eSCO connection, you can specify

- eSCO channel latency in ms
- eSCO Retransmission preferences:
  - No Retransmission
  - Power Saving Optimized
  - Link Quality Optimized, or
  - No Preference

You can select Wideband Speech mode, which uses a sample rate of 16 kHz and the mSBC codec for HFP/HSP connection.

You can choose any combination of supported packet types:

- EV1
- EV2
- EV3
- 2-EV3
- 3-EV3
- 2-EV5
- 3-EV5

mSBC tab

This tab is only available if the mSBC codec is supported by your Bluetooth module hardware. See Bluetooth Option on page 9 for more information about Bluetooth codec support.

This tab enables management of mSBC parameters. mSBC requires an eSCO link. mSBC is the modified SBC codec, a component of HFP 1.6.

For an mSBC connection, you can specify

- mSBC channel latency in ms
- mSBC Retransmission preferences:
  - No Retransmission
  - Power Saving Optimized
  - Link Quality Optimized, or
  - No Preference

and any combination of supported packet types:

- EV1
- EV2
- EV3
- 2-EV3

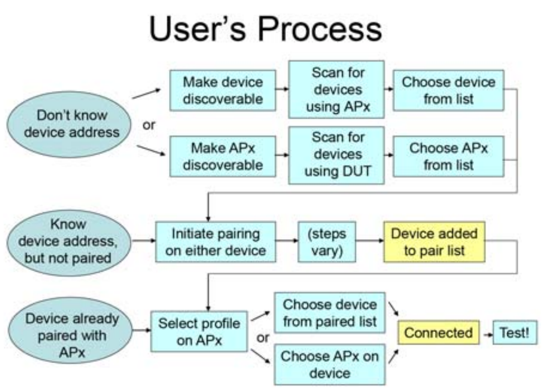
- 3-EV3
- 2-EV5
- 3-EV5

Steps in connecting APx to a Bluetooth device

There are 3 steps involved in setting up a connection between any Bluetooth devices. APx500 software provides more visibility and control of options for these communication steps than most DUTs.

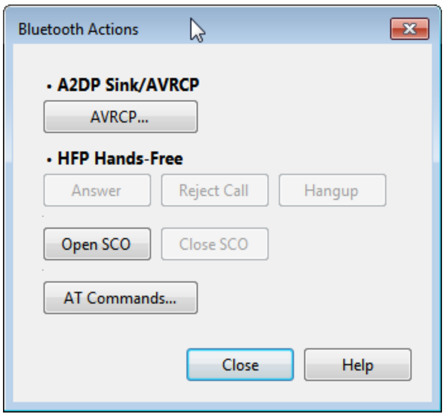
- Discovery
  Discovery scans the area and lists any Bluetooth devices that are not "undiscoverable" (hidden). Many devices are undiscoverable by default, and must be set to a discoverable mode through a switching sequence.
- Pairing
  Pairing establishes a mutual, secure relationship between devices that have at least one compatible profile. Pairing is stored in non-volatile memory in each device. Devices can pair with more than one other device.
- Connection
  Paired devices can connect using compatible profiles and roles. Connection enables exchange of audio or control data. Each device can connect with only one device at a time.

Process for discovery, pairing and connection



Actions Dialogs

Bluetooth Actions are commands that can be sent from the APx analyzer to the Bluetooth device. The current Bluetooth profile(s) and connection determine which commands are available in the Bluetooth Actions dialog box.



Typical Actions Dialog

Bluetooth Actions for A2DP Source, HFP AG, AVRCP Target

For this profile set, these actions are available:

A2DP Source

For A2DP Source, the actions are

- Start Streaming / Stop Streaming

HFP Audio Gateway

For HFP Audio Gateway, the actions are

- Open SCO
- Dial
- Ring
- Close SCO
- Hangup

A "SCO" is a "Synchronous connection-oriented" link, which is the type of radio link Bluetooth uses for voice data. The SCO link is used in the APx supported protocols HFP and HSP.

Bluetooth Actions for A2DP Source, HSP AG, AVRCP Target

For this profile set, these actions are available:

A2DP Source

For A2DP Source, the actions are

- Start Streaming / Stop Streaming

HSP Audio Gateway

For HSP Audio Gateway, the actions are

- Open SCO

- Close SCO
- Ring

A “SCO” is a “Synchronous connection-oriented” link, which is the type of radio link Bluetooth uses for voice data. The SCO link is used in the APx supported protocols HFP and HSP.

Bluetooth Actions for A2DP Sink, HFP Hands Free, AVRCP Controller

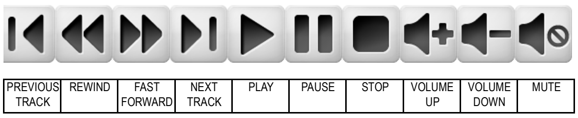
For this profile set, these actions are available:

A2DP Sink/AVRCP

For A2DP Sink/AVRCP, click the

- AVRCP...

button to open the Send AVRCP Command transport controls panel:



HFP Hands-Free

For HFP Hands Free, the actions are

- Answer
- Reject Call
- Hangup
- Open SCO
- Close SCO

AT Commands

For AT Commands, click the

- AT Commands...

button to open the Send AT Commands panel. See page 129.

Bluetooth Actions for A2DP Sink, HSP Headset, AVRCP Controller

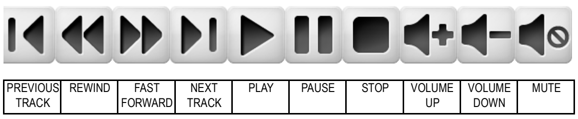
For this profile set, these actions are available:

A2DP Sink

For A2DP Sink/AVRCP, click the

- AVRCP...

button to open the Send AVRCP Command transport controls panel:

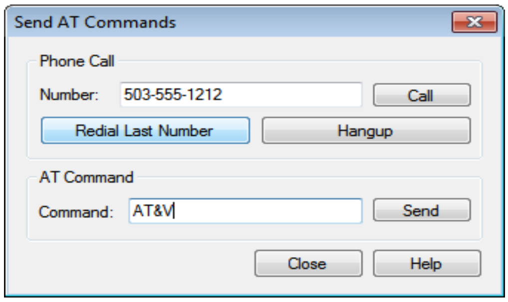


HSP Headset

For HSP Headset, the actions are

- Button
- Open SCO
- Close SCO

Bluetooth: Send AT Commands Dialog



You can open the Send AT Commands dialog by clicking Actions > AT Commands from Signal Path Setup, or from the context menu opened by right-clicking in the Bluetooth Monitor.

You must have an APx input or output configured for Bluetooth, and be paired and connected to an appropriate Bluetooth HFP profile device.

Phone Call

To make a phone call, enter a new phone number in the Number field and click **Call**, or click **Redial Last Number**. Disconnect a phone call by clicking **Hangup**.

AT Command

Enter any valid AT command as a text string, and click **Send** to transmit the command to the connected device.

The Bluetooth Monitor

The status of a number of current Bluetooth settings is shown in the Bluetooth Monitor. Additionally, Bluetooth actions, settings and utility functions are available through a context menu (right-click in the Bluetooth Monitor display.) See Bluetooth Monitor on page 34.

Running a Sequence with Bluetooth set as an interface

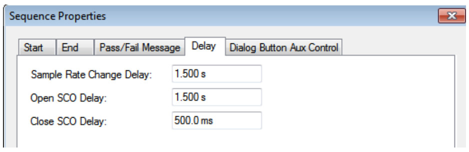
Unlike other APx Output and Input Configurations, which are simply established by a software selection and a cable connection, Bluetooth configurations require discovery, pairing, profile selection, etc. in an extended negotiation between the APx analyzer and the Bluetooth Device Under Test.

When automating Bluetooth testing, settings and steps that provide a means of specifying and negotiating the connection, connecting and disconnecting, and initiating actions must be included.

The Bluetooth Global settings must be set in the project before the sequence is run.

Sequence Delays for Bluetooth

Sufficient delays may be required to allow the DUT time to execute SCO commands and sample rate changes.



Such delays can be specified on the Delay tab of the Project Sequence Properties dialog. See page 480.

Connecting a device in a Sequence

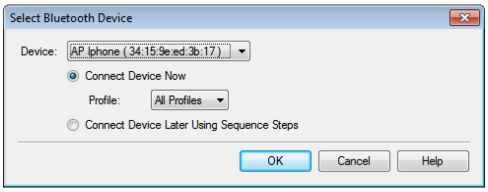
A Bluetooth device used in a sequence must be connected for testing. If there are multiple signal paths in the sequence, each signal path can be connected to a different device. However, global settings such as sink/source profile sets are in force across the entire project.

Signal Path Setup > Edit Prompts and Properties > Selected Bluetooth Device

The Signal Path sequence properties offer an opportunity to use the currently connected device (if present), or to prompt the user at run time to select a device.

- Use Connected Device if Present
  This uses the connected device. If not present, the Select Bluetooth Device dialog opens at sequence run time.
- Always Choose New Device
  The Select Bluetooth Device dialog opens at sequence run time.

Sequence run time: Select a Bluetooth Device



This is a sequence run time prompt. If, when a sequence is started, no Bluetooth device is selected, or the Signal Path sequence properties asserts “Always Choose New Device,” the Select Bluetooth Device dialog is opened.

Device

Use the menu to choose from

- None
- ** a list of discovered devices ** (this may be empty)
- Select New Device (which opens the Pair Bluetooth Device dialog)

Connect Device Now

Select this choice to connect to the selected device immediately, when OK is clicked.

Profile

Choose the profile to be used for the immediate connection.

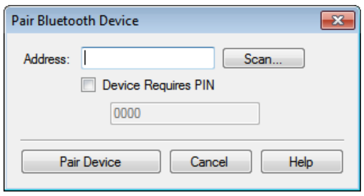
Connect Device Later Using Sequence Steps

Select this choice to connect later in the sequence. Be sure to set a Connect step (and specify profile) in a subsequent Signal Path, using

- Signal Path Setup > Edit Prompts and Properties > Add Step > Bluetooth > Connect Bluetooth Device Profile(s)...

See page 131.

Sequence run time: Pair Bluetooth Devices



This is a sequence run time prompt, called by the Select a Bluetooth Device prompt. Pair with a device, if the device address is known, shown, or Scan for a device in the area.

Address

If you know the device address, enter it in the Address field. Otherwise, click Scan...

Scan...

The **Scan...** button is available if this dialog has opened during a sequence. Click this button to open the **Scan for Bluetooth Devices** dialog.

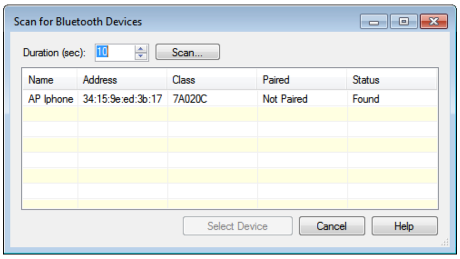
Device Requires PIN

Check **Device Requires PIN** if required. The default PIN is 0000; other PINs (such as the common 1234) can be entered here.

Pair Device

Click to pair.

Sequence run time: Scan for Bluetooth Devices



This is a sequence run time prompt, called by the **Scan** button on the **Pair Bluetooth Devices** prompt.

Scan for Devices

This button initiates a scan to identify all the discoverable Bluetooth devices within range. Discovered devices are listed in the device grid.

Scan Duration

This setting limits the scan time to the value set. The maximum time is 48 seconds. The default is 10 seconds.

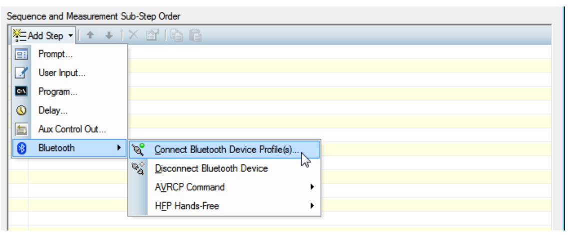
Select

Click Select to choose the highlighted device and return to the Pair Bluetooth Device dialog.

Connecting later in the Sequence

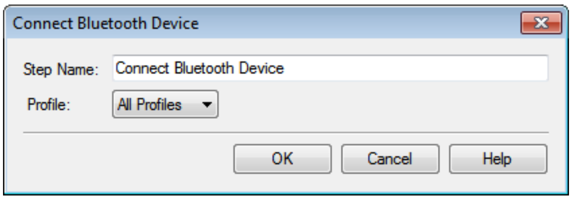
If you have chosen **Connect Device Later Using Sequence Steps**, you must add a Connect Step, as shown below.

Signal Path Setup > Edit Prompts and Properties > Add Step > Bluetooth



- **Connect Bluetooth Device Profile(s)...**
  This step connects a device at the beginning of a signal path, if "Connect Device Later..." is selected in Select Bluetooth Device dialog. See page 130.
- **Disconnect Bluetooth Device**
  This step disconnects the current connected Bluetooth device.

Sequence: Connect to a Bluetooth Device Step



This dialog is called by the Signal Path Setup sequence step "Connect Bluetooth Device Profile(s)."

This step requires that a Bluetooth device has already been discovered, paired and selected.

Step Name

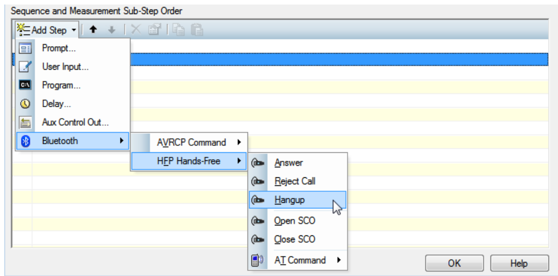
Enter a Step Name here.

Profile

Choose a profile to use when connecting.

Adding Actions to a Sequence

You can initiate Bluetooth Actions at measurement nodes, using **[Navigator Measurement node] > Edit Prompts and Properties > Add Step > Bluetooth**. Here is an example:



More About Bluetooth...

General

Bluetooth is a short-distance (a few meters) control, data, and audio communications wireless technology. Bluetooth replaces annoying wires in the user's "personal area."

Bluetooth uses low power, frequency-hopping radio in the 2.4 GHz band. Communication is two-way (for handshaking, metadata, etc); some profiles (HFP, for example) support duplex audio (both directions simultaneously); some profiles (A2DP) support only simplex audio (one direction per connection).

Typical uses are mouse, keyboard, cell phone, headphones, hands-free talk and listen.

APx Bluetooth Option

An APx Bluetooth Option module must be fitted in the analyzer instrument to enable Bluetooth transmission and reception. See Bluetooth Option on page 9.

Bluetooth Profiles

Bluetooth has about 30 “profiles” that describe the capabilities and/or current operating modes for Bluetooth devices. For devices to communicate, they must support and share a common profile.

For example, a wireless mouse uses the “HID” profile, which has no audio capabilities. Wireless headphones use the “A2DP” profile, which has no cursor control capabilities. The Bluetooth profiles these devices use are not compatible with each other.

APx supported profiles

The APx Bluetooth Option supports four Bluetooth profiles. The supported profiles are

- A2DP (Advanced Audio Distribution Profile)
  This is a one-way (source or sink), relatively high-quality stereo audio profile.
- HFP (Hands Free Profile)
  This is a bi-directional voice communications audio profile that includes AT-type commands for phone use. Legacy Bluetooth DUTs support only the CSVD codec at sample rate of 8 kHz; HFP 1.6 compatible DUTs support the wideband voice mSBC codec at a sample rate of 16 kHz. See Bluetooth Option on page 9 for more information about Audio Precision support of HFP 1.6.
- HSP (Head Set Profile)
  This is a simpler version of HFP, using the CSVD codec at 8 kHz and a reduced set of AT commands.
- AVRCP (Audio Video Remote Control Profile)
  This provides Play-Pause-Forward-Reverse “remote control” transport-type commands to control an audio source. This profile is typically used in conjunction with A2DP for personal audio player applications.

More about Supported Bluetooth Profiles

HSP

HSP is the “Head Set Profile.” This profile supports voice-quality audio, using the CVSD codec at 8 kHz sample rate. This profile is used for phone-to-headset communication.

Roles

Supported roles are AG “Audio Gateway” (the phone) and HS “Headset”.

Audio

Audio flows in a duplex (bi-directional) mode, connecting the Audio Gateway device to the Headset device.

Actions

A subset of AT commands are also supported for phone operations.

HFP

HFP is the “Hands Free Profile.” This profile supports voice-quality audio, using the CVSD codec at 8 kHz sample rate or the HFP 1.6 “wideband speech” mSBC codec at 16 kHz*. This profile is used for phone-to-headset communication and for phone-to-car kit hands free communication.

**Note: HFP 1.6 wideband speech is only available if the mSBC codec is supported by your Bluetooth module hardware. See Bluetooth Option on page 9 for more information.*

Roles

Supported roles are AG “Audio Gateway” (the phone) and HF “Hands Free” (the headset or car kit mic/speaker).

Audio

Audio flows in a duplex (bi-directional) mode, connecting the Audio Gateway device to the Hands Free device.

Actions

A subset of AT commands are also supported for phone operations. Compared to HSP, HFP provides a few more AT commands for hands-free convenience, such as last number redial.

A2DP

A2DP is the “Advanced Audio Distribution Profile.” This profile supports higher bit rate, higher performance stereo audio. The SBC codec is mandatory; codecs such as mp3, AAC, apt-X and others are optionally supported.

Roles

APx supported profile roles are “source” (transmitting audio) and “sink” (receiving audio). Audio is distributed in one direction only, from the source device to the sink device.

Audio

APx supports the mandatory SBC and the optional apt-X codecs. In A2DP Source, the APx Bluetooth transmitter mutes the audio when the average signal level falls below –54 dBFS for more than 1 second. We recommend maintaining test levels above –54 dBFS for meaningful results.

AVRCP

AVRCP is the “Audio/Video Remote Control Profile.” This profile is used in conjunction with A2DP, and provides “transport” controls such as Play, Pause, Reverse, Forward, etc.

Roles

Supported profile roles are “controller” and “target.”

Note: when APx is the target, AVRCP commands are ignored.

More about Bluetooth SSP (Secure Simple Pairing) Mode

SSP is required for devices using Bluetooth v 2.1 and later. Bluetooth v 2.0 and earlier devices use Legacy Pairing.

APx provides four SSP (Secure Simple Pairing) options. You must choose the option that is appropriate for the Bluetooth device you are testing.

Man-in-the-middle (MITM)

Man-in-the-middle or MITM is the term used in Bluetooth technology to refer to Secure Simple Pairing mechanisms that require human interaction. One device may display a PIN, for example, and the user may be required to enter the same PIN using a keyboard on a second device. MITM methods add security to the pairing process.

APx SSP Modes

Just Works

This mode is used in Bluetooth device relationships that require little security, such as headsets. User interaction is not required.

Numeric Comparison, Display Only

Some Bluetooth devices have only a display screen to interact with a user; an example is a car kit (car stereo head unit). Use this mode when APx is playing the role of such a device.

A use case would be testing a smart phone. APx would be an A2DP Sink, HFP with a display only (the car kit), and the DUT would be the smart phone. The phone would initiate pairing, and APx would display the PIN transmitted from the phone. The user would confirm (using a button on the phone) that the PIN displayed in APx matched the PIN on the phone.

Numeric Comparison, Display+Buttons

Some Bluetooth devices have a display screen and one or more buttons to interact with a user; the button may be used for a binary (Yes/No) response to a query. An example is a smart phone. Use this mode when APx is playing the role of such a device.

A use case would be testing a car kit. APx would be an A2DP Source, HFP Hands-Free with a display and buttons (the smart phone), and the DUT would be the car kit. APx would initiate pairing and transmit a PIN to the DUT. The DUT would display the PIN, and the user would confirm (using a button in the APx prompt dialog) that the PIN in the DUT display and the APx PIN matched.

Numeric Comparison, Keyboard

Some Bluetooth devices have a numeric or alphanumeric keyboard to interact with the user. An example is a computer used as an audio source, distributing iTunes audio to a home entertainment system. Use this mode when APx is playing the role of such a device.

There are not obvious use cases in audio test for this mode, which is include in APx for completeness. A non-audio test use case would be pairing a Bluetooth keyboard to a tablet computer. The computer would display a PIN, and the user would enter a PIN (using the keyboard) that matches the PIN on the tablet screen.

Legacy Pairing

Bluetooth v2.1 and later devices (such as APx) are permitted to use Legacy Pairing modes when pairing with a Bluetooth v2.0 or earlier device.

Glossary of Bluetooth terms

These terms are used in the Audio Precision Bluetooth audio testing implementation.

A2DP is the Advanced Audio Distribution Profile, with two roles: source and sink. This profile supports higher bit rate, higher performance stereo audio, with sample rates up to 48 kHz. The SBC codec is mandatory; codecs such as mp3, AAC, apt-X and others are optionally supported.

AG is the duplex Audio Gateway role, used in HFP and HSP. AG is the node (such as a car kit) that mediates between the user and the mobile phone.

apt-X is an optional high-performance codec used in A2DP.

AT command is an audible signal used to control a device. AT commands are a PSTN (Public Switched Telephone Network) legacy.

AVRCP is the “Audio/Video Remote Control Profile.” This profile is used in conjunction with A2DP, and provides “transport” controls such as Play, Pause, Reverse, Forward, etc. Roles are Controller and Target.

Connect
Paired devices can connect using compatible profiles and roles. Connection enables exchange of audio or control data. Each device can connect with only one device at a time.

CVSD is Continuously Variable Slope Delta modulation, the codec used in HSP and legacy HFP profiles. With HFP 1.6, higher data rates enable the use of a higher quality codec called mSBC.

Device address
Every Bluetooth device has a unique 48-bit device address, in APx500 displayed in hex format, such as 00:f4:b9:c3:a0:cc.

Device class
A Bluetooth device class is a code that identifies the type of device. Some Bluetooth devices will only recognize devices of a certain class.

Discovery
Bluetooth devices that are not paired can discover each other when they are in range. Some devices allow users to make the device undiscoverable.

duplex refers to a bi-directional audio channel, across which speaking and listening can occur simultaneously.

eSCO
Extended SCO, available in the Bluetooth 1.2 specification. Adds new packet types (EV1, etc.) and more flexibility in channel parameters, allows retransmission of bad packets. Also see SCO.

EV1
First of a series of packet types available with eSCO, as opposed to the HV1, etc. packet types in SCO.

Friendly name is an optional name for a Bluetooth device, more easily understood and remembered than the device address. An example is “iPhone”.

HF is Hands Free, the duplex “phone” role in the HFP profile.

HFP is the Hands Free profile, intended to allow hands-free device operation in an automobile. It is similar to HSP, with more phone controls. Roles are AG and HF.

HFP 1.6 is a revision of the HFP specification, which allows new packet types and higher data rates compared to legacy HFP. These data rates and newly supported codecs such as mSBC enable WBS (Wideband Speech) performance.

HS is Head Set, the duplex “phone” role in the HSP profile.

HSP is the Head Set profile, intended to enable Bluetooth mobile phone use. Roles are AG and HS.

HV1
First of a series of packet types available with SCO, as opposed to the EV1, etc. packet types in eSCO.

Link key is a shared secret exchanged in pairing.

mSBC is a monaural version of the SBC codec, optimized for use in the HFP 1.6 profile. This codec and the higher available sample rates enable Wideband Speech operation.

Pair
Bluetooth devices that have discovered each other can be paired by exchanging a link key to form a bond. For some devices, pairing is automatic upon discovery; for others, user interaction is required. Pairing establishes a mutual, secure relationship between devices that have at least one compatible profile. Pairing is stored in non-volatile memory in each device. Devices can pair with more than one other device.

PIN code
A Personal Identification Number embedded in a device or provided by a user that is exchanged in a pairing negotiation.

Profile
One of a number of defined Bluetooth relationships covering a range of devices and use cases. A device can support more than one profile, and devices can be paired acknowledging more than one profile.

role
A role is a defined use of a device within a profile. In A2DP, for example, the roles are source and sink. For HFP, the roles are AG and HF.

SBC (Sub-band Codec), the mandatory codec for the A2DP Profile. Other codecs are allowed.

SCO is a synchronous connection oriented channel, a full duplex data channel with 64 kbit/s data rate in each direction. The CVSD codec is used, and the three HV types of data packets are available. Also see eSCO.

sink is the term used for the device that receives audio in a uni-directional system.

source is the term used for the device that transmits audio in a uni-directional system.

SSP is Secure Simple Pairing, the pairing methods required in Bluetooth v2.1. Audio Precision supports the SSP Just Works, Numeric Comparison and Passkey Entry methods.

17

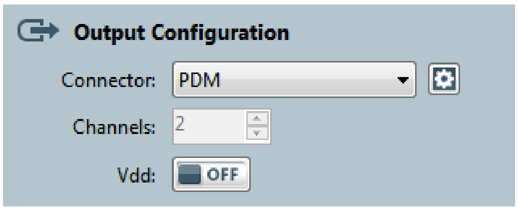
PDM I/O

Introduction

PDM I/O (input/output) requires the APx PDM Option module, described on page 9. This chapter discusses the APx500 Signal Path Setup settings for PDM output and input configurations. Also see More about PDM on page 139.

Output Configuration

PDM (when installed)



These controls are shown on Signal Path Setup when PDM is selected as the Output Connector.

Vdd (ON/OFF)

APx can supply operating power to a PDM device under test. This is available on the PCM module at the Vdd Supply BNC connector, providing DC current up to 15 mA, with a voltage range of 1.50 VDC to 3.60 VDC.

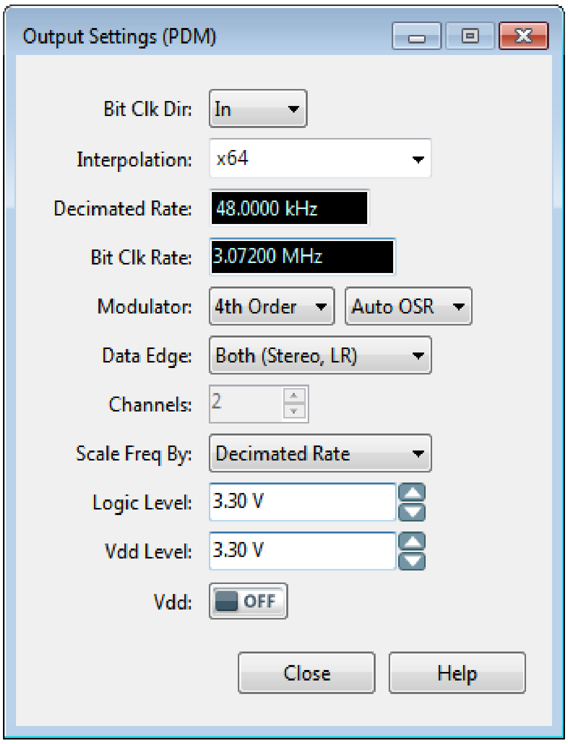
This switch turns the Vdd DC power supply On or Off.

The Vdd ON/OFF switch appears on both the PDM Output Configuration and the PDM Input Configuration panels, and the associated Settings panels.

Settings

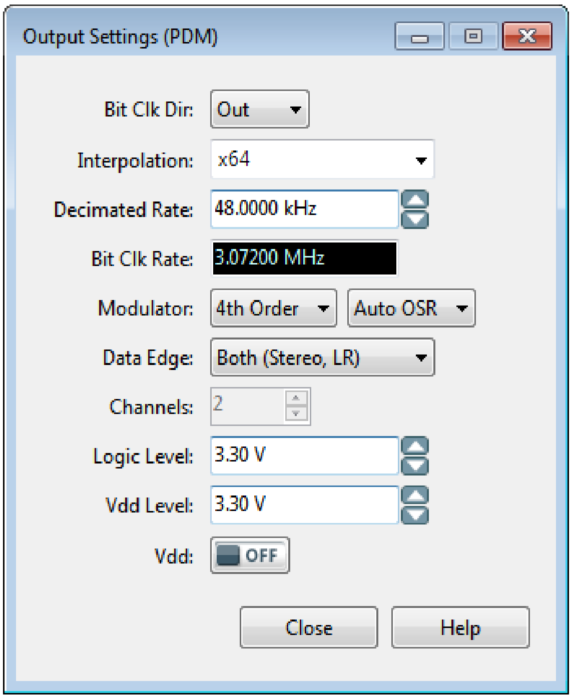
The Settings button opens the Output Settings (PDM) dialog, which reveals more PDM settings and readings.

Output Settings (PDM)



Output Settings with Bit Clock set to In (default).

These controls are in the Output Settings (PDM) dialog, available by clicking the Settings button from Signal Path Setup, when Output Configuration is set to PDM.



Output Settings with Bit Clock set to Out.

Bit Clk Dir:

The Bit Clock connector associated with the PDM Output on the PDM module can be configured as an input or an output. Choose a clock direction by selecting Bit Clk Dir: In (the default), or Out.

A typical use case provides that an upstream PDM transmitter (the APx PDM output, representing a MEMS microphone) is connected to a downstream PDM input (the DUT), and that the transmitter receives the clock from the receiver. In this case the APx output-associated clock is set to In, where APx is the slave and the DUT is the master. This is default clock direction setting.

Interpolation:

Set the transmitter interpolation factor here.

APx provides a wide range of interpolation settings to accommodate common ratios. The supported Decimated Rates (baseband audio sample rates) are constrained by the interpolation setting, as shown in the table here. Power-of-two rates are shown in bold type.

Interpolation Factor	*Maximum Decimated Rate*	*Auto OSR*
16x	216 kHz	x64
16.67x	216 kHz	x64
21.33x	216 kHz	x64
24x	216 kHz	x64
25x	216 kHz	x64
x32	216 kHz	x64
x33.3	216 kHz	x64
x37.5	216 kHz	x64
x42.67	216 kHz	x64
x48	216 kHz	x64
x50	216 kHz	x64
x64	216 kHz	x64
x66.67	216 kHz	x128
x75	216 kHz	x128
x85.33	216 kHz	x128
x96	216 kHz	x128
x100	216 kHz	x128
x128	192 kHz	x128
x150	163.84 kHz	x256
x192	128 kHz	x256
x200	122.88 kHz	x256
x256	96 kHz	x256
x300	81.92 kHz	x512
x384	64 kHz	x512
x400	61.44 kHz	x512
x512	48 kHz	x512
x600	40.96 kHz	x512
x768	32 kHz	x512
x800	30.72 kHz	x512

Decimated Rate:

This control is available when the Bit Clock connector associated with the PDM output is set to Out. The clock direction is set with the Bit Clk Dir control, above.

The Decimated Rate control specifies the decimated sample rate (the baseband audio sample rate) by setting the PDM input Bit Clock output rate. The relationship between the decimated rate and the bit clock rate depends upon the interpolation ratio, set by the Interpolation control, above. For example, a specified Decimated Rate of 48 kHz and an interpolation factor of 64 would set the PDM Bit Clock output to 3.072 MHz (48000 × 64 = 3072000).

To set the decimated rate, click the up/down arrows to select a standard rate, or enter an arbitrary value in the **Decimated Rate** field. Maximum decimated rate is shown in the adjacent table. Minimum supported decimated rate is 4 kHz. Default is 48 kHz.

This control is unavailable when the PDM output Bit Clock is set to In. In that case, the field becomes a display that shows the current decimated rate.

Bit Clk Rate:

This is a reading field. The rate of the Bit Clock (which is Decimated Rate × interpolation ratio, the factor set in Interpolation) is displayed here.

Modulator

Select the PDM modulator with this control.

APx can use a 4th-order or 5th-order PDM modulator. A 4th-order modulator is commonly used in MEMS microphone systems, and is the default. The 5th-order modulator provides better noise and distortion performance at most levels and can be selected as an alternative. See More about PDM on page 139.

Auto OSR

APx500 provides four 4th-order and four 5th-order modulators, each optimized for one of four common oversampling ratios (x64, x128, x256 and x512). When Auto OSR is selected, these modulators are mapped to other oversampling ratios as shown in the table above. You can force a different mapping by selecting one of the OSRs in the drop-down menu. The current OSR and the current modulator version are shown in the Status Bar.

Data Edge:

One channel of audio data can be carried on each edge (the rising edge or the falling edge) of the clock signal. Select **Both (Stereo, LR)**, **Both (Stereo, RL)**, **Rising** or **Falling** for the PDM transmitter setting.

This selection affects the APx output channel count. For PDM, there are 2 output channels when Data Edge is set to either **Both** setting, and 1 output channel when Data Edge is set to **Rising** or **Falling**. **Both (Stereo, LR)** routes the Channel 1 (left) audio to the rising edge, and the Channel 2 (right) audio to the falling edge; **Both (Stereo, RL)** reverses this.

Channels:

This control is unavailable, and displays the input channel count as set by the Data Edge control, above.

Scale Freq By:

This control is available when the Bit Clock connector associated with the PDM output is set to **In**.

- **Decimated Rate** is the default setting. This sets the APx generator to a sample rate that is the same as the decimated rate.
- **Fixed Rate.** This sets the APx generator to a fixed sample rate, entered below in the Fixed Rate field.

Fixed Rate:

This field is available if Scale Freq By: is set to **Fixed Rate.**

Logic Level

This control sets nominal logic level for the PDM bitstream, and the Bit Clock level when Bit Clock is an output. The range is 1.80 V to 3.30 V.

Vdd Level

This control sets the nominal DC voltage at the Vdd Supply BNC connector on the PDM module. The range is +0.8 VDC to +3.60 VDC. See **Vdd ON/OFF**, below.

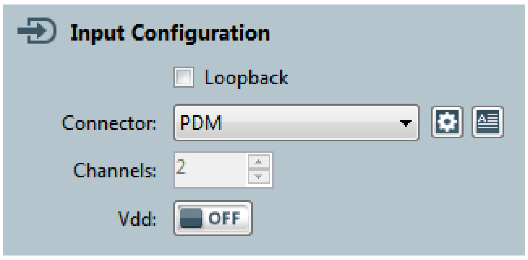
The Vdd Level control appears on both the Output Settings (PDM) Configuration and the Input Settings (PDM) panels.

Vdd ON/OFF

This switch turns the Vdd DC power supply **On** or **Off**.

The **Vdd ON/OFF** switch appears on both the PDM Output Configuration and the PDM Input Configuration panels, and the associated Settings panels.

PDM Input Configuration



Note: APx measurements are all performed on the baseband audio signal, at the Decimated Rate. However, it is possible to directly view the undecimated PDM bitstream in the Signal Analyzer, when the Signal control is set to PDM Bitstream.

These controls are shown on Signal Path Setup when **PDM** is selected as the Input Connector:

Vdd (ON/OFF)

APx can supply operating power to a PDM device under test. This is available on the PCM module at the Vdd Supply BNC connector, providing DC current up to 15 mA, with a voltage range of 1.50 VDC to 3.60 VDC.

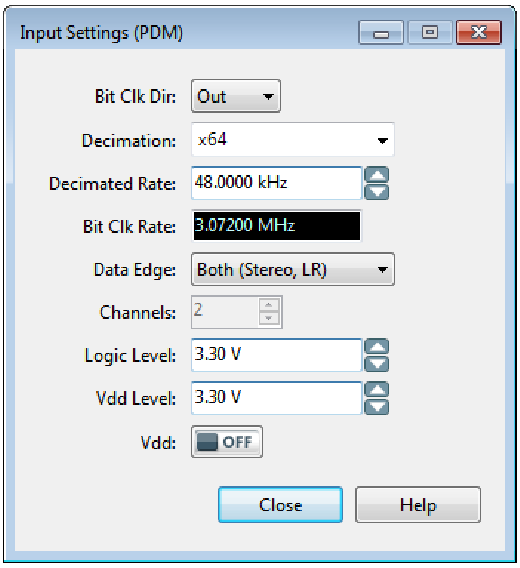
This switch turns the Vdd DC power supply **On** or **Off**.

The **Vdd ON/OFF** switch appears on both the PDM Output Configuration and the PDM Input Configuration panels, and the associated Settings panels.

Settings

The Settings button opens the Input Settings (PDM) dialog, which reveals more PDM settings and readings.

Input Settings (PDM)



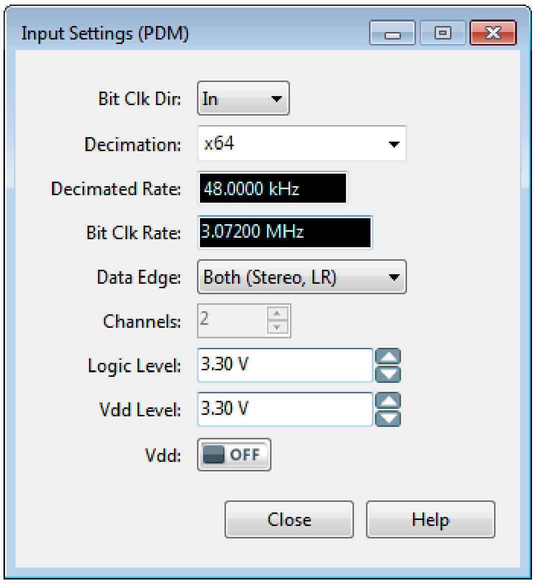
Input Settings with Bit Clock set to Out (default).

These controls are in the Input Settings (PDM) dialog, available by clicking the Settings button from Signal Path Setup, when Input Configuration is set to PDM.

Bit Clk Dir:

The Bit Clock connector associated with the PDM Input on the PDM module can be configured as an input or an output. Choose a clock direction by selecting **Bit Clk Dir: Out** (the default), or **In**.

A typical use case provides that a downstream PDM receiver (the APx PDM input) is connected to a MEMS microphone PDM output (the DUT), and that the receiver also provides the clock to the MEMS module. In this case the APx input-associated clock is set to **Out**, where APx is the master and the DUT is the slave. This is the default clock direction setting.



Input Settings with Bit Clock set to In.

Decimation:

Set the receiver decimation factor here.

APx provides a wide range of decimation settings to accommodate common ratios. The supported Decimated Rates (baseband audio sample rates) are constrained by the decimation setting, as shown in the table here. Power-of-two rates are shown in bold type.

Note: the x1 setting does not decimate the PDM signal. The undecimated PDM bitstream is presented to the APx measurements.

Decimation Factor	*Maximum Decimated Rate*
x1 (Bitstream)	216 kHz
x3.125	216 kHz
x4	216 kHz
x6.25	216 kHz
x8	216 kHz
x8.33	216 kHz
x10.67	216 kHz
x12.5	216 kHz
x16	216 kHz
x16.67	216 kHz
x18.75	216 kHz
x21.33	216 kHz
x24	216 kHz
x25	216 kHz
x32	216 kHz
x33.3	216 kHz
x37.5	216 kHz

Decimation Factor	*Maximum Decimated Rate*
x42.67	216 kHz
x48	216 kHz
x50	216 kHz
x64	216 kHz
x66.67	216 kHz
x75	216 kHz
x85.33	216 kHz
x96	216 kHz
x100	216 kHz
x128	192 kHz
x150	163.84 kHz
x192	128 kHz
x200	122.88 kHz
x256	96 kHz
x300	81.92 kHz
x384	64 kHz
x400	61.44 kHz
x512	48 kHz
x600	40.96 kHz
x768	32 kHz
x800	30.72 kHz

Decimated Rate:

This control is available when the Bit Clock connector associated with the PDM input is set to Out. The clock direction is set with the Bit Clk Dir control, above.

The Decimated Rate control specifies the decimated sample rate (the baseband audio sample rate) by setting the PDM input Bit Clock output rate. The relationship between the decimated rate and the bit clock rate depends upon the oversampling ratio, set by the Decimation control, above. For example, a specified decimated rate of 48 kHz and an oversampling ratio of 64 would set the PDM Bit Clock out to 3.072 MHz (48000 × 64 = 3072000).

To set the decimated rate, click the up/down arrows to select a standard rate, or enter an arbitrary value in the Decimated Rate field. Maximum decimated rate is shown in the above table. Minimum supported decimated rate is 4 kHz. Default is 48 kHz.

This control is unavailable when the PDM input Bit Clock is set to In. In that case, the field becomes a display that shows the current decimated rate.

Bit Clk Rate:

This is a reading field. The rate of the Bit Clock (which is Decimated Rate × decimation ratio, the factor set in Decimation) is displayed here.

Data Edge:

One channel of audio data can be carried on each edge (the rising edge or the falling edge) of the clock signal. Select Both (Stereo), Rising or Falling for the PDM receiver setting.

This selection affects the APx input channel count. For PDM, there are 2 input channels when Data Edge is set to either Both setting, and 1 input channel when Data Edge is set to Rising or Falling. Both (Stereo, LR) routes the data on the rising edge to Channel 1 (left), and the data on the falling edge to channel 2 (right); Both (Stereo, RL) reverses this.

Channels:

This control is unavailable, and displays the input channel count as set by the Data Edge control, above.

Logic Level:

This control sets the nominal logic level sensitivity for the PDM Input connector, and the Bit Clock level when Bit Clock is an output. The range is 1.80 V to 3.30 V.

Vdd Level:

APx can supply operating power to a PDM device under test. This is available on the PDM module at the Vdd Supply BNC connector, providing DC current up to 15 mA, with a voltage range of +0.8 VDC to +3.60 VDC. Set the Vdd Level here.

Vdd (ON/OFF)

This switch turns the Vdd DC power supply On or Off.

The Vdd switch appears on both the PDM Output Configuration and the PDM Input Configuration panels, and the associated Settings panels.

Coupling:

Input signals can be analyzed and displayed in one of two coupling modes:

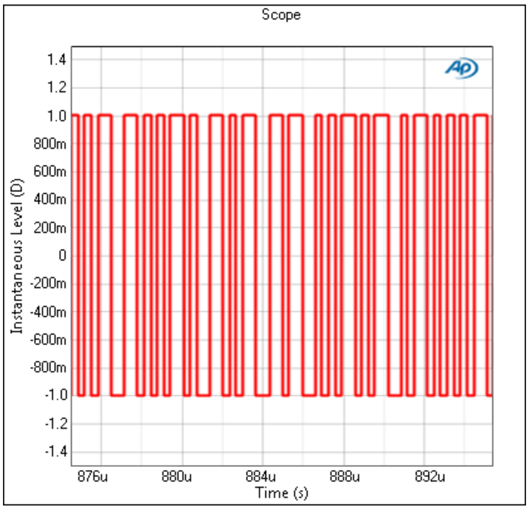
- AC
  AC coupling blocks the dc signal component from analysis, displaying results for only the ac components. As a consequence, AC coupling introduces a rolloff below 10 Hz. AC coupling is the default setting.
- DC
  DC coupling includes the dc signal component in analysis and display.

See More about DC in APx analyzers on page 240.

More about PDM

PDM is Pulse Density Modulation, a system for representing a sampled signal as a stream of single bits using delta sigma conversion (sometimes called sigma delta conversion). PDM is the technology used in many oversampling ADCs and DACs, and is the basis of the

Sony/Philips commercial digital format and disc trade named DSD and SACD, respectively.



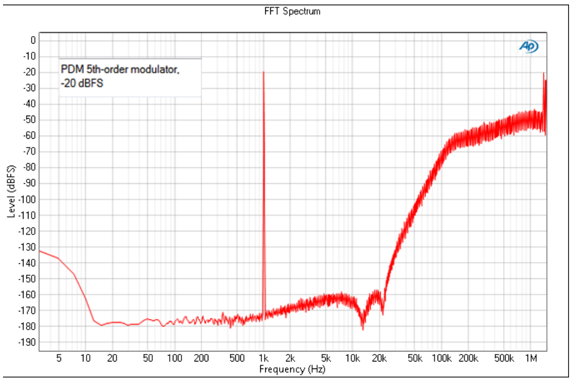
PDM bitstream, time domain view.

To configure the Signal Analyzer to display the PDM bitstream as shown above, see page 439.

A one-bit stream is unacceptably noisy, but very high sampling rates and noise shaping techniques are used to greatly reduce the noise in the audio spectrum. This noise energy is moved from the audio baseband into the area of the spectrum above 20 kHz, where it is inaudible.

Why PDM?

Properly done, PDM can digitally represent high quality audio, and is inexpensive and easy to implement. For these reasons, a PDM stream is now commonly used as the data output of a MEMS (Micro Electro-Mechanical System) microphone. MEMS microphones can be made to be very small and are inexpensive to implement on silicon chips, and are found in many small devices such as cell phones or Bluetooth devices.



Spectrum of PDM bitstream with 5th-order modulator, –20 dBFS

To configure the Signal Analyzer to display the PDM bitstream as shown above, see page 439.

Analog to PDM to Analog

An analog signal can be directly sampled at a high sampling rate (several megahertz or more) and converted to a PDM stream. The PDM signal can be converted back to analog audio by passing it through a low-pass filter.

PCM to PDM

A signal that is coded as PCM (pulse code modulation, the coding widely used in digital audio) can be converted to PDM by interpolating it and reducing the word length to one bit. The ratio of the interpolated PDM bit rate to the PCM sample rate is called the oversampling ratio.

PDM to PCM

A signal that is coded as PDM can be converted to PCM by decimating it and increasing the word length. As mentioned above, the ratio of the PDM bit rate to the decimated PCM sample rate is called the oversampling ratio.

PDM modulator

PDM modulators have a series of integrators, or accumulating nodes. The “order” of a modulator refers to the number of its integrators.

Higher-order modulators provide improved noise and distortion performance over much of the amplitude range, but suffer from increased instability at amplitudes approaching maximum level. To allow full amplitude levels without driving a PDM modulator into instability, limiting techniques are typically employed. In APx, an Overload indicator (shown in the Status Bar) is activated when the modulator exceeds the defined overload point. For the APx 4th-order modulator, the overload point is about –7.5 dBFS. For the 5th-order modulator, the overload point is about –9 dBFS. Limiting is used above this point, enabling both the 4th and 5th order modulators to operate to maximum level without instability. However, noise and distortion rise for signals above the overload point. In practical applications, such performance is deemed acceptable, since voice signals at maximum levels are typically brief transients. In APx, modulator distortion at maximum level is about 1% THD+N.

MEMS microphones

A MEMS microphone system consists of the MEMS sensor, an analog preamplifier, and a PDM modulator, often implemented on a small silicon chip. DC power and a clock signal at the sampling rate are typically provided by downstream devices.

Manufacturers of such devices typically use 4th-order sigma delta modulators at a clock frequency of 3.072 MHz, which in the receiver is typically decimated by a factor of 64 to a baseband sampling rate of 48 kHz.

Control Codes

Some PDM devices can send or receive control codes. A control code is a 2-digit hex number (8 bits) that is inserted into the PDM bit stream, replacing the audio for the duration of the control code sequence. Well-behaved devices will prevent the codes from being played as audio. Control codes are often repeated many times to assure identification and reception.

Control codes are typically used to configure a downstream PDM device. If the device is an amplifier, the code may request a mute or an unmute, or adjust the volume, or request that the device power down.

The codes and their interpretations are not standardized and are specific to manufacturers and devices.

APx implementation

The PDM Monitor reads control codes and provides access to the Send PDM Control Codes dialog. See pages 35 and 36.

18

ASIO I/O

ASIO Output and Input Configurations

For the APx515, this interface requires a software option key. See page 166 for more information about software options.

To connect APx to an ASIO device, first be sure that you have the current manufacturer's ASIO driver for your device properly installed on the PC that is running APx500, and that you have the device connected to the PC and turned on.

An ASIO device that uses USB for its physical interface should not be connected to the PC through the same root USB hub device as the APx instrument, to avoid bandwidth contention between the APx instrument and ASIO device.

Go to Signal Path Setup and select ASIO as an output or input configuration. Then select your device from the Device list.

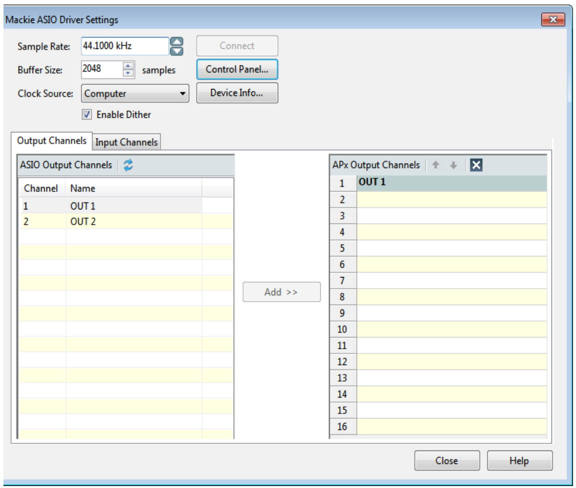
Note: in Demo Mode you can select Demo Device to inspect the ASIO User Interface.

Although many ASIO devices can be connected to a PC and displayed in the Device list, only one device can be selected as an active, connected device for each Signal Path. A different ASIO device can be selected for other Signal Paths within a Project.

Click the **Settings** button to make additional settings for your ASIO device.

ASIO Settings

There is one Settings dialog for both ASIO output and ASIO input. Click the **Output** or **Input** tab to view the corresponding channel mapping.



The ASIO Settings dialog provides a panel to set buffer size and clock source, and to map the ASIO device output and input channels to the APx analyzer output and input channels. From this dialog, you can also open the device control panel (when supported), and view device information (when supported).

The APx ASIO interface supports mapping of up to 16 input and 16 output channels.

Sample Rate

This field presents you with the supported sample rates in the currently selected ASIO audio device. Changing the sample rate here will issue a command to the ASIO device to change its sample rate.

Connect

When you have made your settings, click **Connect** to connect to the ASIO device.

Buffer Size

This field presents you with the supported buffer sizes in the currently selected ASIO audio device. Changing the buffer size here will issue a command to the ASIO device to change its buffer size.

Control Panel

Most ASIO devices provide a control panel where other parameters and readings can be addressed. Some devices provide access to their control panels through an ASIO-specified protocol. In such cases, clicking the Control Panel button will open the device control panel. Other devices may provide control panels that must be accessed in some other way, such as launching a mixing application, etc.

Clock Source

This drop-menu presents you with a list of clock sources supported by the currently selected ASIO audio device. Selecting a the clock source here will issue a command to the ASIO device to use that clock source.

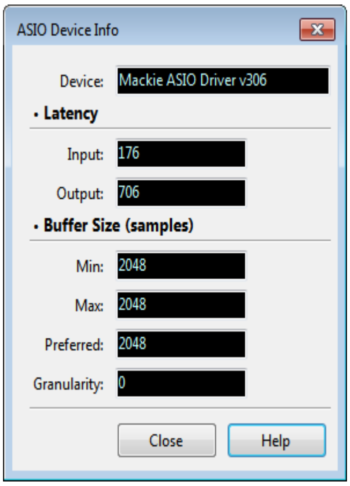
Enable Dither

When the **Enable Dither** checkbox is checked, dither is **ON** (the default). When **ON**, dither is set according to the bit depth, using TPDF (triangular probability density function) dither at ±1 LSB.

Uncheck the checkbox to turn dither **OFF**.

Device Info

Most ASIO devices provide device information that can be accessed through an ASIO-specified protocol. In such cases, clicking the Device Info button will populate the APx ASIO Device Info panel with values.



The Device info panel displays this information:

Device

The device manufacturer, model, serial number, etc. can be listed here.

Latency (samples):

Latency is the delay through the ASIO device, which includes both the physical sound device and delay occurring in consequence of PC activity such as disk access, buffering, etc. Latency is reported in samples.

Input

This is the input latency reported by the ASIO driver.

Output

This is the output latency reported by the ASIO driver.

Buffer Size (samples)

A certain amount of PC memory is allotted to the ASIO device driver to ensure that PC interrupts will not adversely affect the ASIO data. Smaller buffer sizes will decrease latency but will present a higher risk of dropped samples. Larger buffer sizes will increase latency but provides a lower risk of dropped samples. Buffer size is reported in samples, and therefore latency will change with a sample rate change.

Min:

This is the minimum buffer size supported by the ASIO driver.

Max:

This is the maximum buffer size supported by the ASIO driver.

Preferred:

This is the default buffer size of the ASIO driver.

Granularity:

This is the degree of resolution supported by the ASIO driver in specifying buffer size. This is typically “powers of 2” (32, 64, 128, etc.), which is indicated by the ASIO signifier “-1”.

ASIO Output Channels / ASIO Input channels

These two tabbed views show the ASIO device’s output and input channels. To map an output or input channel to an APx output or input channel, highlight the ASIO channel and click the **Add >>** button.

Once ASIO channels have been mapped to the APx, the mapped channels can be moved up or down in the APx Channels table, changing their mapping assignment. Channels can also be unmapped (removed) from the APx inputs or outputs by highlighting the channel and clicking the **Delete Field** button.

More about ASIO

ASIO is a license-free sound device protocol for Microsoft Windows that provides a high-quality low-latency connection to an internal sound card or external audio device (typically via USB or IEEE 1394 Firewire interface). Other common interfaces such as PCI, Ethernet, etc. are supported. The APx500 implementation enables input and output through any ASIO-compatible device. ASIO was developed and specified by Steinberg GmbH.

ASIO avoids the intermediary layers of the Windows operating system, bypassing its relatively poor quality sample rate conversion and mixing capabilities and allowing applications to have bit exact access to the audio streams from a large number of independent channels from a sound card. ASIO supports the use of 24 bit samples and sample rates up to 192K samples per second.

ASIO devices vary in their compatibility with other ASIO systems. Features such as channel count, sample rate, bit depth and overall audio quality also vary widely. Properly implemented, ASIO devices can be bit-accurate while signals remain unmolested in the digital domain; whether or not a particular device achieves this performance depends upon its design.

The APx ASIO I/O capability was added as a means to test ASIO devices, where the ASIO device itself is the DUT. Of course, an ASIO device can serve as an output for the APx generator, or as an input for the APx analyzer. For test and measurement applications, the audio quality of any ASIO device will always be inferior to the APx hardware input and outputs in the analyzer hardware, and will degrade your measurements significantly. We never recommend using an ASIO device as a replacement for the Audio Precision instrument.

19

Open Loop measurements

Overview

Most APx measurements, whether in Sequence Mode or Bench Mode, are made in the Closed Loop configuration, where the APx generator output is connected to the inputs of the DUT, and the outputs of the DUT are connected back to the analyzer.

However, this is not possible with playback-only devices such as CD or DVD players and personal audio players, or with received broadcast signals or split-site signals carried over a long distance.

APx has solutions for these circumstances, the **External Source** and **File Input** configurations.

External Source configuration

To enter External Source configuration, go to the Signal Path Setup Input/Output menu, and select **None (External)** under Output Configuration. This configures APx to make measurements without using the internal generator.

Compatibility

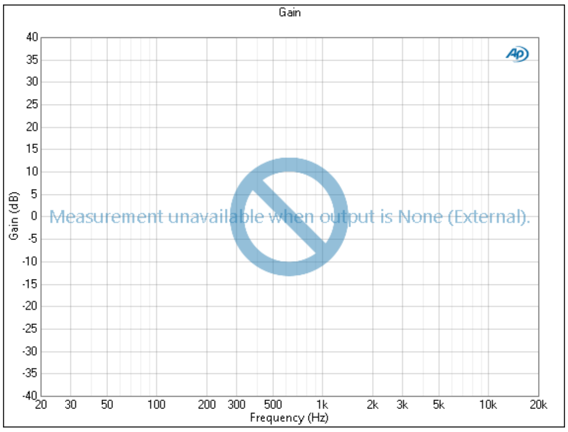
Some measurements are entirely compatible with External Source configuration, and no special considerations are required. For example, many meter measurements such as Level, Phase, Frequency and so on will operate perfectly with or without a signal from the APx generator.

Other measurements, while compatible, require special attention in External Source, such as the application of a sweep table or a multitone signal definition file.

Measurements based on continuous sweep (Continuous Sweep, Frequency Response, Acoustic Response, Impedance/Thiele-Small, Loudspeaker Production Test) require a synchronization between generator and analyzer, and are not compatible with open-loop measurements.

Level sweeps and regulated measurements, some crosstalk measurements, and any gain result depend upon a dynamic relationship between generator and analyzer, and are not compatible with open-loop measurements. Some batch measurements depend upon pre-loaded files (POLQA Averaged, for example), and are not compatible with open-loop.

When Output Configuration is set to **None (External)**, incompatible measurements and results are displayed with text and a mark, like this:



Using the External Source configuration

Setup

In Signal Path Setup Input/Output, go to Output Configuration and select **None (External)** as the Output Connector. This will set the project signal path to the External Source configuration.

No Generator

In Bench Mode, the Generator panel disappears when in **None (External)**. In Sequence Mode, compatible measurement views will no longer show the Generator, but, will display text information or a drop-down list of known source signals that you must use for the measurement. Other measurements do not require known signals, and the External Source panel provides a brief textual explanation of acceptable sources.

File Input Configuration

File Input is another open-loop configuration, where the signal from the APx Generator is not routed through a DUT and then into the Analyzer input.

APx500 software can analyze the data in an audio file accessible to the PC connected to the instrument. This is useful for analyzing DUTs that present their output as an audio file, or for remotely analyzing an audio signal.

Applications

There are a number of applications for File Input.

- The WAV files may be created and saved by the DUT during the test, using the APx500 generator as a signal source. This is typical of DUTs like a digital recorder or a sound card.
- The files may be output from a decoder device testing coded audio streams.
- The files may be from a remote testing session, saved by the Measurement Recorder, for example, to pass on a recording of the analog output of a DUT for further analysis.
- In a sequence, it may be useful to analyze the same audio data more than once using different parameters, such as filtering.

A colleague separated by time or distance could record the output of a DUT to a digital file (using the APx Save to File function or a third-party recorder) for later analysis.

A digital recorder's recording function could be analyzed as the DUT, by using its output file as an analyzer input.

A complex system such as a smart phone can be tested using a combination of External Source and File Input techniques, as described in the Audio Precision Technote 120.

Measurements that support File Input Configuration

All measurements that support the External Source configuration also support File Input configuration, with one exception: the Metadata Recorder. This is because linear WAV files do not contain metadata of interest.

Supported audio file formats

See page 542 for a table of supported audio file formats for file input analysis of audio files.

Setup

In Signal Path Setup, go to Input Configuration: Connector and choose File (Analog Units) or File (Digital Units) from the drop-down list.

Both of these menu choices configure APx500 for a WAV file input, but with different units selected for the results displays.

File (Digital Units)

File (Digital Units) displays the audio data level in the file in digital units.

File (Analog Units)

When analyzing a WAV file that represents an analog measurement acquisition, it is often convenient to view the audio level in the file in the original analog units. This requires scaling the file's digital levels to correspond to the analog levels represented.

Pre-scaled files

Analog measurement acquisitions saved as WAV files by the APx Measurement Recorder contain proprietary metadata that includes this scaling factor, which is used for displaying results when configured for File (Analog Units). You don't need to do any re-scaling.

Unscaled files

Files not recorded by the Measurement Recorder can be scaled by the factor entered in the Vrms/FS field. With the default scaling factor of 1.0, for example, a digital level of 1.000 FS (0 dBFS) will be scaled to a display value of 1.000 Vrms.

Operation

Sequence Mode

When File Input has been selected in Sequence Mode, each compatible measurement will display a File List: Choose Files button to open the Input File List dialog. Add the files you want to analyze in the Input File List.

When the files are chosen, use the Analyze button to make the measurement.

Bench Mode

In Bench Mode, use the File List button in Signal Path Setup to open the Input File List dialog. The analog scaling field Vrms/FS is located just above the File List button. Add the files you want to analyze in the Input File List.

When the files are chosen, use the Analyze in a measurement panel button to make the measurement.

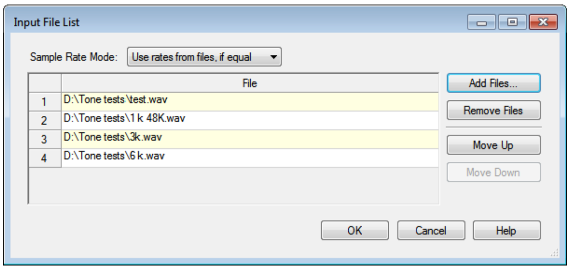
Configuring the DUT

If recording from a DUT, set it up to save WAV files to the location and with the file names specified in the Input File List. Turn on the Generator to feed a signal to the DUT, if necessary, and configure the DUT to save the WAV file(s).

If you are measuring an existing file (not recording from the generator but working in External Source), be sure that the file name(s) and location match the name(s) specified in the File List.

Finally, click the **Analyze** button. The first settled reading (typically within the first 0.10 second of the file) is displayed. To take a reading later in the file, go to the **Settling** tab in **Advanced Settings** or **Settled Readings** and increase the **Delay Time** to the desired point in the file.

Input File List



The **File List...** button opens the **Input File List** dialog.

The Input File List allows you to select WAV files for analysis. Adding a file to this list does NOT load the file into memory, but it inserts a pointer to the file location into the list. You can even add a file name for a WAV file that does not yet exist on the disk (see **Specifying files that do not yet exist**, below). When the **Analyze** button on the measurement panel is clicked, each file is loaded into memory, analyzed, and then cleared from memory.

Sample Rate Mode

You can choose one of three modes to manage the sample rate used to play the input files for analysis.

- **Use rates from files, if equal** (the default)
  In this mode, all files are required to have exactly the same embedded sample rate. Files are played at that rate. An error is flagged during analysis if files of mixed sample rate are selected.
- **Use rate from first file**
  In this mode, all files are played at the sample rate of the first file in the list. If subsequent files have other sample rates, their audio content will be pitch-shifted.
- **Use fixed rate**
  In this mode, all files are played at the sample rate set in the Fixed Rate field. If files have other sample rates, their audio content will be pitch-shifted.

Action Buttons

- The **Add Files...** button opens a file browser for file selection. Select one or more files to add to the list.
- The **Remove Files** button removes highlighted files from the list.
- The **Move Up** and **Move Down** buttons change the list position of a highlighted file.

When all the desired files are in the list, click **OK** to close the dialog and return to the measurement panel, where you can continue and analyze the files.

Specifying files that do not yet exist

It is possible to add to the list WAV files that do not yet exist on the hard drive. This is useful for setting up a test that involves a DUT (such as a digital recorder) that is expected to save a WAV file during the test procedure.

You can specify a file that does not exist by typing in the **File Name:** field in the **Add Files...** browser, or by editing the text in the **Input File List**. Remember to include a valid path to the expected file.

Maximum number of files and channels

The **Input File List** will accept a maximum of 16 files.

Since each file in the **Input File List** could have 1 or 2 channels of audio, it is possible that an acceptable file list (fewer than 16 files) could exceed the analyzer audio channel count maximum, displaying an error message at analysis. If this occurs, remove 1 or more files from the **Input File List**.

Supported file characteristics

- Mono (single channel) or stereo (dual channel) 24-bit WAV files containing linear PCM can be analyzed. Each file, mono or stereo, occupies one line in the list.
- For most measurements, files can have any audio content. For Stepped Frequency Sweep, only compliant sweeps will provide useful results.
- If Sample Rate Mode is **Use rates from files, if equal** (the default), all the files must be of the same sample rate.
- Files can be of any length up to the maximum WAV file length (4 GB). Files can be of mixed length; however, analysis stops at the end of the shortest file.

Errors

Errors may be flagged during analysis:

- if a file on the list does not exist;

- if a file on the list is not a valid linear WAV file;
- if the number of channels in the listed files exceeds 16;
- if files of mixed sample rate are selected, and the Sample Rate Mode is **Use rates from files, if equal** (the default).

External Source Test Signals

audio Precision provides a collection of external stimulus signals in a several different distributions. These include playable media discs and a collection of files optimized for generator waveform (arbitrary waveform) use with the APx generator. For the stepped frequency sweeps in these collections, Sweep Tables are embedded in the APx software.

The small Audio Precision utility “WfmGeneration.exe” is a convenient way to create any of these files (or a large part of the collection) at a specified resolution and sample rate. WfmGeneration.exe is available as a download from the AP Web site.

Audio Precision Test Signals on Disc

...for External Source use

We provide a full suite of test signals for external source as audio tracks on DVD and audio CD. The audio CD signals are linear audio at 44.1 kHz sample rate and 16-bit resolution; the DVD includes other sample rates and Dolby and dts coded audio as well. The signals are compatible with various measurement views in APx500.

Many of these test signals are also available as linear format audio files at various sample rates and resolutions, using the Waveform Generator utility provided on the Audio Precision Web site at ap.com.

- Playable test signals on video DVD: APx-DVD1
- Playable test signals on audio CD: APx-CD1

...for Generator Waveform use

Versions of these test signals optimized for Generator Waveform use are also available on the Audio Precision Web site at ap.com. Coded files are provided on disc; linear files can be created using the Waveform Generator utility. See the documentation in the Audio Test Signals folder.

...making your own test signals

For many measurements, you can make your own test signals for external source testing. See Creating Your Own Test Signals on page 19 for guidelines.

Audio Precision Test Signals on DVD: APx-DVD1

These test signals are provided as video-menu-driven audio tracks on the APx-DVD1. Video is 720p 30 frame. Formats available are:

- Linear PCM 48 kHz & 96 kHz (2 channels)
- Dolby Digital @ 48 kHz (5.1 channels)
- dts Digital Surround @ 48 kHz (5.1 channels)

All signals provide the same audio on both channels except as noted for Channel ID, Reference Level with LFE, and Crosstalk signals.

Bit test notes

The Bit test signals are prepared differently across the formats.

Bit test

For linear PCM, both channels carry the Bit test random signal. This signal uses the same algorithm used by the APx500 and by other Audio Precision instruments, and it is compatible with all Audio Precision Bittest Random analysis. These tracks provide a stereo Bit test signal that maintains bit accuracy.

Bit test (bitstream)

It is possible to bypass an encoder and insert a Bit test signal directly into a coded audio bitstream. Since lossy codecs cannot maintain bit-accuracy through an encode/decode cycle, this is the only way to test bit-accuracy for Dolby Digital and DTS Digital Surround paths.

A Bit test (bitstream) signal must be played directly into the APx digital input using no decoder to maintain bit accuracy.

APx-DVD1: Audio Test Signals on DVD

Test Signal Name	*Length*	*Description*
Channel_ID_2	0:14	997 Hz @ -20 dBFS. Channel 1 only for 5 seconds, followed by Channel 2 only for 2 seconds. Repeats.
Reference_Level_0dB	1:00	997 Hz @ 0 dBFS
Reference_Level_-20dB	1:00	997 Hz @ -20 dBFS
Reference_Level_-60dB	1:00	997 Hz @ -60 dBFS
Silence_dithered	1:00	Digital zero with triangular dither
Silence_undithered	1:00	Digital zero
MOD-SMPTE_4-to-1	1:00	60 Hz + 7 kHz, 4:1
MOD-SMPTE_1-to-1	1:00	60 Hz + 7 kHz, 1:1

APx-DVD1: Audio Test Signals on DVD		
Test Signal Name	***Length***	***Description***
MOD_10-to-1	1:00	60 Hz + 7 kHz, 10:1
DFD	1:00	12.5 kHz mean, 80 Hz difference (12460 Hz + 12540 Hz)
Crosstalk_left_only	1:00	9.997 kHz @ 0 dBFS, L
Crosstalk_right_only	1:00	9.997 kHz @ 0 dBFS, R
Crosstalk_both_channels	1:00	9.997 kHz @ 0 dBFS, L+R
Freq_sweep_11_0dB	0:18	997 Hz 2 sec, followed by 11 pt sweep 20 Hz-20 kHz @ 0 dBFS
Freq_sweep_11_-1dB	0:18	997 Hz 2 sec, followed by 11 pt sweep 20 Hz-20 kHz @ -1 dBFS
Freq_sweep_11_-20dB	0:18	997 Hz 2 sec, followed by 11 pt sweep 20 Hz-20 kHz @ -20 dBFS
Freq_sweep_31_0dB	0:42	997 Hz 2 sec, followed by 31 pt sweep 20 Hz-20 kHz @ 0 dBFS
Freq_sweep_31_-1dB	0:42	997 Hz 2 sec, followed by 31 pt sweep 20 Hz-20 kHz @ -1 dBFS
Freq_sweep_31_-20dB	0:42	997 Hz 2 sec, followed by 31 pt sweep 20 Hz-20 kHz @ -20 dBFS
Freq_sweep_61_0dB	1:18	997 Hz 2 sec, followed by 61 pt sweep 20 Hz-20 kHz @ 0 dBFS
Freq_sweep_61_-1dB	1:18	997 Hz 2 sec, followed by 61 pt sweep 20 Hz-20 kHz @ -1 dBFS
Freq_sweep_61_-20dB	1:18	997 Hz 2 sec, followed by 61 pt sweep 20 Hz-20 kHz @ -20 dBFS
Slow_freq_sweep_11_0dB	0:36	997 Hz 2 sec, followed by 11 pt slow sweep 20 Hz-20 kHz @ 0 dBFS
Slow_freq_sweep_11_-1dB	0:36	997 Hz 2 sec, followed by 11 pt slow sweep 20 Hz-20 kHz @ -1 dBFS
Slow_freq_sweep_11_-20dB	0:36	997 Hz 2 sec, followed by 11 pt slow sweep 20 Hz-20 kHz @ -20 dBFS
Slow_freq_sweep_31_0dB	1:36	997 Hz 2 sec, followed by 31 pt slow sweep 20 Hz-20 kHz @ 0 dBFS
Slow_freq_sweep_31_-1dB	1:36	997 Hz 2 sec, followed by 31 pt slow sweep 20 Hz-20 kHz @ -1 dBFS
Slow_freq_sweep_31_-20dB	1:36	997 Hz 2 sec, followed by 31 pt slow sweep 20 Hz-20 kHz @ -20 dBFS
Slow_freq_sweep_61_0dB	3:06	997 Hz 2 sec, followed by 61 pt slow sweep 20 Hz-20 kHz @ 0 dBFS
Slow_freq_sweep_61_-1dB	3:06	997 Hz 2 sec, followed by 61 pt slow sweep 20 Hz-20 kHz @ -1 dBFS
Slow_freq_sweep_61_-20dB	3:06	997 Hz 2 sec, followed by 61 pt slow sweep 20 Hz-20 kHz @ -20 dBFS
Level_sweep	0:49	997 Hz @ -20 dBFS for 2 seconds followed by 400 Hz, 0 dBFS to -110 dBFS, 5 dB steps, 2 sec. per point, both channels
Multitone_32	1:00	Multitone with approximate ISO 1/3 octave frequencies from 16 Hz to 20 kHz, with each tone @ -22 dBFS
Bit test Bit test (bitstream)	0:30	Bit test linear form on LPCM tracks; Bit test (bitstream) form on Dolby Digital and DTS Digital Surround tracks.

Audio Precision Test Signals on CD: APx-CD1

These test signals are provided as compact disc audio tracks on APx-CD1. Format available is:

- Linear PCM 44.1 kHz, 16 bit (2 channels)

All signals provide the same audio on all channels except as noted for Channel ID, Reference Level with LFE, and Crosstalk signals.

Bit test notes

Both channels carry the Bit test random signal. This signal uses the same algorithm used by the APx500 and by other Audio Precision instruments, and it is compatible with all Audio Precision Bittest Random analysis. These tracks provide a stereo Bit test signal that maintains bit accuracy.

Note: CD audio has a bit depth of 16 bits. For bit accurate Bit test playback, you must set the analyzer input to 16 bits.

APx-CD1: Audio Test Signals as CD audio tracks

Signal Name	***Track***	***Length***	***Description***
Channel_ID_2	01	0:14	997 Hz @ −20 dBFS. Channel 1 only for 5 seconds, followed by Channel 2 only for 2 seconds. Repeats.
Reference_Level_0dB	02	1:00	997 Hz @ 0 dBFS
Reference_Level_-20dB	03	1:00	997 Hz @ −20 dBFS
Reference_Level_-60dB	04	1:00	997 Hz @ −60 dBFS
Silence_dithered	05	1:00	Digital zeros with triangular dither
Silence_undithered	06	1:00	Digital zeros
MOD-SMPTE_4-to-1	07	1:00	60 Hz + 7 kHz, 4:1
MOD-SMPTE_1-to-1	08	1:00	60 Hz + 7 kHz, 1:1
MOD_10-to-1	09	1:00	60 Hz + 7 kHz, 10:1
DFD	10	1:00	12.5 kHz center, 80 Hz difference (12460 Hz + 12540 Hz)
Crosstalk_left_only	11	1:00	9.997 kHz @ 0 dBFS, L
Crosstalk_right_only	12	1:00	9.997 kHz @ 0 dBFS, R
Crosstalk_both_channels	13	1:00	9.997 kHz @ 0 dBFS, both L and R
Freq_sweep_11_0dB	14	0:18	997 Hz 2 sec, followed by 11 pt sweep 20 Hz–20 kHz @ 0 dBFS
Freq_sweep_11_-1dB	15	0:18	997 Hz 2 sec, followed by 11 pt sweep 20 Hz–20 kHz @ −1 dBFS
Freq_sweep_11_-20dB	16	0:18	997 Hz 2 sec, followed by 11 pt sweep 20 Hz–20 kHz @ −20 dBFS
Freq_sweep_31_0dB	17	0:42	997 Hz 2 sec, followed by 31 pt sweep 20 Hz–20 kHz @ 0 dBFS
Freq_sweep_31_-1dB	18	0:42	997 Hz 2 sec, followed by 31 pt sweep 20 Hz–20 kHz @ −1 dBFS
Freq_sweep_31_-20dB	19	0:42	997 Hz 2 sec, followed by 31 pt sweep 20 Hz–20 kHz @ −20 dBFS
Freq_sweep_61_0dB	20	1:18	997 Hz 2 sec, followed by 61 pt sweep 20 Hz–20 kHz @ 0 dBFS
Freq_sweep_61_-1dB	21	1:18	997 Hz 2 sec, followed by 61 pt sweep 20 Hz–20 kHz @ −1 dBFS
Freq_sweep_61_-20dB	22	1:18	997 Hz 2 sec, followed by 61 pt sweep 20 Hz–20 kHz @ −20 dBFS
Slow_freq_sweep_11_0dB	23	0:36	997 Hz 2 sec, followed by 11 pt slow sweep 20 Hz–20 kHz @ 0 dBFS
Slow_freq_sweep_11_-1dB	24	0:36	997 Hz 2 sec, followed by 11 pt slow sweep 20 Hz–20 kHz @ −1 dBFS
Slow_freq_sweep_11_-20dB	25	0:36	997 Hz 2 sec, followed by 11 pt slow sweep 20 Hz–20 kHz @ −20 dBFS
Slow_freq_sweep_31_0dB	26	1:36	997 Hz 2 sec, followed by 31 pt slow sweep 20 Hz–20 kHz @ 0 dBFS

APx-CD1: Audio Test Signals as CD audio tracks			
Signal Name	***Track***	***Length***	***Description***
Slow_freq_sweep_31_-1dB	27	1:36	997 Hz 2 sec, followed by 31 pt slow sweep 20 Hz–20 kHz @ –1 dBFS
Slow_freq_sweep_31_-20dB	28	1:36	997 Hz 2 sec, followed by 31 pt slow sweep 20 Hz–20 kHz @ –20 dBFS
Slow_freq_sweep_61_0dB	29	3:06	997 Hz 2 sec, followed by 61 pt slow sweep 20 Hz–20 kHz @ 0 dBFS
Slow_freq_sweep_61_-1dB	30	3:06	997 Hz 2 sec, followed by 61 pt slow sweep 20 Hz–20 kHz @ –1 dBFS
Slow_freq_sweep_61_-20dB	31	3:06	997 Hz 2 sec, followed by 61 pt slow sweep 20 Hz–20 kHz @ –20 dBFS
Level_sweep	32	0:49	997 Hz @ –20 dBFS for 2 seconds followed by 400 Hz, 0 dBFS to –110 dBFS, 5 dB steps, 2 sec. per point, both channels
Multitone_32	33	1:00	Multitone with approximate ISO 1/3 octave frequencies from 16 Hz to 20 kHz, with each tone @ –22 dBFS
Bit test	34	0:30	Bittest Random signal on each track. Set analyzer input to 16 bits.

Creating your own test signals

Some of the measurement views in External Source configuration do not require specific known test signals. Suggestions for custom signals for these views follow:

- The Level measurement view will measure the RMS level of any audio signal within the input range of the analyzer.
- The THD+N measurement view will measure the THD+N of any periodic audio signal within the input range of the analyzer. A sound wave is called periodic if the wave pattern repeats continuously. Sine waves, square waves and other repetitive waves are periodic.
- The Crosstalk measurement view will measure the degree to which the signal on one channel is present on another. This requires external source signals that by design have signal present or absent on individual channels in a known way, and that the "mode" set in Crosstalk Signal Acquisition and Analysis corresponds to the signal and to your intent. In External Source configuration, the analyzer software examines the inputs signals to determine driven and undriven channels.
  A crosstalk signal is typically high-frequency (we use approximately 10 kHz), because the most common crosstalk mechanism is capacitive coupling, which increases with frequency. Other frequencies can be used.
- The Interchannel Phase measurement view will measure the phase between channels for any periodic audio signal within the range of the analyzer. A sound wave is called periodic if the wave pattern repeats continuously. Sine waves, square waves and other repetitive waves are periodic.
- The DC Level measurement view requires no audio signal. If a DUT has a dc offset with no audio signal, it will be measured by the analyzer. However, it is possible that a DUT's dc offset might change in the presence of signal, and any audio signal can be used during the measurement of DC Level. The DC Level meter provides excellent ac rejection.
- The DFD IMD measurement view requires a twin-tone signal, with a pair of tones of equal level set apart by a difference frequency, the pair centered around a mean frequency. The difference frequency must be in the range from 80 Hz to 2 kHz; we use 80 Hz. The mean frequency choice depends to some extent on the difference frequency choice, since the maximum range is the sum and the difference of the two. Mean frequency minimum is 2.5 kHz, maximum is 20 kHz; we use 12.5 kHz. See page 273 for more information about IMD.
- The SMPTE IMD and MOD IMD measurement views require a twin-tone signal. The higher frequency must be in the range from 2 kHz to 20 kHz; we use 7 kHz. The lower frequency choice depends to some extent on the higher frequency choice, with a range from a minimum of 40 Hz to a maximum between 333.33 Hz to 1 kHz; we use 60 Hz. To satisfy industry standards, for MOD the ratio [low:high] of the level of these tones must be 1:1, 4:1 or 10:1; for SMPTE, the ratio must be 1:1 or 4:1. See page 273 for more information about IMD.

- The Signal Analyzer measurement view will provide time domain or frequency domain views of any audio signal within the input range of the analyzer.

The Measurement Recorder measurement view will provide a record of RMS level versus elapsed time for any audio signal within the input range of the analyzer. The Measurement Recorder also provides THD+N records, which require a periodic waveform.

Creating custom stimulus signals

Stepped Frequency Sweeps

You can create custom Sweep Tables and corresponding WAV files using the dialogs available from the Stepped Frequency Sweep and Bandpass Frequency Sweep measurements.

Multitones

You can create custom multitone signal definitions and corresponding WAV using the dialogs available from the Multitone measurement.

Using third-party tools

For high-performance testing, stimulus signals must be properly constructed and very pure; that is, they must have very low inherent distortion, noise or spurious components. It is possible to create digital files that satisfy these requirements using third-party software such as Matlab, Adobe Audition or Audacity, but care must be taken.

Section III: Measurements

20

Making Measurements

Overview

This chapter contains general information about making measurements in APx500, and about using the common settings and controls in the APx500 generator and analyzer.

For a more focused look, go to chapters 21 through 78, which discuss each measurement in detail. The Navigator and the Selector are covered in Chapter 3. Chapters 93 and 95 look closely at Results. Reports are covered in Chapter 81, as is Automation in Chapter 80.

Sequence Mode or Bench Mode?

Sequence Mode and Bench Mode present two different approaches to audio testing.

If you would prefer pre-defined measurements and results with easy automation and reporting, work in Sequence Mode.

On the other hand, if you are interested in specifying your own generation and analysis tools and working in a more free-form, stimulus and response method, give Bench Mode a try.

Or, mix it up. Run some Sequence Mode measurements, then look at interesting discoveries in Bench Mode, all in the same project.

How to choose measurements

APx500 provides many audio measurements for you to choose from. Which should you use? The answer to that question depends on a number of factors, such as:

The nature of the device under test (DUT)

The type of DUT you are testing will determine which measurements are appropriate. Most audio devices can be evaluated by the “Big Six” set of measurements (see page 157), for example, but if your DUT is a single-channel device, it cannot be tested for crosstalk.

Specified measurement requirements

You may have a requirement to make a specific measurement, such as SMPTE IMD.

The required testing speed

In a production environment, you may want the complete set of measurements to run very quickly. In this case, select only measurements that are essential, and be sure to choose the fastest methods.

The “Big Six”

Audio engineers agree that there is a small group of measurements that will reveal the key characteristics and performance of almost any audio device. At Audio Precision we talk about the “Big Six” measurements as the benchmark set of audio tests.

Level

The basic level measurement provides the output level or amplitude of an audio device. Depending on the device and your test configuration, the result might be displayed in volts, watts, dBV or other level units.

Gain gives the ratio of the DUT output level to its input level.

THD+N

THD+N is an abbreviation for total harmonic distortion plus noise. THD+N measurements have been used for many years as a comprehensive single-value statement of an audio device’s overall performance.

Frequency Response

A frequency response test reveals how a device “responds” across the audible spectrum. For most devices, a “flat” response curve, where all frequencies are passed with the same gain, is desirable.

Signal to Noise Ratio

Signal to Noise Ratio or SNR gives a single value that indicates how “noisy” an audio device is. Two measurements are actually required: the first measures the DUT output level (usually at either maximum level or at a nominal operating level); the second measures the residual noise in the device. The signal level is divided by the noise level, and the resulting ratio is usually expressed in decibels.

Crosstalk

Crosstalk is an expression of the degree of signal leakage between DUT channels. Ideally, none of the signal in one channel will appear in a second channel, but in practical devices there is always measurable crosstalk. Crosstalk often varies with frequency.

Phase

Phase is a measure of the lagging or leading of an alternating current waveform with regard to a reference waveform.

Interchannel phase measures the phase relationships between channels in a multichannel device. One channel is chosen as the reference.

Other measurements

Note: the APx515 requires software option keys for some measurements. *See page 166.*

APx500 performs many measurements beyond the Big Six, including dc level measurements, intermodulation distortion (IMD) measurements, regulated measurements, tone burst measurements and more. There is also a Signal Analyzer tool with Scope and FFT Spectrum displays at very high resolution (FFT length up to 1 million samples).

Measuring using Meter measurements

Many of these methods measure streaming audio, where the generator continuously outputs a stimulus signal, and the analyzer measures the DUT output in real time using a Meter measurement. Meter measurements have single value (meter) results and are displayed as a Bar Graph and a numerical value.

Meter measurements have a ON / OFF button on the Generator panel. Meter measurements typically have single value (meter) results.

Stimulus signals for meter measurements include:

- Sine wave;
- Mixed twin tone sine wave;
- Special Sine wave;
- Square wave;
- Mixed square wave and sine wave;
- Generator waveform (an arbitrary waveform played from a file on disk);
- Diagnostic digital signal.

Sine wave

Sine waves are generated in APx DSP (digital signal processing), and are the default waveform for most measurements. For analog configurations, the default frequency is 1 kHz; the default level is 100.0 mVrms. For digital configurations, the default level is –20.000 dBFS. Most APx measurements offer a sine wave.

The High Performance Sine Generator in the APx555 can generate sine waves of lower distortion and wider frequency range than the DSP-generated sine waves used by the DAC generator.

Mixed twin tone sine wave

Mixed twin tone sine waves are generated in APx DSP, and are available for IMD measurements (Chapter 44). The frequencies of the two tones, and for some measurements, the level ratio of the two tones can be set in the corresponding IMD measurement settings panel. The output level is the rms level of the complex waveform that is the sum of the two component tones. For analog configurations, the default level is 100.0 mVrms. For digital configurations, the default level is –20.000 dBFS.

Special Sine

You have the option of choosing Split Sine, Split Phase or Sine + Offset generator waveforms for the following measurements:

• Reference Levels	• Dynamic Range AES17	• Frequency
• Interchannel Phase	• Level and Gain	• Maximum Output
• Measurement Recorder	• Signal Analyzer	• Signal to Noise Ratio
• Stepped Level Sweep	• THD+N	• CMRR-IEC
• CMRR-IEC		

For these measurements, a single sine wave is the default generator waveform. A Special Sine selection allows you to output sine waves of two different frequencies to different channels, sine waves of the same frequency but different phase to different channels, or sine waves with DC offset.

Split Frequency

Split Frequency allows you to assign a sine wave of Frequency A and a sine wave of Frequency B to the output channels. By default, Frequency A is assigned to odd channel numbers and Frequency B to even channel numbers, but these can be reassigned in the Advanced Settings dialog for the measurement.

Split Phase

Split Phase allows you to assign a sine wave of Phase A (the reference at 0 degrees) and a sine wave of Phase B to the output channels. By default, Phase A is assigned to odd channel numbers and Phase B to even channel numbers, but these can be reassigned in the Advanced Settings dialog for the measurement.

Enter a value for Phase B in degrees. The range is –180.00 degrees to +179.99 degrees. A negative value sets the B waveform to lag the A waveform; a positive value sets the B waveform to lead the A waveform.

Split (IMD)

For IMD measurements, the Split IMD setting defeats the summing of the two IMD frequencies within the generator. Instead, the signals are “split” and routed independently to channel outputs.

Acoustic testing of microphones sometimes uses two stimulus loudspeakers, each with a single frequency component of an IMD stimulus. The stimulus signals are summed acoustically at the microphone element. This method provides a distortion result for the microphone while minimizing the distortion contributions of the loudspeakers.

Offset

The Offset filed allows you to add a DC offset to the generator output. If the generator level is set to zero (0.000 Vrms or 0 FS), the generator can optionally output DC only (pure DC).

For analog output, when sine level is zero the maximum

offset is $\sqrt{2}$ times the maximum RMS level (±) for the current instrument and output connector.

For digital output, when sine level is zero the maximum offset is 1 D (±).

Offsets can be set at different levels across the channels in Advanced Settings. Unset Track First Channel and enter the desired values in the Offset column of the Signal Generation grid.

Square wave

A fast rise time square wave is available in the APx analog generator for the APx555 and analyzers equipped with the AG52 option (page 5) installed. AG52 can be fitted in the APx582 and in the APx52x family of analyzers.

A DSP-generated square wave is available in the APx digital generator for all APx analyzers.

When the hardware or output configuration allows, the square wave is available for the following measurements:

- Frequency
- Level and Gain
- Measurement Recorder
- Signal Analyzer

Mixed square and sine wave (DIM)

A special stimulus signal (the sum of a square wave and a low amplitude sine wave) is required for the DIM IMD measurement (chapters 39 and 40). DIM measurements are only supported in the APX555 and analyzers with the AG52 option (page 5) installed.

Generator waveform

In addition to the DSP-generated waveforms, APx500 has the capability to use custom (or *arbitrary*) waveforms, loaded into the generator from waveform files available to the computer running APx500. This is particularly useful when coded audio files (such as dts, Dolby or others) are required to stimulate a decoder in your DUT. See page 163.

For perceptive audio measurements, speech sample files are used, and can be opened as generator waveform files. See More About POLQA on page 413, and More About PESQ on page 402.

Diagnostic digital signals

Certain digital signals are useful to diagnose or characterize digital transport systems. In APx500, these include

- Walking Zeros
- Walking Ones
- Constant Value
- Bit test

See page 48 for more information about these signals.

Reading Rate

APx meters have a reading rate, the rate at which the meter reads the incoming data and determines and displays the measured value. Faster reading rates put a higher demand upon the PC, and in some circumstances may affect PC performance. The reading rate also has an effect on analyzer performance at low frequencies: each reading rate has a low-frequency point below which analyzer performance falls off. THD+N performance degrades rapidly below this point, with level accuracy degrading a little more slowly.

Faster rates are required to resolve short-lived phenomena; slower rates are required to accurately read values at lower frequencies.

Monitor Meters

Bench Mode meter monitors have a fixed reading rate of 4/s, with accuracy falling off below 12 Hz.

Recorders

A reading is taken at the beginning of the record and at (approximately) the reading rate throughout the record. The final reading is taken at the reading instant that equals or exceeds the sweep end time.

Sequence Mode

Sequence Mode Measurement Recorder can have the reading rate set to a number of fixed choices. Accuracy limits are listed here, by reading rate and frequency:

at 20/sec > 60 Hz

at 10/sec > 30 Hz (the default)

at 5/sec > 15 Hz

at 3/sec > 9 Hz

at 2/sec > 6 Hz

at 1/sec > 3 Hz

at .5/sec > 1.5 Hz

at .2/sec > .6 Hz

at .1/sec > .3 Hz

Bench Mode

Bench Mode Recorder can have the reading rate set to a number of fixed choices. Accuracy limits are listed here, by reading rate and frequency:

at 256/sec > 1536 Hz

at 128/sec > 768 Hz

at 64/sec > 184 Hz

at 32/sec > 96 Hz (the default)

at 16/sec > 48 Hz

at 8/sec > 24 Hz

at 4/sec > 12 Hz

at 2/sec > 6 Hz

at 1/sec > 3 Hz

Bench Mode Sweep

Bench Mode Sweep can have reading rate set to the same fixed choices as Bench Mode Recorder, up to a maximum of 32/sec. An Auto selection is added to the Sweep. The default is Auto.

Auto reads at a rate of 32/s at higher audio frequencies. At lower sweep frequencies the rate is reduced to maintain accuracy.

Measuring using Batch measurements

Other methods are “batch” measurements, where the stimulus is a signal or series of signals of a certain duration, and the analyzer acquires DUT output that corresponds to the stimulus. Batch measurements have a Start button on the Measurement Settings panel. Batch measurements typically have both single value (meter) results and XY graph results.

Signals for batch measurements

Stimulus signals for batch measurements include:

Sweeps:

- Continuous sweep (log-swept sine or *chirp*)
- Stepped sine sweep
- Stepped mixed twin tone sweep
- Regulated sine sweep

Sequential measurements:

- sine on sequential channels, for crosstalk
- sine at two sequential levels, for SNR or Dynamic Range
- regulated sine, for Maximum Output

Digital diagnostic signals:

- Bit test
- Generator waveform (an arbitrary digital diagnostic signal played from a file on disk).

Multitone signals.

Sweeps

Stepped sweeps are a classic method of audio testing, where one parameter is moved (swept) across a range of values while one or more other parameters are measured. The results are usually displayed on an XY graph, with the swept parameter on the X-axis and the measured results on the Y-axis.

Continuous sweep

A continuous sweep is a brief log-swept sine wave (sometimes called a log chirp) that moves continuously across a specified range of frequencies. The DUT output is acquired by the analyzer and is mathematically processed to provide a number of results.

Continuous sweep stimulus signals are used in the Acoustic Response and Continuous Sweep measurements in both Sequence Mode and Bench Mode, and in Frequency Response, Loudspeaker Production Test and Impedance Thiele-Small measurements.

Stepped sweeps

In practical measurements, the sweep is usually stepped across a number of discrete points rather than moved continuously. This provides precision in the determination of point values, and allows the use of settling algorithms to wait for stable data. Bench Mode has a general purpose, configurable Sweep engine.

APx Sequence Mode stepped sweeps include:

Stepped Frequency sweep measurement

The Stepped Frequency Sweep (chapter 76) measurement provides a sine wave stimulus signal that is moved across a range of frequencies in a specified number of points. The DUT output is acquired by the analyzer and processed for display.

Stepped level sweep measurement

The Stepped Level Sweep (chapter 77) measurement provides a sine wave stimulus signal that is moved across a range of levels in a specified number of points. The DUT output is acquired by the analyzer and processed for display.

Regulated Frequency Sweep

Regulated Frequency Sweeps use a sine wave stimulus signal that is moved across a range of frequencies in a specified number of points. At each point, the generator level is adjusted regulated (adjusted to attain a certain level or distortion figure) at each step. See Regulated Frequency Sweep measurements in Chapter 71. This method is often employed to plot an amplifier frequency sweep at the amplifier's maximum output level.

IMD Level Sweep

A stepped mixed twin tone sweep is a IMD stimulus signal that is moved across a range of levels in a series of points or steps. See IMD Level Sweep in Chapter 46.

Nested Sweeps

Bench Mode measurements and some Sequence Mode sweeps provide a Nested Sweep function.

When Sweeps are nested, one defined sweep (driven by a Primary Source, such as Frequency) is run at steps of second defined sweep (driven by a Secondary Source, such as Level). The effect is to nest frequency sweeps (to follow the example here) in a series of level sweeps, providing a family of responses at different levels.

Ordered measurements

Sine moved sequentially across channels

The Crosstalk measurement uses a sine wave stimulus on each of the active output channels, one at a time. See Chapter [crosstalk chapter].

Sine at two sequential levels

The Signal to Noise measurement (chapter 74) first measures the DUT output with a sine wave stimulus, and then makes a second measurement with the generator output level set to zero.

The Dynamic Range (AES17) (chapter 41) measurement first measures the DUT output with a sine wave stimulus at maximum level, and then makes a second measurement with the generator output level set to a very low value, usually 60 dB below the maximum level.

Other batch measurements

Regulated sine

A regulated sine wave stimulus signal is regulated (adjusted to attain a certain level or distortion figure) before the measurement is made. Sequence Mode and Bench Mode both offer regulation in Generator Level setting, and as an option in stepped sweeps.

Digital error rate (bit test}

Bittest Random (also simply called Bit test) is a pseudo-random waveform with values uniformly distributed between plus and minus full scale. Algorithms in the generator and analyzer determine sample values. Bit test exercises a wide range of levels and frequencies and is the most thorough of the bit-accurate waveforms. See Chapter 38.

Generator waveform

In addition to the DSP-generated waveforms, APx has the capability to use custom (or *arbitrary*) waveforms, loaded into the generator from waveform files available to the computer running APx500. This is particularly useful when coded audio files (such as dts, Dolby or others) are required to stimulate a decoder in your DUT. See page 163.

Multitone Analyzer

The Multitone Analyzer uses a special stimulus signal called a *multitone*. Multitone analysis is very fast, and provides a broad range of results. See Chapter 60.

Diagnostic tools

Signal Analyzer

The Signal Analyzer is a general-purpose diagnostic tool, with a time domain view that emulates a multichannel oscilloscope display, and a frequency domain view that is an FFT Spectrum. The Signal analyzer uses a high-resolution FFT technique for both views. FFT length can be set as high as 1048576 points. Triggering, coupling, FFT window and number of averages are selectable.

A version of the Signal Analyzer is available for both Sequence Mode and Bench Mode.

Signal Analyzer tools in APx500 are:

- Scope
- FFT Spectrum
- Amplitude Spectral Density
- Power Spectral Density

Additionally, a special function in Signal Analyzer allows you to view either the embedded audio or the PDM bitstream, when the PDM option module is installed. See page 439.

Measurement Recorder

The Measurement Recorder is a general-purpose diagnostic tool that provides a record of level or THD+N versus elapsed time. The length of the record can be very long (up to one week), providing a means of monitoring the output of a device under test over an extended period of time. The Measurement Recorder also provides a means of recording the audio acquisition to a file.

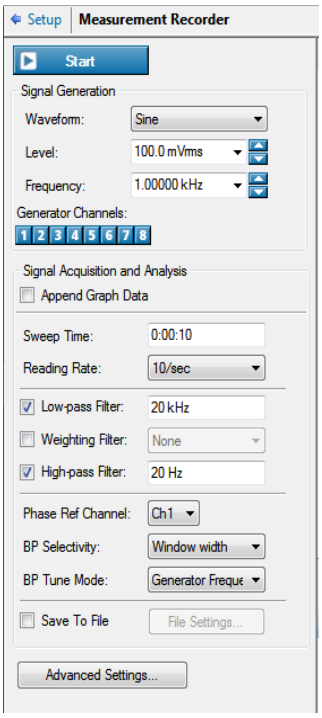
The Measurement Recorder can be used in External Source configuration, but it does not require a specific test signal. It will record any audio signal within the input range of the analyzer.

A version of the Measurement Recorder is available for both Sequence Mode and Bench Mode.

Metadata Recorder

The Metadata Recorder measurement records the state of selected metadata settings in the incoming digital audio signal. If one of the selected settings changes state during the acquisition, the transition instant and the new state are recorded and displayed.

Measurement Settings panel



Each measurement has a measurement settings panel to the right of the Navigator, containing Signal Generation in the upper area, and Signal Analysis controls in the lower area. A range of different controls are available, depending upon system hardware, signal path setup and measurement. This illustration is a typical example.

Signal generation controls are discussed here. Go to page 164 to read about signal acquisition and analysis controls.

Signal Generation controls

Start / Generator ON

Most measurements use a generator stimulus during the measurement process. For the APx500 meter methods, there is a **Generator On** button.

Other methods begin a batch process that returns a result: these measurements use a
Start button.

DC level, Noise and external source measurements do not use a generator stimulus signal. There is no Generator **On** or **Start** button on these measurement panels.

Signal Path Setup

There is a **Signal Path Setup** button that allows you to switch to the Signal Path Setup panel and make changes. A **Back** button on Signal Path Setup allows a quick return.

Waveform

Select the stimulus waveform here. The waveforms available depend upon the analyzer hardware, configuration and the current measurement. See the relevant measurement chapter for more information.

Depending upon the measurement, you may have a Frequency control(s), Sweep settings or other controls available.

Enter the generator output level in the **Level** field.

EQ

For certain sweep measurements, generator equalization allows you to impose a response curve on the stimulus signal to compensate for a non-flat response in the device under test. See Generator Equalization on page 170.

Channel mute switches

The generator output to any channels or channels can be muted (turned OFF) independent of the generator status. A dark blue channel mute button indicates an unmuted channel (channel ON). Muted (channel OFF) channels are indicated by a pale gray button. By default, all channels are unmuted (turned ON).

Generator Waveforms

Test signal generation from a waveform file

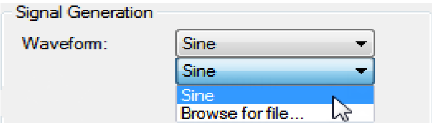
APx500 can use custom generator waveforms (also called arbitrary waveforms) loaded into the generator from waveform files available to the computer running the measurement software. This is particularly useful when coded audio files (dts, Dolby or others) are required to stimulate a decoder in your DUT.

The loaded waveform is "looped" for generator output, so that the end of the waveform is immediately followed by the beginning, providing continuous playback without a jump discontinuity.

Generator waveform files are not restricted to sine waveforms; other waveforms can be used. A number of generator waveform files are available on the Audio Precision Web site at ap.com.

You can also create your own generator waveform files To be useful for measurement, user-created files must be designed with the requirements of the measurement context in mind. Not all audio signals are compatible with certain analyzer views nor are useful for measurement. See page 541 for a table listing the audio file formats supported for generator waveforms.

Using a generator waveform file



A measurement that supports generator waveform files displays a Waveform selection list. The default is **Sine** (generated in DSP). Click the Waveform menu arrow in Signal Generation and choose **Browse for file...** to select an audio file for the generator to use. You can select multiple files.

A selected file becomes the source for that measurement's generator output, and is attached to the project file. Once attached to the project, an audio file is available to all other supported measurements as a selection on the Waveform menu. Files attached to the project but not currently used by any measurement can be removed by navigating to the dialog at **File > Manage Attached Project Items** (see page 24.)

Signal Output

Level adjustment

For analog output, and for digital output of linear waveforms when **Bit Exact** is not set, the level of a generator waveform file can be adjusted in the Generator panel.

Level adjustment is not available for coded waveforms. Coded waveforms are always transmitted bit-exact.

The actual signal level within a generator waveform file is unknown. The file may contain a sine wave at 0 dBFS, or it may contain pink noise at –42 dBFS. This being the case, the generator output is also indeterminate. Level adjustments for generator waveform files are calibrated assuming the maximum possible level in the file.

For analog output, a 1 Vrms level setting would output a 0 dBFS sine wave at 1 Vrms; lower signal levels in the file would be output at correspondingly lower voltages.

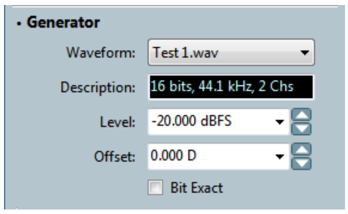
For digital output, a 0 dBFS level setting would output a 0 dBFS sine wave at 0 dBFS; lower signal levels in the file would be output at correspondingly lower digital levels.
A –15 dBFS signal in the file with a –12 dBFS level setting would be output at –27 dBFS.

Dither

For digital output (when **Bit Exact** is not set), a generator waveform file signal is always redithered.

Bit Exact

When a digital output is selected, the Bit Exact checkbox is available when a linear generator waveform is in use. Checking Bit Exact forces the APx generator to output the waveform with no changes whatsoever, with every audio bit exactly as it is in the disk file. Level adjustment is unavailable and dither is off. Coded waveforms are always transmitted bit-exact.



Generator waveform length constraints

Generator waveforms are limited in file length by APx instrument memory resources (see below). A waveform file that exceeds the maximum length will be truncated as it is loaded. A truncated waveform is marked with a warning icon when selected as a generator source.

Truncated waveforms should be avoided, as a jump discontinuity may be introduced on playback at the point of truncation. Best practice is to use the shortest generator waveform file consistent with your measurement requirements.

Memory resources

A generator waveform file is downloaded from the PC into the APx analyzer hardware as needed for a measurement. The hardware memory has a capacity of 128 MB, and audio waveforms that exceed 128 MB will be truncated.

Since APx500 converts 8- and 16-bit PCM waveforms to 32-bit for generator output, and expands coded audio for playback, generator waveform files may use considerably more memory than indicated by their file size on disk.

Maximum File Length table

File formats and sample rates affect the amount of memory used. The table below is provided as a guide, indicating the running times that represent the maximum memory available for various formats.

Generator Waveform Maximum File Length Table

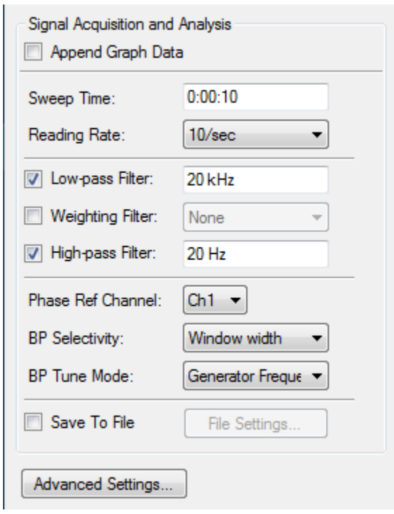
Waveform Type	*Sample Rate*	*Playback Time (sec)*
Mono .WAV	48 kHz	699
Mono .WAV	96 kHz	349
Mono .WAV	192 kHz	174
Stereo .WAV	48 kHz	334
Stereo .WAV	96 kHz	174
Stereo .WAV	192 kHz	87
AC3	48 kHz	699
EC3	192 kHz	174
DTS	48 kHz	699
DTS-HD	192 kHz	43

Supported generator waveform file formats

The table above describes the maximum file length, in seconds, of an audio waveform file that can be transmitted by APx. The playback times are approximate and can vary depending on the audio content, if the file is encoded.

Sample Rate is the rate at which the audio samples are transmitted from APx via the selected output interface. EC3 and DTS-HD are always transmitted over HDMI at 192 kHz regardless of the encoded bit rate of the embedded audio content. AC3 and DTS are always transmitted over the digital interface at 48 kHz, regardless of the choice of digital interface or the encoded bit rate of the embedded audio content.

Analyzer controls



Typical Analyzer controls (Measurement Recorder shown)

Signal acquisition and analysis controls are found in the lower area of the Measurement Settings panel. A range of different controls are available, depending upon system hardware, signal path setup and measurement. This illustration is a typical example.

Signal acquisition and analysis controls are discussed here. Go to page 162 to read about signal generation controls.

Common acquisition and analysis controls

Filtering

Signal Path Setup provides Low-pass filtering (including AC/DC coupling control) High-pass and Weighting filtering for the entire Signal Path. See chapter 91.

Bench Mode Analyzer Controls

Bench Mode provides Settling and some filtering, reference channel and IMD Type controls for the Analyzer.

Sequence Mode Analyzer Controls

Measurements in Sequence Mode often provide additional local filtering, and other controls specific to the measurement.

Append Graph Data

Most batch measurements allow you to append acquisitions. Bench Mode also enables repeating measurements.

Other acquisition and analysis controls

Other controls are available for specific measurements, primarily in Sequence Mode. These include the following:

Ref Channel

For phase measurements, a reference channel can be specified. See chapter 49.

Save to File

The Measurement Recorder and the Multitone Analyzer allow you to save the acquired audio to a file. See chapters 57 and 60.

Sweep Time / Reading Rate

These are adjustments for the Measurement Recorder acquisition. See chapter 57.

Averages

Acoustic Response, Multitone and Signal Analyzer allow you to average a number of acquisitions. See chapters 21, 60 and 73.

Selectivity / Filter Tune Mode

Bandpass Level allows you to adjust and steer the bandpass filter. See chapter 23.

IMD Mode

The IMD measurements allow you to specify which distortion products or combinations of distortion products to view. See chapters 44 and 46.

Trigger

Multitone and Signal Analyzer allow free run or adjustable trigger settings. See chapters 60 and 73.

Measured Channel / Target THD+N

The Maximum Output measurements allow you to specify the regulation target.

Appending and Importing data into Data Sets

Sweep and other batch measurements allow appending and importing of data. Multiple instances of data are managed as Data Sets.

For example, with a new sweep acquisition, the graph is populated with a new set of data for display.

This data can be seen as a curve on the graph or as a table of values in the graph data grid for each view. This data is called Measured Data, and is shown in the Data Sets panel as Measured 1.

Append

Set the Append checkbox in the Measurement Settings Panel to add more measured data sets to the graph data already acquired. Data added in this way is a shown in the Data Sets panel as Measured 2, Measured 3, etc.

Import

You can also use the Import command on the graph data grid panel to bring external data into the graph. Data added in this way is a shown in the Data Sets panel as Imported 1, Imported 2, etc.

Read more about managing Data Sets on page 573.

Advanced Settings

Each Sequence Mode measurement has additional settings available in an Advanced Settings dialog. These vary with the measurement, and are discussed in the individual measurement chapters.

External Source

Some DUTs (disc or other media players, broadcast receivers, transmission links) have no available signal inputs. How is the stimulus signal introduced into the DUT?

The answer is to use the APx External Source configuration, which is an open-loop solution that does not use the instrument generator, but instead provides a test signal that can be transmitted, played or otherwise generated externally.

Not all measurements support the External Source configuration. See Chapter 5 for more information about External Source measurements.

Results

Each measurement has one or more primary results (measured results) associated with it. A primary result is the direct result of a measurement; a primary Level result, for example, is the measured amplitude of the audio signal. See chapter 93.

Result data can be also imported from other measurements, from other applications or from arbitrary lists of values.

Navigating results

Results are represented as branch nodes on the Navigator tree display, and as thumbnails in the Selector filmstrip display. The currently selected result is displayed in the Graph panel.

Primary or derived results can be added to or deleted from a measurement, and can be renamed to suit your needs. Multiple instances of a result can be added to a measurement.

Derived results

A measurement can have one or more user-defined derived results. A derived result is a data point or set of data points derived from an existing result (called the source result) by a mathematical computation such as averaging or normalizing. Derived results can be attached to primary results, imported results or existing derived results. See chapter 95.

Software Options

You can purchase software options to add features to your analyzer. Software options are enabled by one or more iButton keys, which are mounted in the APx Software Option module (pn BSWO.0000). The software option module (when present) is connected to the Software Options connector on the analyzer rear panel.

Adding software options

You can easily add more software options in the field by ordering and installing a factory-keyed iButton.

Viewing installed software options

To see if a software option is installed in your APx analyzer, run APx500 with the analyzer connected. Go to Help > About and scroll to the bottom of the Product Components list. If there is an iButton installed, you will see the text “iButton” followed by an identifying code. Click on an iButton line, and the software options enabled by that button will be shown in the Component Details panel.

Software options available for all APx analyzers

SW-POLQA-2

For Sequence Mode POLQA perceptual audio testing and MOS results, software option SW-POLQA-2 provides

- POLQA
- POLQA Averaged

SW-POLQA-2 provides POLQA operation for only 2 audio channels.

SW-PESQ

For Sequence Mode PESQ perceptual audio testing and MOS results, software option SW-PESQ provides

- PESQ
- PESQ Averaged

SW-SPK-PT

For fast Sequence Mode production testing of loudspeakers and drivers, software option SW-SPK-PT provides

- Acoustic Response (with the Rub and Buzz result)
- Loudspeaker Production Test
- Modulated Noise.

SW-SPK-RD

For Sequence Mode loudspeaker testing in R&D, SW-SPK-RD provides

- Acoustic Response (with the Rub and Buzz result)
- Impedance/Thiele-Small.
- Loudspeaker Production Test and
- Modulated Noise

Also included in SW-SPK-RD is authorization for two utilities:

- APx Polar Plot utility and
- APx Waterfall Graph utility.

These utilities are not part of the APx500 measurement software, but can be downloaded from ap.com.

Software options available for the APx515

Without options, the APx515 enables Sequence Mode and a basic set of APx500 features and measurements. This set can be expanded to the full complement of instrument-appropriate features and measurements by the installation of iButton software option keys.

Basic Measurements in the APx515

These measurements are enabled by an APx515 with no software option keys:

- CMRR
- Crosstalk, Custom
- Crosstalk, One Channel Driven
- Crosstalk, One Channel Undriven
- DC Level
- Frequency
- Frequency Response
- Interchannel Phase
- Level and Gain
- Measurement Recorder
- Noise (Q-Peak)
- Noise (RMS)
- Pass/Fail
- Signal to Noise Ratio
- SINAD
- Stepped Frequency Sweep
- Stepped Level Sweep
- THD+N

SW-BEN Option for the APx515

Software Option SW-BEN enables Bench Mode for the 515.

SW-ACR Option for the APx515

For acoustic testing, software option SW-ACR enables the following measurement:

- Acoustic Response (the Rub and Buzz result is not included with this option).

SW-HST Option for the APx515

For high-speed testing, software option SW-HST enables the following measurements:

- Continuous Sweep
- Multitone Analyzer

SW-AML

Software option SW-AML (the Advanced Measurement Library) enables the following measurements:

- Bandpass Level
- Bandpass Frequency Sweep
- Bandpass Level Sweep
- Crosstalk Sweep, Custom
- Crosstalk Sweep, One Channel Driven
- Crosstalk Sweep, One Channel Undriven
- Digital Error Rate
- Dynamic Range - AES17
- IMD (DFD/MOD/SMPTE/CCIF)
- IMD Frequency Sweep
- IMD Level Sweep
- Maximum Output
- Maximum Output (CEA-2006)
- Metadata Recorder
- Noise Recorder
- Regulated Frequency Sweep
- Signal Analyzer

DC Level Sweep

Acoustic Testing

Acoustic testing brings some additional considerations to test. Preferred units are acoustic, microphone calibration may be necessary, and stimulus EQ may be required to compensate for transducers and provide Leveled Acoustic Output.

Specific acoustic and electro-acoustic measurements such as Acoustic Response, Loudspeaker Production Test and Impedance/Thiele-Small are supported by special Signal Path acoustic features, such as

- Acoustic output configuration
- Acoustic input configuration and
- Output EQ (equalization)

and the generator feature

- Generator EQ.

Although these features certainly have uses beyond acoustic testing (such as preemphasis/deemphasis applications or normalizing a response for flat display) their primary application is in acoustic test.

Acoustic Output Configuration

For convenience when performing acoustic tests, APx500 can be set to Acoustic output configuration. Acoustic output configures all generator level setting fields and generator level graph displays to units of pascals or dBSPL. An acoustic signal path has a scaling factor called Voltage Ratio associated with it, expressed in Pa/V and displayed in the References dialog.

Entering Acoustic output configuration

In Signal Path Setup, place a check in the Acoustic checkbox in Signal Path Setup: Output Configuration.

Acoustic output configuration and the Acoustic output checkbox are only available for analog outputs.

When in Acoustic output configuration, the Auto Gen Level button is available in Signal Path Setup: Output Configuration.

To set the Voltage Ratio, connect the APx output to a power amplifier and a loudspeaker. Face the loudspeaker into a calibrated measurement microphone at a standard distance. Connect the microphone to an APx input. Use Auto Gen Level to determine the generator voltage that achieves the target dBSPL.

Acoustic Input Configuration

For convenience when performing acoustic tests, APx500 can be set to Acoustic input configuration. Acoustic input configures all APx500 measurements in the signal path to display levels relative to channel-specific acoustic reference levels. This is useful for making acoustic measurements with multiple microphones.

Specifically,

- Level results are reported in dBSPL or Pa (in this case the Pa values are rms).

- Gain results are reported in Pa/Vrms.
- Instantaneous voltages are reported in Pa (in this case the Pa values are instantaneous).
- Impulse response is reported in Pa/V (in this case the Pa values are instantaneous).

The Pa and dBSPL values use the Sensitivity input reference(s), set in Set dBSPL Per-channel / Microphone Calibration dialog, and also displayed in the References dialog. See pages 56 and 65 for information about Acoustic input configuration and setting references.

Entering Acoustic input configuration

In Signal Path Setup, set the Acoustic checkbox. Acoustic input configuration and the Acoustic checkbox are only available for analog inputs.

Stimulus Equalization

APx offers two methods to equalize the stimulus signal: Output Equalization, which inserts a filter in the output signal path, and Generator Equalization, which changes the signal level by frequency as it is being generated. Output Equalization is a signal path feature, discussed on page 168. Generator Equalization is a feature of some measurements, discussed on page 170. Also see a Comparison of Generator EQ and Output EQ on page 171.

To EQ and re-save one or more .wav files, download the APx EQ .wav Files Utility at www.ap.com/display/file/661.

Application

An equalized stimulus signal is useful in a number of cases, including:

Leveled Acoustic Output

In testing microphones or systems with embedded microphones (such as a hearing aid or a mobile device), it is essential that frequency response of the power amplifier, the loudspeaker and the acoustic chamber be characterized and compensated for in the stimulus signal, so that the response of the system under test can be accurately measured. This process is called “leveling the acoustic output.”

In APx, this process requires characterizing the system response using a measurement microphone and a measurement such as Frequency Response or Acoustic Response.

The response curve is exported, and then imported into Output EQ and inverted.

This is best performed with Acoustic output and Acoustic input configurations on, and normalizing the level with Auto Gen Level before the characterization sweep and again after the Output EQ is applied.

Preemphasis and deemphasis

Some devices, such as phonographs, analog tape players or analog radio receivers, are designed to have a flat frequency response across a complete signal path, but non-flat frequency response at points within the signal path, with the goal of reducing noise introduced in the storage media or distribution channel. The response of the recording or transmitting device is modified by a preemphasis equalization curve, and the response of the playback or receiving device is modified by a complementary deemphasis equalization curve.

To test DUTs that incorporate a deemphasis equalization response, it is convenient to use a test sweep that varies in amplitude as if it were passed through the required preemphasis filter, producing a flat measurement response result.

Standard preemphasis and deemphasis curves are provided on the APx Resources disc as EQ tables. These include the RIAA phonograph curves and the 50 µs and 75 µs FM broadcast curves.

Characterization of a loudspeaker “golden unit”

In production loudspeaker testing, a common procedure is to characterize a well-performing loudspeaker as a “golden unit,” and to use its response curve as a basis for pass/fail limits. It is convenient to view the response of the golden unit as relatively flat, even if the true response is irregular.

Output EQ

For analog outputs, Output EQ (equalization) allows you to insert a filter in the output signal path, after the APx generator. This enables compensation for a non-flat response in the device under test, as described in Stimulus Equalization on page 168.

An alternative to Output EQ that can accomplish a similar goal for some measurements is to vary the APx generator level by frequency in a sweep or multitone signal. See Generator EQ, next.

Creating or importing an EQ curve

Open the EQ menu in Signal Path Setup: Output Configuration. Choose

- Create New or
- Browse for file.

The filter response curve is specified in the Edit Output EQ Table dialog, and filter response curves can be imported and saved. The response curve can be inverted at the push of a button.

Design

The filter is implemented in the digital domain near the end of the output signal path, before the digital-to-analog converters (DACs). Th e filter has a fixed number of poles (30), and provides a best approximation (the **Design** curve) to the requested response (the **Target** curve). The EQ response beyond the range specified in the Target curve is unspecified.

Application

The filter is only applied to analog outputs.

The filter can be applied to a maximum of 2 channels. If more than two output channels are specified in **Signal Path Setup: Output Configuration**, the filter is made inactive.

The filter is applied to all audio signals generated in DSP, including sine wave, square wave, sweeps, IMD signals, multitone, noise and any non-coded arbitrary generator waveform signal output from files on disc, including POLQA and PESQ stimuli. The filter cannot be applied to the square wave and DIM signals generated in the AG52 option.

Generator output limiting

DAC generator

If the combination of generator level and offset settings and the applied Output EQ curve would require the APx generator to exceed its maximum level, the filter is scaled so that the maximum never exceeds a point 1 dB below the maximum generator output level (which varies with instrument hardware and configuration), while preserving the Target EQ curve shape. A warning is shown in the Status Bar when such a condition exists.

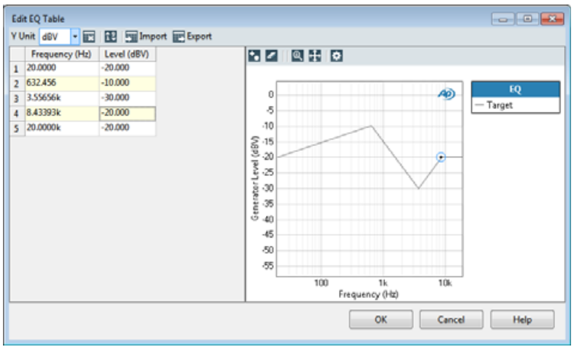
High Performance Sine Generator

If the combination of generator level setting and the applied Output EQ curve requires the APx generator to exceed its maximum level, the APx generator is held at its maximum level until the required level falls below the maximum. A warning is shown in the Status Bar when such a condition exists.

Interaction with Generator EQ

If **Generator EQ** is in effect for a measurement, it overrides **Output EQ**. **Output EQ** is disabled for that measurement.

Editing the Output EQ table



This dialog allows you to import, edit and export Output EQ tables.

The points in the table are represented graphically by the **Target** trace in the graph. The **Design** trace represents the actual filter curve.

Add Point

Add Point inserts a blank frequency point (row) below the current selection in the table. Enter a level value for each new point. You can edit the frequency or level value arbitrarily.

Delete Selected Points

Delete Selected Points removes the selected points (rows) from the table.

Invert

Use this control to invert the EQ curve. This is useful for creating a compensation curve from a measured response curve.

Zoom

Click on this button and then drag a marquee in the graph to zoom.

Reset Zoom/Pan

Click on this button to revert to the original zoom and pan views.

Graph Properties

This button opens a **Graph Properties** dialog, where you can select ranges, names and log/lin characteristics for the EQ curve graph.

Clear Table

Clear Table removes all points (rows) from the table.

Sort Ascending

Sort Ascending sorts the table by frequency, ascending toward the bottom of the table.

Import

Import opens a file browser window to select and open compatible EQ table files from disk. Excel, CSV and Audio Precision legacy *.adx and *.atsx are supported.

Export

Export opens a file browser window to save the current EQ table as an Excel or CSV file.

Y Unit

Choose the Y-axis (level) unit here.

Automation

You can import (and optionally invert) an EQ curve into **Signal Path Setup** during an automated sequence. See **Adding an Import Output EQ Curve Step** on page 489.

Generator EQ

Generator EQ (equalization) allows you to vary the APx generator level by frequency in a sweep or multitone signal, in compliance with a curve defined in an EQ table. This enables compensation for a non-flat response in the device under test, as described in Stimulus Equalization on page 168.

> *An alternative to Generator EQ that can accomplish a similar goal (for analog outputs) is to insert a filter in the output signal path. See Output EQ, previous.*

Measurements that can use Generator EQ

The Generator EQ feature addresses this requirement by allowing you to specify and attach an equalization table to the APx sine sweep or multitone generator for the following measurements:

- Acoustic Response
- Continuous Sweep
- Frequency Response
- Acoustic Response
- Stepped Frequency Sweep
- Multitone (see the next paragraph)

Operational details can be found in the chapters for Acoustic Response, Continuous Sweep, Frequency Response, Acoustic Response and Stepped Frequency Sweep.

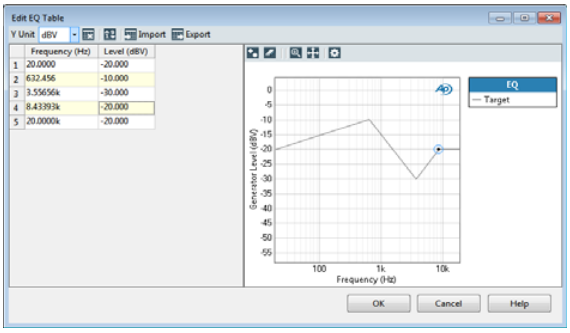
Multitone signal equalization

Since the multitone measurement does not use a swept sine signal, generator equalization cannot be applied in the same way. You can, however, effect the same functionality for multitone measurements. For multitone, the equalization is applied by adjusting the levels of individual tones during the multitone waveform creation process. See page 378.

Applying Generator EQ

For instructions for using a generator equalization table with specific measurements, see the EQ heading in the Common Controls section for Acoustic Response, Continuous Sweep, Frequency Response, Acoustic Response or Stepped Frequency Sweep. For Multitone, Import EQ Table on page 378.

Editing the Generator EQ table



This dialog allows you to import, edit and export generator equalization tables.

Add Point

Add Point inserts a blank frequency point (row) below the current selection in the table. Enter a level value for each new point. You can edit the frequency or level value arbitrarily.

Delete Selected Points

Delete Selected Points removes the selected points (rows) from the table.

Invert

Use this control to invert the EQ curve. This is useful for creating a compensation curve from a measured response curve.

Zoom

Click on this button and then drag a marquee in the graph to zoom.

Reset Zoom/Pan

Click on this button to revert to the original zoom and pan views.

Graph Properties

This button opens a Graph Properties dialog, where you can select ranges, names and log/lin characteristics for the EQ curve graph.

Clear Table

Clear Table removes all points (rows) from the table.

Sort Ascending

Sort Ascending sorts the table by frequency, ascending toward the bottom of the table.

Import

Import opens a file browser window to select and open compatible EQ table files from disk. Excel, CSV and Audio Precision legacy *.adx and *.atsx are supported.

Export

Export opens a file browser window to save the current EQ table as an Excel or CSV file.

Y Unit

Choose the Y-axis (level) unit here.

Comparison of Generator EQ to Output EQ.

Output Equalization	*Generator Equalization*
analog output only	any analog or digital output
any signal except AG52 special signals (square and DIM). This includes noise, perceptual audio, arbitrary (generator) waveforms, unswept signals, split sine, etc.	Stepped Frequency Sweep, Acoustic Response, Frequency Response, Continuous Sweep, Acoustic Response and Multitone measurements only.
two channels maximum	all channels
applied per signal path	applied per measurement

21

Acoustic Response Measurements

Introduction

For the APx515, this measurement requires a software option key. See page 166 for more information about software options.

The Acoustic Response measurement is designed to facilitate loudspeaker testing, using an APx500 Series audio analyzer with a measurement microphone and an audio power amplifier.

Typically, the APx500 generator output is connected to the power amplifier, which in turn is connected to the loudspeaker under test. A measurement microphone is located in the acoustic field of the loudspeaker, and the output of the microphone preamplifier is connected to an APx500 analyzer input. A specially configured continuous sweep measurement (called the Acoustic Response measurement) is made. See page 220 for more about continuous sweep.

Acoustic Response results available in APx500 are:

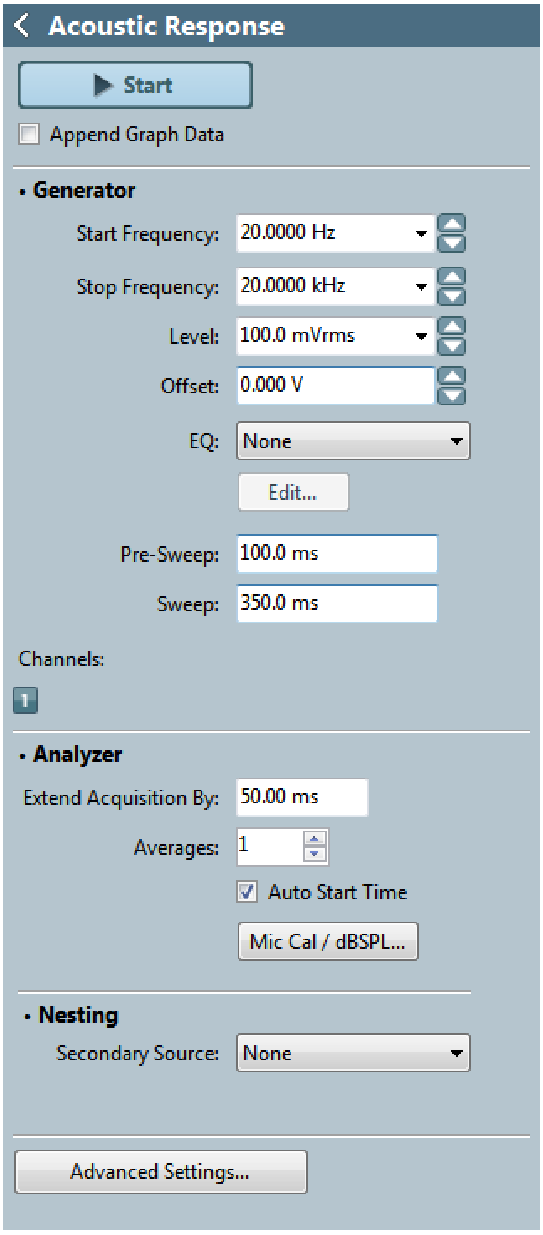
- Energy Time Curve (ETC)
- Delay
- THD Level
- Impulse Response
- Phase
- Distortion Product Ratio
- Level (Freq. Response)
- Group Delay
- Distortion Product Level
- Relative Level
- Level and Distortion
- Rub and Buzz *
- Deviation
- THD Ratio
- Acquired Waveform

All of these results are available from a single acquisition.

** The Rub and Buzz result requires a software option key. See page 166 for more information about software options.*

Loudspeaker Testing using Acoustic Response

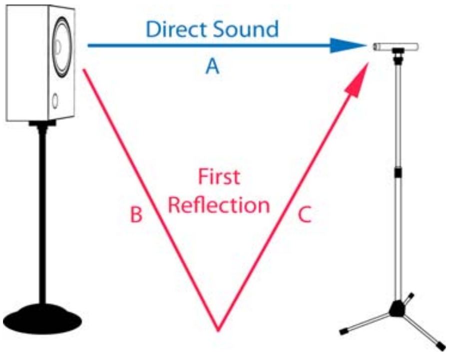
Acoustic Response needs careful setup and precision third-party instruments to obtain good loudspeaker testing results. You must have a measurement micro-



phone and its associated power supply and preamplifier, a microphone calibrator, a suitable acoustic space, and in most cases a high-quality audio power amplifier.

This first section takes you through the basics of setting up a loudspeaker testing system, using Acoustic Response result tools such as Energy Time Curve and Impulse Response to optimize the APx processing for best measurements.

Positioning the Loudspeaker and Measurement Microphone in the Acoustic Space



Loudspeaker measurements can be made in various acoustic spaces: outdoors, within rooms, or even within special isolation enclosures. The choice of acoustic space will affect the optimal positioning of the loudspeaker and microphone, and will affect the measurement results. This is because it is desirable to measure only the sound traveling directly from the loudspeaker to the microphone, and to avoid measuring sound that has traveled a reflected path, from the loudspeaker to a reflecting surface and then to the microphone.

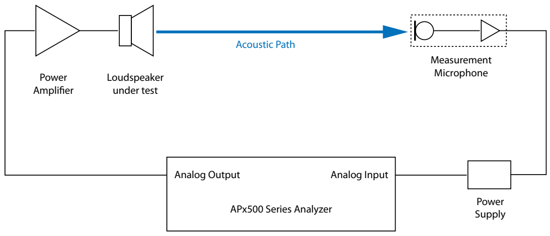
Close reflections are more troublesome than distant reflections, because they typically have greater amplitude and because they may reach the microphone before the impulse response of the system has decayed.

The best measurement results, then, are typically made in acoustic spaces that create minimal close reflections: outdoors, or in large rooms. Floors, walls, ceilings, tables and other objects reflect sound waves, and even in a large room care must be taken to mount the loudspeaker and microphone at a distance from any reflecting surface.

Sound travels at approximately 343 meters per second (1125 feet per second) in air at 20 °C (68 °F); this speed varies primarily with air temperature and humidity, and to a minor extent with pressure and density. This means that in ordinary conditions a sound wave takes approximately 3 ms to travel 1 m, or approximately 0.9 ms to travel 1 ft.

A standard distance between the loudspeaker under test and the measurement microphone is 1 m. Other distances can be used. Ideally, the path of first reflection is longer (in time) than the system impulse response.

Connecting the Loudspeaker and Measurement Microphone



Configuring APx500 for Acoustic Response measurements

After launching APx500, go to Signal Path Setup to configure the analog outputs and inputs for proper interface to the audio power amplifier and the measurement microphone.

Reduce the number of output and input channels to the number actually required for your test. This improves both system performance and clarity of display.

For best performance, reduce the system bandwidth to 40 kHz or even 20 kHz. Few loudspeakers have ultrasonic response, and the APx500 software runs more efficiently at reduced bandwidth.

Verify system electrical and acoustic paths

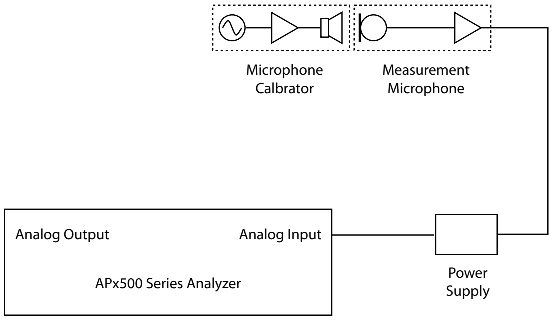
In Verify Current Connections, be sure that the generator level is not set too high, and that the generator frequency is appropriate for your loudspeaker. If the audio power amplifier has high gain or the loudspeaker is very sensitive, it may be possible to damage the loudspeaker, even at the default generator setting of 100 mVrms.

Click the Generator **ON** and verify that the loudspeaker and microphone are connected and operating properly.

Calibrating the Measurement Microphone and APx500

It is not required that loudspeaker measurements be made in reference to an acoustic calibrator, but it is often done. APx500 can use an acoustic calibration as a dBSPL reference and report measurement results against that reference.

You must have an acoustic calibrator such as the MMC-3 available from Audio Precision to calibrate the system.



Two calibration levels are in standard use for acoustic calibration: 94 dB SPL and 114 dB SPL. If both are available on your calibrator, select the level most suited to your measurement. Mount the calibrator firmly on the microphone capsule and turn it ON.

In APx500, go to Signal Path Setup > References > Input References and click Mic Cal / Set dBSPL... In the Calibrator Level field, enter the calibrator setting (94 dB SPL or 114 dB SPL, set in the previous step. Then select the input channel (usually Channel 1) and click Set 1 to enter that level as dBSPL1. See Mic Cal / Set dBSPL on page 64.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 6 for information about Signal Path Setup.

Append Graph Data

Measured data are grouped in a Data Set, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the Append Graph Data box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 573.

Generator

The Acoustic Response measurement uses a log-swept sine chirp stimulus signal, swept between the frequencies entered in Start Frequency and Stop Frequency, at the value set in Level.

Note: Loudspeakers under test can be damaged by signals that exceed their level or frequency limits. Be sure to set generator and amplifier level and sweep frequencies to values within the limits of the loudspeaker.

Running the Measurement

To use Acoustic Response, click Start. The generator will output the test signal to the DUT on the selected generator channels. Read the results in the selected result view.

Start Frequency and Stop Frequency

These controls set the frequency range of the sweep. See page 220 for more about continuous sweep ranges, which are dependent upon the instrument.

Level

Set the generator level here.

Offset

Set any DC offset to the generator signal here.

EQ

You can equalize the generator signal to make a pre-emphasized stimulus. See Generator Equalization (page 170) for a general discussion of this feature.

Choosing generator equalization

Select None, Relative or Absolute from the EQ drop-down menu.

- None applies no equalization to the generator.
- Relative applies the gain or loss specified in the EQ table to the current generator level for each channel.
- Absolute sets the generator level for all channels to the level specified in the EQ table. Note: when Absolute is selected, current channel level settings are lost.

Edit

If you have chosen to equalize the generator signal, you can edit the EQ table here. See Edit EQ Table (page 170).

Pre-Sweep and Sweep duration fields

A continuous sweep stimulus consists of a sine wave swept logarithmically across the sweep range in the period of time set in these fields.

To enable the DUT to stabilize before the beginning of the defined sweep measurement, there is variable Pre-Sweep time. The actual sweep stimulus signal begins below the defined Start Frequency and is swept upward, reaching the Start Frequency at the end of the Pre-Sweep time. The sum of the times in Pre-Sweep and Sweep is the total sweep length.

- The default for **Pre-Sweep** is 100.0 ms. Minimum is 0 s; maximum is 1.0 s.
- The default for **Sweep** is 350.0 ms. Minimum is 50.0 ms; maximum is 5 s.

Longer **Pre-Sweep** settings allow more time for DUT stabilization, but increase measurement time. Longer **Sweep** settings provide greater resolution and signal-to-noise ratio, but increase measurement time.

Analyzer

Extend Acquisition By

By default, a continuous sweep measurement makes an acquisition slightly longer than the stimulus, to include possible time-delayed artifacts created in the DUT. By default, the acquisition is extended 50 ms longer than the stimulus.

In some cases, you may want to extend the acquisition further. Enter a new value in the **Extend Acquisition By:** field. Minimum extension is 0 s; maximum is 3 s.

Averages

Averaging several sweeps can remove uncorrelated data from the results (for acoustic measurements, this is usually background noise). The sweep will be run the number of times set in the **Averages** field, and the results will be averaged for display.

Auto Start Time

When **Auto Start Time** is checked, the **Time Window Start** adjustment control is not available, and **Time Window Start** is automatically set to the generator **Start** time. **Time Window End** is initially set just after the decay of the impulse, but can be adjusted. This is the default behavior and is correct for the vast majority of acoustic tests. For more detail and optional manual control of **Time Window Start**, see Adjusting the Time Window following on this page, and Deselecting Auto Start Time on page 177.

Mic Cal / dBSPL...

This button opens the **Microphone Calibration / Set dBSPL Per-channel** dialog, also available from the Signal Path Setup > Input Configuration and References panels when in Acoustic Mode.

Nesting

Acoustic Response sweeps can be nested. Read about Nested Sweeps beginning on page 161.

Secondary Source (nested sweeps)

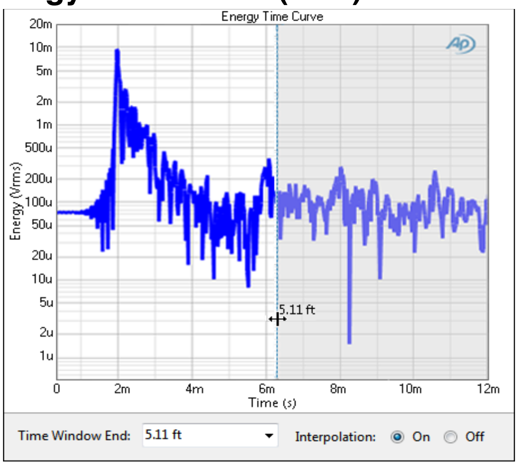
Select the secondary sweep source here. This is the outside, nesting parameter that is varied to change the conditions for the primary sweep through a number of iterations.

Available choices are dependent upon instrument hardware, options, Input/Output settings and Bench Mode Generator settings.

Advanced Settings

Advanced Settings include output Level settings and tracking. To view or modify these settings, click **Advanced Settings**. See Advanced Settings for Continuous Sweep on page 223.

Energy Time Curve (ETC)



The Energy Time Curve (ETC) provides a graphical display of the ETC of the loudspeaker and acoustic space. This view is useful in setting up an acoustic measurement. See More About the Energy Time Curve on page 185.

Adjusting the Time Window

The ETC and Impulse response views are useful in adjusting the Time Window to optimize the measurement results.

Time Window End

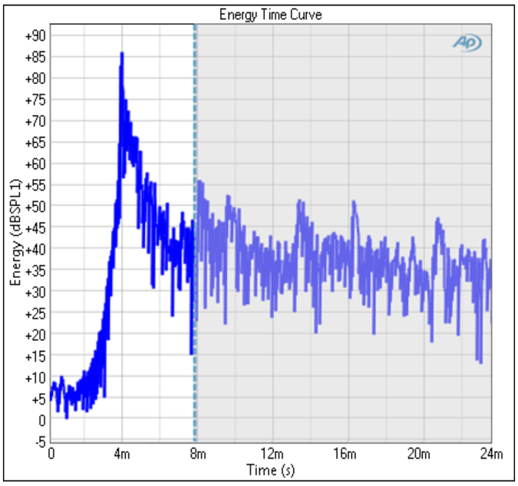
The **Time Window End** control, which can be set in either the Energy Time Curve or the Impulse Response view, selects the portion of the acquired data that is used for analysis. For loudspeaker measurement, it is important to measure only the sound from the loudspeaker, excluding reflections from the acoustic environment.

By default, **Auto Start Time** is checked, causing the beginning of the Time Window to correspond to the generator **Start** time. In most cases, the end of the Time Window should be set just before the first reflection) by entering a value in the **Time Window End** field, or by sliding the Time Window cursor on the graph.

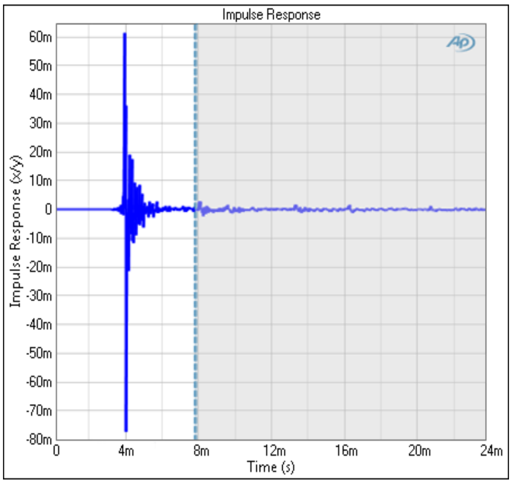
Let’s look at how to use it on the ETC view.

Set up an Acoustic Response measurement as described in the Introduction. Click **Start**.

The ETC result will look something like this. Move the Time Window cursor to a point just before the first reflection, as shown:



Confirm this by viewing the Impulse Response, which will look like this:



Further confirm by measuring the lengths of the acoustic paths, and obtaining the difference between the direct path and the first reflection path (B+C–A, as shown in the diagram on page 174). That distance should agree closely with the position of the Time Window cursor, when viewed in distance units.

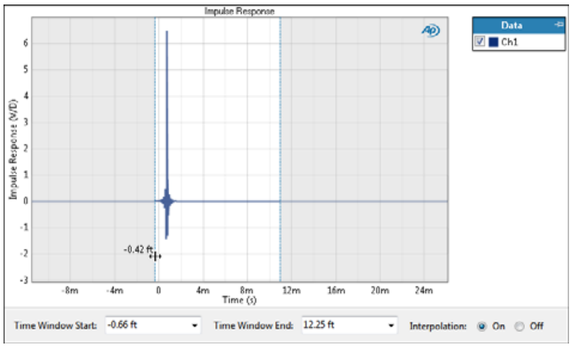
Deselecting Auto Start Time

Time zero (t0) is always set to the start time of the APx generator (the instant that the first audio sample is output when **Start** is pressed). An additional 50 ms is added to the acquisition before t0 to include any activity within the APx system, such as internal ADC/DAC ringing. (This is required for correct processing of the continuous sweep acquisition).

In Loopback, this places the peak of the impulse response at t0. In real testing, device and acoustic delay will delay the acquired signal, placing the Impulse Response later on the graph. The arrival time of the impulse shown on the IR graph is the same as the value displayed in the **Delay** result.

When checked, **Auto Start Time** sets **Time Window Start** at –50 ms, and **Time Window End** just after the decay of the impulse. You can move **Time Window End** to some other point if desired. This is the default behavior and is correct for the vast majority of acoustic tests.

In some circumstances, however, acoustic reflections or ambient noise can cause the **Time Window** span to be chosen in error (typically, in situations where the amplitude of the first reflection is larger at the microphone than the amplitude of the initial signal). In such a case, deselect **Auto Start Time**. A second gray area will be displayed on the left of the **Energy Time Curve** and **Impulse Response** graphs. You will now be able to adjust both **Time Window Start** and **Time Window End** (by dragging with the mouse cursor or by entering a new value into the **Time Window** fields) to include the desired portion of the signal for analysis.



Note: The Auto Start Time checkbox and Time Window Start feature are not available for the acoustic response component of the Loudspeaker Production Test measurement.

Gray areas in other result graphs

The adjustment of the Time Window affects the reliability of measurement results that depend upon the Impulse Response.

In these measurement views, you will notice that the graphs show a gray mask overlaying a portion of the low frequency results. This indicates that the measurement results in the masked area are unreliable. The masked area will change as the Time Window is adjusted.

For a first acquisition, when there is only one data set in the Acoustic Response results, the Time Window can be adjusted and readjusted at will. When other data sets are appended or imported, the position of the Time Window becomes fixed.

In summary...

In APx500, click **Add Measurement** in the Navigator and select **Acoustic Response**.

Set both the **Start** and **Stop** frequencies and the generator Level to prevent damage to your loudspeaker.

Click **Start**. In the **Energy Time Curve (ETC)** view, move the Time Window cursor to exclude the first reflection. Confirm that this is indeed the first reflection by comparing with measurement, if necessary, and by viewing impulse response, if necessary.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Time Window Start

This control is only available when **Auto Start Time** is unchecked. You can enter a **Time Window Start** value here, or read the value obtained by dragging the **Time Window Start** cursor. Read more about **Auto Start Time** and **Time Window Start** in Adjusting the Time Window beginning on page 176.

Time Window End

You can enter a **Time Window End** value here, or read the value obtained by dragging the **Time Window End** cursor. Read more about **Time Window End** in Adjusting the Time Window beginning on page 176.

Units for the **Time Window** controls include seconds (s), feet (ft) and meters (m).

Interpolation

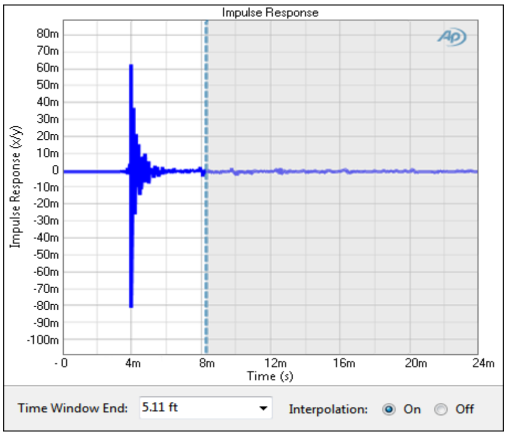
For time-domain displays, the APx500 graphs are normally plotted using sinc function interpolation. This is the default setting. However, digital domain signals are sometimes best understood when viewing actual samples, with no interpolation. Turn display interpolation **ON** or **OFF** here.

Units

Units available for Energy Time Curve are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• s	• Vrms	• dBFS
	• dBV	• FS
	• dBu	• %FS
	• dBrA	• dBrA
	• dBrB	• dBrB
	• dBSPL1	• dBSPL1
	• dBSPL2	• dBSPL2
	• dBm	
	• W (watts)	

Impulse Response



The Impulse Response provides a graphical display of the IR of the loudspeaker and acoustic space. This view is useful in setting up an acoustic measurement in conjunction with the Energy Time Curve view, using the Time Window. For more information about impulse response, see page 218.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Time Window Start

This control is only available when **Auto Start Time** is unchecked. You can enter a **Time Window Start** value here, or read the value obtained by dragging the **Time Window Start** cursor. Read more about **Auto Start Time** and **Time Window Start** in Adjusting the Time Window beginning on page 176.

Time Window End

You can enter a **Time Window End** value here, or read the value obtained by dragging the **Time Window End** cursor. Read more about **Time Window End** in Adjusting the Time Window beginning on page 176.

Units for the **Time Window** controls include seconds (s), feet (ft) and meters (m).

Interpolation

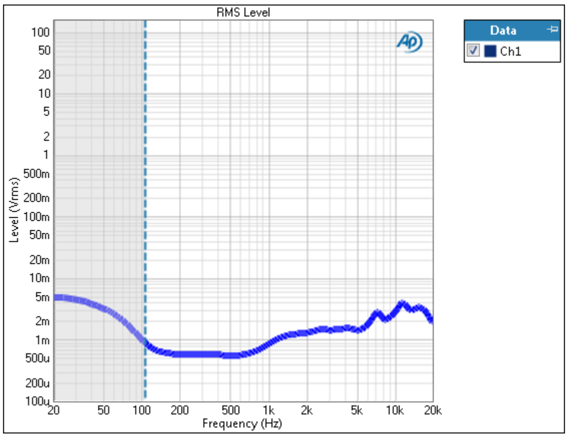
For time-domain displays, the APx500 graphs are normally plotted using sinc function interpolation. This is the default setting. However, digital domain signals are sometimes best understood when viewing actual samples, with no interpolation. Turn display interpolation ON or OFF here.

Units

Units available for Impulse Response are:

X-axis	*Y-axis same-domain*	*Y-axis cross-domain*
• s (seconds)	• x/y	• D/V — or— • V/D

Level (Frequency Response)



The Level result provides a graphical display of the frequency response of the loudspeaker.

Result Settings

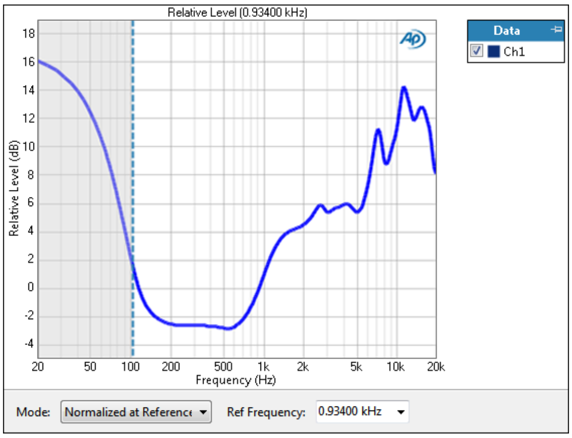
These settings are made in the Result Settings bar, beneath the graph display.

Units

Units available for Acoustic Response Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz	• Vrms	• dBFS
• dHz	• dBV	• FS
• F/R	• dBu	• %FS
• %Hz	• dBrA	• dBrA
	• dBrB	• dBrB
	• dBSPL1	• dBSPL1
	• dBSPL2	• dBSPL2
	• dBm	
	• W (watts)	

Relative Level



The Relative Level result is a continuous sweep measurement that provides a graphical display of the frequency response of each channel. In this result the DUT output level is plotted against frequency, relative to the level at a selected frequency set in the **Ref Frequency** field.

This enables you to specify the frequency that will be set as 0 dB and view the response in relation to that frequency. A midband frequency such as 1 kHz is often set as the reference frequency, and is the default here.

Mode

Normalized at Reference

Enter a reference frequency for normalization in the **Ref Frequency** field. The measured level at the frequency you choose is set as 0 dB on the graph, and the DUT response is plotted in relation to this reference. You can change the **Ref Frequency** at any time (except after appending) and the graph will immediately be redrawn to reflect the new setting.

Auto-Centered in Limits

This control is only usable if upper and lower limits are set on this graph. In some applications, the general conformance of a trace to the shape of a set of limits is of importance, with absolute level less important or irrelevant. **Auto-Centered in Limits** centers the data trace relative to the upper and lower limits. To exclude possible exceptional data points at frequency extremes, the frequency range that is considered for the centering operation is constrained by the values in the **Min** and **Max** fields.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Ref Frequency

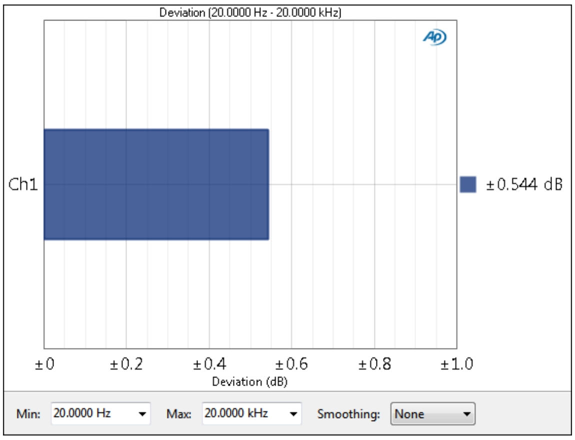
Choose a Reference Frequency here.

Units

Units available for Acoustic Response Relative Level are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Deviation



The Acoustic Response Deviation result is a single value measurement computed from the continuous sweep acquisition. In this result the frequency deviation (the total range of frequency variation) of each channel is displayed as a meter bar. You can specify a minimum and maximum frequency to define the range be considered in the deviation measurement.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Min

Set the minimum frequency of the range of interest here.

Max

Set the maximum frequency of the range of interest here.

Smoothing

Octave smoothing is a common technique in loudspeaker response measurement, useful in revealing trends by smoothing out anomalies in the response curve. The APx500 implementation uses a hybrid FFT bin averaging and interpolation technique to achieve smooth results even at very low bin densities.

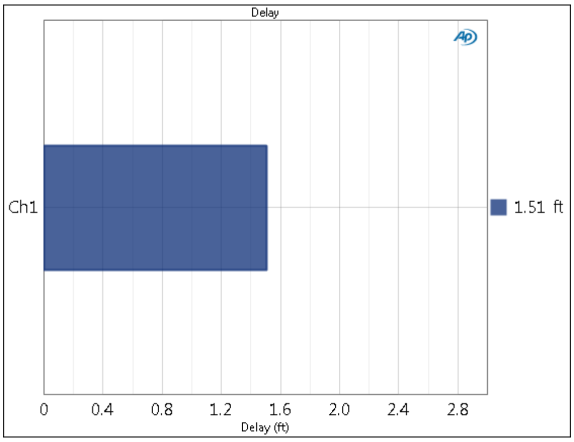
Smoothing effectively passes the raw response data through a constant-Q bandpass filter that is swept with the continuous sweep signal. The bandwidth of this filter is selected in the **Smoothing** field.

Units

Units available for Acoustic Response Deviation are

- dB

Delay



The Acoustic Response Delay result is a single value measurement computed from the impulse response. Delay displays the time delay between the generator output and analyzer input for each channel. For Acoustic Response testing, this is effectively the acoustic delay between the loudspeaker and the measurement microphone.

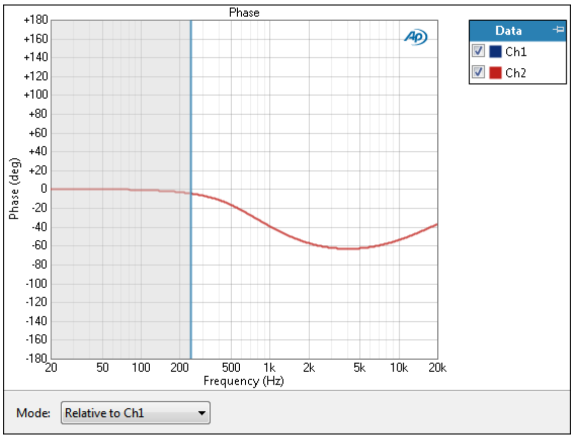
This result is primarily useful for setting up an Acoustic Response test, in conjunction with the ETC view and the Impulse Response view.

Units

Units available for Acoustic Response Delay are

- dB

Phase



The Acoustic Response Phase result is a continuous sweep measurement that displays the DUT output phase for each channel, plotted against frequency.

See page 300 for more information about phase measurements.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Mode

Acoustic Response Phase has four result modes:

Relative to Ch1

In this mode, the absolute phase of channel one is subtracted from the absolute phase of channels numbered greater than 1. The result is plotted against frequency for each channel numbered 2 and above, showing the phase difference (from channel 1) for each channel. Since channel 1 is used as the reference, it is not plotted in this result. This mode shows "unwrapped" phase differences.

Input-to-output

In the input-to-output mode, the absolute phase of each channel, from device input to device output, is plotted against frequency, "unwrapped." Input-to-output mode includes device delay.

Input-to-output, wrapped

This mode shows the same result as input-to-output phase, but "wrapped" within the range of –180° to +180°.

Input-to-output, excess

This mode shows the input-to-output phase (unwrapped), but removes the linear component (the average group delay of the system), leaving the "excess phase."

Some devices have linear phase only over a portion of their passband. For such devices, including frequency ranges where the phase is non-linear affects the excess phase result, which becomes dependent on the stop frequency of the measurement. Although the measurement is technically correct, the variation of the result with stop frequency is disconcerting and not particularly useful.

You may (optionally) use the **Min:** and **Max:** settings to set the expected linear phase range of the device under test. For this result, the device delay is computed as the best linear fit to the phase data between the **Min** frequency and the **Max** frequency. Excess phase is then computed as the difference from the linear fit.

Min and **Max** frequencies cannot extend beyond sweep **Start** and **Stop** frequencies.

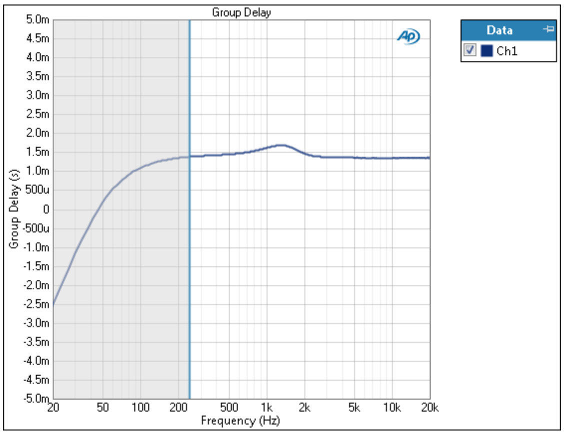
See page 215 for more information about wrapped and unwrapped phase.

Units

Units available for Acoustic Response Phase are:

X-axis	*Y-axis*
• Hz	• deg
• dHz	• rad
• F/R	
• %Hz	

Group Delay



Group delay is a measure of the rate of change of phase shift as a function of frequency. Group delay can also be described as a time delay of a group of frequencies with respect to the generator.

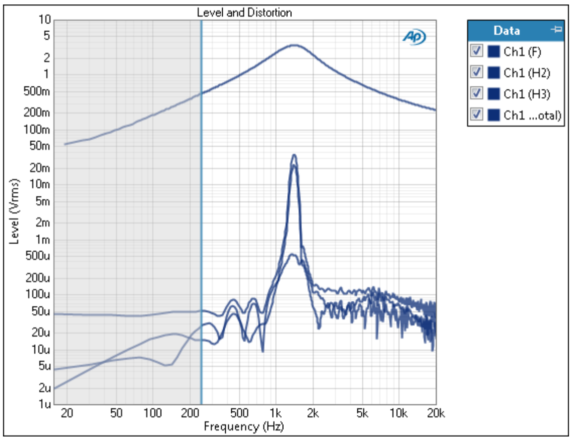
The Group Delay result is a continuous sweep measurement that displays the group delay measured in each channel, plotted against frequency.

Units

Units available for Acoustic Response Group Delay are:

X-axis	*Y-axis*
• Hz	• s (seconds)
• dHz	
• F/R	
• %Hz	

Level and Distortion



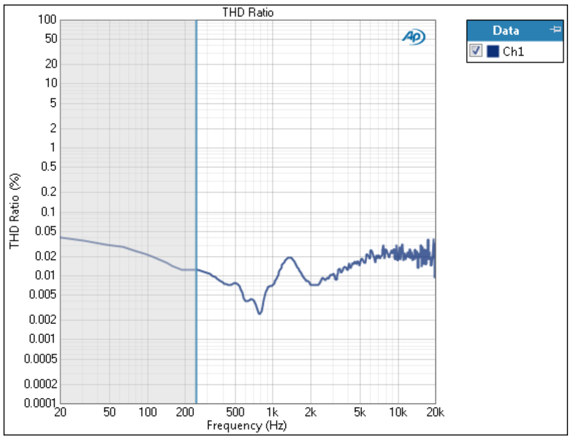
The Acoustic Response: Level and Distortion result is a continuous sweep measurement that provides a graphical display of the fundamental level vs. frequency and the distortion level vs. frequency of the loudspeaker connected to each channel, plotted on the same graph. Distortion curves included total distortion level, H2 distortion level and H3 distortion level.

Units

Units available for Acoustic Response Level and Distortion are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz	• Vrms	• dBFS
• dHz	• dBV	• FS
• F/R	• dBu	• %FS
• %Hz	• dBrA	• dBrA
	• dBrB	• dBrB
	• dBSPL1	• dBSPL1
	• dBSPL2	• dBSPL2
	• dBm	
	• W (watts)	

THD Ratio



The THD Ratio result is a continuous sweep measurement that provides a graphical display of the harmonic distortion response of each channel. In this result the ratio of the level of the THD (total harmonic distortion) to the total signal in the DUT output is plotted against frequency.

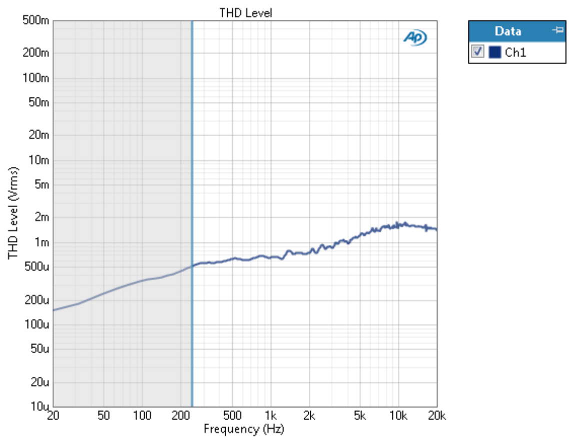
See page 216 for more information about THD.

Units

Units available for Acoustic Response THD Ratio are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

THD Level



The THD Level result is a continuous sweep measurement that provides a graphical display of the harmonic distortion response of each channel. In this result the level of the THD (total harmonic distortion) in the DUT output is plotted against frequency.

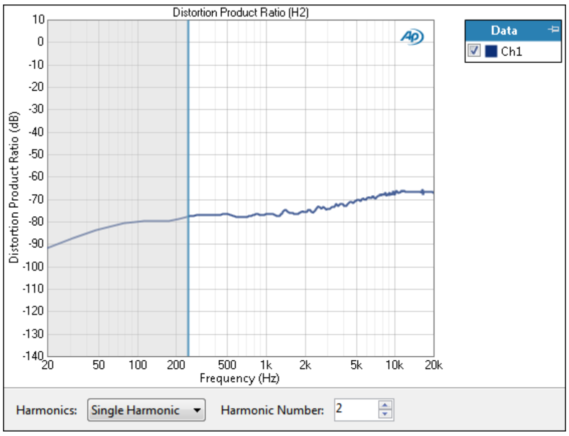
See page 216 for more information about THD.

Units

Units available for Acoustic Response THD Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz	• Vrms	• dBFS
• dHz	• dBV	• FS
• F/R	• dBu	• %FS
• %Hz	• dBrA	• dBrA
	• dBrB	• dBrB
	• dBSPL1	• dBSPL1
	• dBSPL2	• dBSPL2
	• dBm	
	• W (watts)	

Distortion Product Ratio



The Acoustic Response Distortion Product Ratio result is a continuous sweep measurement that provides a graphical display of the selected harmonic distortion products present in each channel. In this result the ratio of the level of the selected harmonic distortion product to the fundamental in the DUT output is plotted against frequency.

Additional Controls for Distortion Product Ratio

Harmonics

For a graph of the ratio of one specific harmonic product to the signal level, select **Single Harmonic** and set the **Harmonic Number** to your choice, from 2nd harmonic to 20th harmonic.

For a graph of the sum of the levels of any combination of harmonic products (from **H2** through **H20**), divided by the signal level, select **Sum of Harmonics**. Then click **Select Harmonics...** to open a result definition dialog.

A note about the gray area on the graph

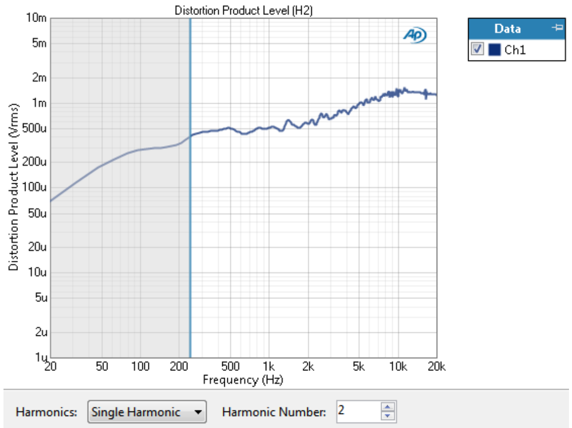
An inevitable consequence of the finite length of an impulse response is that in the frequency domain, the data below some low frequency will be unreliable. In this graph, the area of unreliable data is indicated by gray shading. As the Time Window is increased, the frequency of the bound defining this area is decreased.

Units

Units available for Acoustic Response Distortion Product Ratio are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Distortion Product Level



The Acoustic Response Distortion Product Level result is a continuous sweep measurement that provides a graphical display of the selected harmonic distortion products present in each channel. In this result the level of the selected harmonic distortion product in the DUT output is plotted against frequency.

Additional Controls for Distortion Product Level

Harmonics

For a graph of the level of one specific harmonic product, select **Single Harmonic** and set the **Harmonic Number** to your choice, from 2nd harmonic to 20th harmonic.

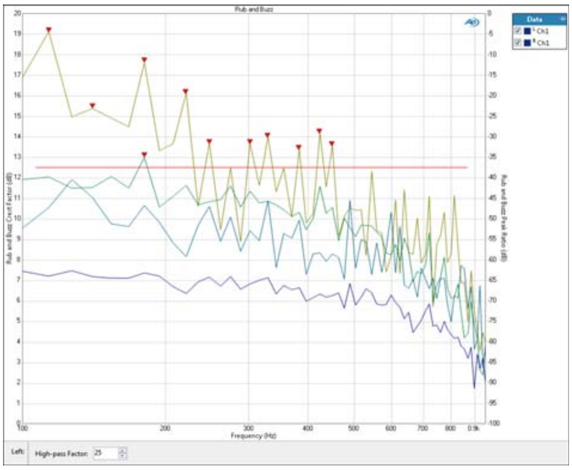
For a graph of the level of the sum of several or all harmonic products (from **H2** through **H20**), select **Sum of Harmonics**, then click **Select Harmonics...** to open a result definition dialog.

Units

Units available for Acoustic Response Distortion Product Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz	• Vrms	• dBFS
• dHz	• dBV	• FS
• F/R	• dBu	• %FS
• %Hz	• dBrA	• dBrA
	• dBrB	• dBrB
	• dBSPL1	• dBSPL1
	• dBSPL2	• dBSPL2
	• dBm	
	• W (watts)	

Rub and Buzz



The Rub and Buzz result requires a software option key. See page 166 for more information about software options.

The Acoustic Response: Rub and Buzz result uses a continuous sweep measurement to display the Rub and Buzz Crest Factor and Peak Ratio for each channel. See More about Loudspeaker Testing on page 185 for a more detailed discussion of using rub and buzz and modulated noise in loudspeaker testing.

Rub and Buzz presents two results on one dual axis graph. In the APx implementation, detection of rub and buzz requires a simultaneous interpretation of both of these results; a rub and buzz crest factor result is plotted in reference to the left Y-axis, and a rub and buzz peak ratio result is plotted in reference to the right Y-axis. Limit behavior is different on a graph with dual axes. See Drawing limits for XY Graphs (dual axis) on page 579 and Editing limits for XY Graphs (dual axis) on page 581.

You can split the dual axis graph into two individual graphs. Right-click on the Rub and Buzz result and choose Split from the context menu. Two new results will be added, presenting the data for the Left and Right axes individually.

Additional Controls for Distortion Product Level

High-pass Factor

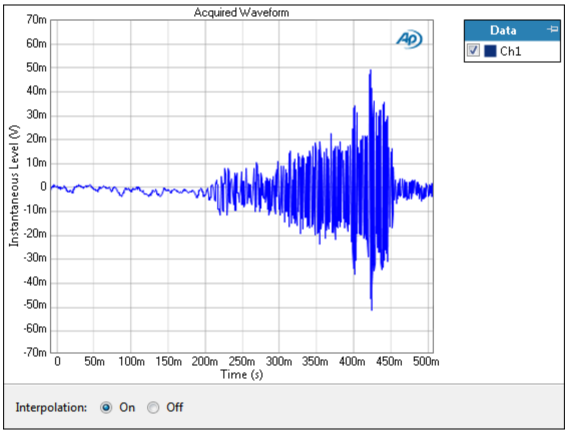
The data plotted on these graphs is high-pass filtered to reveal the rub and buzz signature results. The filter tracks the chirp stimulus frequency, moving upward with the sweep. The corner frequency of the filter is the product of the instantaneous chirp frequency multiplied by the **High-pass Factor**, which can be set in the range from 10 to 30. This results in a graph where the X-axis represents the stimulus frequency, and the plotted data represent the much higher frequency sounds generated by the interaction of the stimulus frequency with loose particles or rubbing components within the loudspeaker system. For interpretation of the results, see More About Loudspeaker Testing on page 185.

Units

Units available for Rub and Buzz are:

X-axis	*Y-axes*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Acquired Waveform



The Acquired Waveform view is a continuous sweep measurement that displays the acquired waveform for each channel. Acquired waveform is plotted against time.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Interpolation

For time-domain displays, the APx500 graphs are normally plotted using sinc function interpolation. Interpolation adds points to the displayed trace that do not exist in the measured data, to make data trends more easily visualized. The default in APx is to have Interpolation On. However, digital domain signals are sometimes best understood when viewing actual samples, with interpolation switched Off. Also see “Interpolation and limit failure markers” on page 581.

Units

Units available for Acquired Waveform are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• s (seconds)	• V	• D • hex

More about ETC

The acquired Acoustic Response data is transformed into the frequency domain where further processing is performed, including the application of a Hilbert Transform. This result is then inverse transformed back to the time domain for display. This result is commonly called the energy time curve, or ETC.

The ETC is only an approximation of the actual energy arriving at the microphone, since this energy can never be known without the simultaneous measurement of both the velocity and the pressure of the sound; the term energy time curve is actually a misnomer. A more accurate term sometimes found in technical papers is the “analytic signal magnitude,” but as “energy time” has become common usage that term is used here.

Interpretations of ETC measurements can be misleading. The Energy Time Curve view is included in Acoustic Response as an aid in identifying first reflections for optimal setting of the Time Window.

The energy time curve is useful for determining arrival times and relative energy distribution in time. It is the log magnitude of the impulse response, and may be considered as similar to the envelope of the IR, tracing along the top of the IR curve but not showing any of the negative excursions.

More about Loudspeaker Testing

Loudspeakers can exhibit faults in several domains.

Response measurements

Loudspeaker response measurements focus on frequency, phase and distortion response at one or more level settings. Log-swept sine (chirp) measurements such as APx Acoustic Response (see page 173) are particularly effective for loudspeaker response measurements; stepped sine measurement and impulse response measurements can also be useful.

Rub and Buzz detection

Mechanical defects, such as a rubbing coil or particles in the gap, can introduce a number of different spurious signals into the acoustic output of a loudspeaker system. Rub and buzz signals are characteristically of higher frequency than the stimulus tone that excites the defect, and are of high crest factor.

In the Audio Precision implementation (see page 184), rub and buzz is detected by a process that first uses high-pass filtering, providing a residuals-only signal. The filter corner frequency slides upward during the measurement, tracking a multiple of the chirp stimulus frequency. This is followed by a crest factor measurement of the filtered residual, and a peak ratio measurement of the filtered residual vs. the unfiltered rms signal.

- Rub and buzz crest factor = residual signal peak / residual signal RMS.
- Rub and buzz peak ratio = residual signal peak / main (unfiltered) signal RMS.

A combination of high residuals crest factor (above 12 dB) and high peak ratio (above –40 dB) is a reliable indicator of a rub and buzz defect.

In APx500, rub and buzz detection is provided as an additional result to the Acoustic Response measurement. Both crest factor and peak ratio are shown on the same graph, using dual axes. The graph traces can be scaled and panned so that results indicating a defect fall on the same area of the graph, providing a helpful visual correspondence.

Because rub and buzz defects are impulsive and not periodic, averaging of results is not recommended. Averaging can make the defect signature disappear entirely.

Air Leak detection

Loudspeaker designs may specify a completely sealed enclosure, or an enclosure vented with acoustic ports. In either case, air leaks can degrade the performance, and can add spurious signals to the acoustic output of the system. Ideally, an enclosed loudspeaker system has no leaks, and a Modulated Noise Ratio of close to 0 dB. Speaker systems with a leak will present a Modulated Noise Ratio over 5 dB, sometimes as high as 30 dB.

Modulated Noise

A Modulated Noise ratio measurement (page 359) is provided in software option SW-SPK-RD and SW-SPK-PT as an air leak detection tool. In our implementation, an air leak is modeled as noise modulated by a sine wave. The signature of an air leak is short bursts of noise occurring at a peak of the stimulus fundamental frequency.

The Modulated Noise ratio measurement first filters the signal with a comb filter set at the stimulus frequency and its harmonics, followed by an optional comb filter at the mains frequency, followed by a high-pass filter to suppress environmental noise. The signal is then detected (its absolute value taken) and analyzed by FFT to provide the average power of the residual stimulus and its harmonics, divided by the average power of the noise. The square root of this quotient is converted to decibels to provide the final result.

Suppressing environmental noise

Successful detection of air leaks requires analysis uncontaminated by environmental noise, to the degree possible. Ideally, the loudspeaker and measurement microphone are placed in a quiet acoustic chamber. However, in production and in many R&D bench situations, environmental noise such as HVAC rumble, fan noise, and mains hum are unavoidable.

The comb filters in the Modulated Noise measurement do much to isolate the stimulus signal, and to optionally suppress mains interference. The high pass filter can also be very helpful in providing a useful result. If the loudspeaker is in a noisy environment, try moving the filter corner frequency upward from its default setting, with the goal of obtaining the highest ratio.

Using multiple microphones

An air leak may be located on the side or rear surface of an enclosure. If the measurement microphone is only directed to the front surface, the air leak may escape detection. Using an analyzer with two or more inputs enables the simultaneous use of multiple measurement microphones for more complete evaluation and detection.

Impedance/Thiele-Small

See Chapter 47.

The Impedance/Thiele-Small measurement measures the complex impedance of a loudspeaker driver, providing impedance response curves and Thiele-Small parameters.

Loudspeaker Production Test

See Chapter 54.

Loudspeaker Production Test combines two types of measurements in one test to facilitate fast loudspeaker testing in a production environment. An external power amplifier, a measurement microphone with power supply and an impedance fixture with a sense resistor are required. An acoustic isolation chamber is optional.

The two types of measurements are

- an acoustic measurement, providing frequency response, phase, distortion and rub and buzz results
  and
- an electromechanical impedance measurement, providing impedance response curves and a subset of Thiele-Small parameter results.

The underlying technology is the continuous sweep, providing both acoustic and impedance results in one fast measurement.

22

Bandpass Frequency Sweep
(Sequence Mode)

Introduction

For the APx515, this measurement requires a software option key. See page 166 for more information about software options.

The **Bandpass Frequency Sweep** measurement uses a sine wave stimulus signal that is moved across a range of frequencies in a specified number of points. The results are filtered using a bandpass filter that tracks the sweep frequency. Choose the filter width with the **Selectivity** control. X axis is generator frequency; Y axis is DUT output.

Bandpass Frequency Sweep measurement results available in APx500 are:

- RMS Level
- Gain
- Relative Level
- Deviation
- Peak Level
- Average Jitter Level

All of these results are available from a single measurement.

Operation

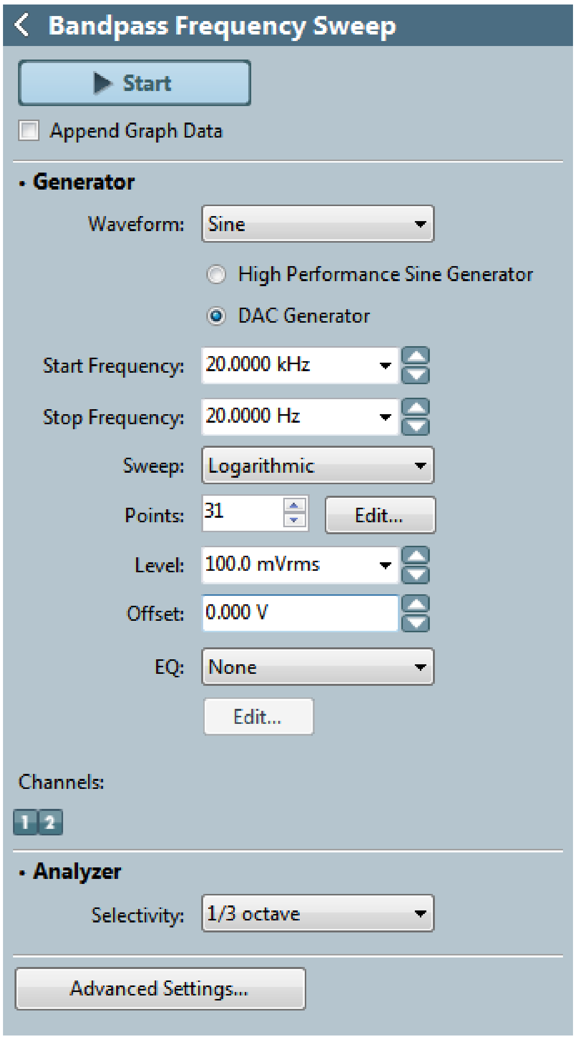
If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 6 for information about Signal Path Setup.

Start

For a default sweep of 20 kHz to 20 Hz in 31 logarithmic steps, simply click **Start**. If you want to specify other parameters for your sweep, review **Generator** and **Analyzer** settings, discussed below. Click **Start** when you have approved the settings.

Append Graph Data

Normally, the current graph data for all results is deleted each time you start a new measurement. If the **Append** box is checked, the current data is kept in memory as a **Data Set** (see page 165), and the new measurement data is appended as a new data set. You can append many **Data Sets**.



Generator

The typical configuration for this measurement is the closed-loop configuration, using the APx generator as a stimulus. Alternatively, you can make open-loop measurements, using either the External Source or File Input configuration.

See Chapter 5 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

Waveform

Select Sine for a closed loop measurement using the internal generator. If you select Browse for file to choose a generator waveform file, see "Using Sweep Tables with Stepped Frequency Sweep Measurements" on page 457.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 49 for a comparison of features.

Start Frequency

Set the sweep start frequency here.

Stop Frequency

Set the sweep stop frequency here.

Sweep

Choose

- Logarithmic point spacing
- Linear point spacing
- Custom
  Edit sweep spacing and number of Points to create a Custom sweep.

Points > Edit

Open the Sweep Points dialog to edit import or export the Sweep Points table.

Step Size

For linear sweeps, enter the Step Size here.

Level

Set the signal level here. This level may be changed if generator equalization is used, and can be adjusted on a per-channel basis in Advanced Settings (see page 458).

EQ

You can optionally modify the sweep with by applying a equalization curve to the generator. See Generator Equalization on page 170 for a general discussion of this feature. Choose

- None, which applies no equalization to the generator.
- Relative, which applies the gain or loss specified in the EQ table to the current generator level for each channel.
- Absolute, which sets the generator level for all channels to the levels specified in the EQ table. Note: when Absolute is selected, current channel level settings are lost.

Click the Edit button to create or change the EQ curve. See the Edit EQ Table dialog discussion on page 170.

Channels

Toggle specific output channels on or off by clicking on the Channel number button.

Analyzer

This measurement can be configured to measure jitter in Signal Path Setup > Input > Measure. Read detailed information about jitter generation and measurement beginning on page 60.

Filters

This measurement uses a tunable bandpass filter. No local low pass, high pass or weighting filters are available for this measurement.
However, low pass, high pass and weighting filter settings made in Signal Path Setup > Input/Output will affect this measurement. Read more about APx Filters in Chapter 91.

Selectivity

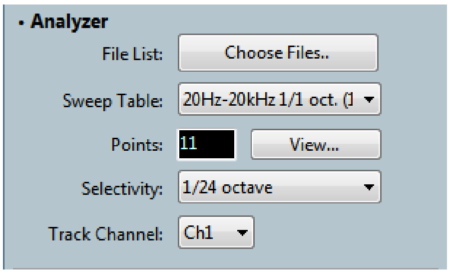
In the Bandpass Frequency Sweep measurement you can choose from a number of bandpass filter widths using the Selectivity control. The list is ordered from narrowest at the top, to widest at the bottom. The filter is tuned to the APx generator frequency, and follows the sweep.

- Window width
  This is the window width of the underlying FFT, typically only a few hertz wide. This selection has very steep skirts and a flat top.
- 1/24 octave
  This selection has –3 dB points that are 1/24 octave apart.

- 1/12 octave
  This selection has --3 dB points that are 1/12 octave apart.
- 1/9 octave
  This selection has –3 dB points that are 1/9 octave apart.
- 1/6 octave
  This selection has –3 dB points that are 1/6 octave apart.
- 1/3 octave
  This selection has –3 dB points that are 1/3 octave apart.
- 1 octave
  This selection has –3 dB points that are 1 octave apart.

All of the “x octave” filters are equivalent to second-order analog bandpass filters, with 6 dB/octave skirts. Q is chosen to achieve the specified bandwidth at the –3 dB points.

File List



This control is available if the Input Configuration in Signal Path Setup is set to one of the two File choices. Read more about analyzing audio files in Chapter 19.

Sweep Table

The Sweep Table menu, and the Points, View/Edit and Track Channel controls discussed below, are not shown for a closed loop sweep that uses the internal generator. In that case, the stimulus sweep definition is directly available to the analyzer and these controls are unnecessary.

When the measurement is configured so that the stimulus sweep definition is not available to the analyzer (in External Source, Generator Waveform or File Input configurations), you must select a Sweep Table (see page 457) that matches the definition of the sweep you are using.

The Sweep Table menu lists the embedded sweep tables and any new tables that have been added to the project. Select a table from the Sweep Table menu, or choose Create New or Browse for a sweep table file.

Points

This field displays the number of points in the Sweep Points table provided to the analyzer.

View / Edit

For embedded sweep tables, click View (see page 457) to view or export the table, or to create a WAV file from the table. For other sweep tables, click Edit to edit, view or export the table, or to create a WAV file from the table (see page 458).

Selectivity

See the Selectivity discussion on the previous page.

Track Channel

When using a sweep table (in External Source, Generator Waveform or File Input configurations), the analyzer must determine the relation of the incoming signal to the sweep table. When Auto is selected, the analyzer continually polls the all channel inputs throughout the sweep, switching between channels as necessary to track the input with the highest signal level.

Alternatively, you can select a specific input channel for the analyzer to use for sweep tracking.

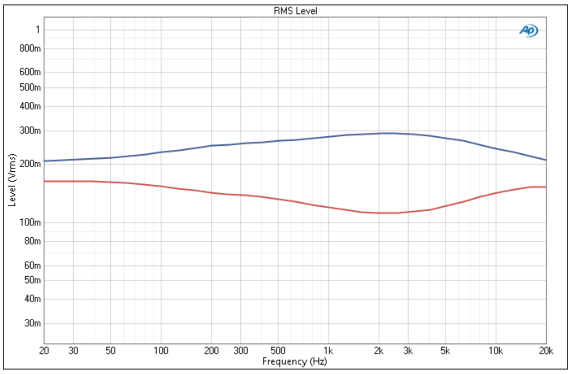
Advanced Settings

If your test required special adjustments or settings, click Advanced Settings. See page 458. See Chapter 98 for more information about units of measurement.

Data Set

If you have appended or imported graph data, the Data Set panel in the area beneath the graph display will be populated. Choose the data set(s) to be displayed on the graph. See page 165 for more information about Data Sets.

RMS Level



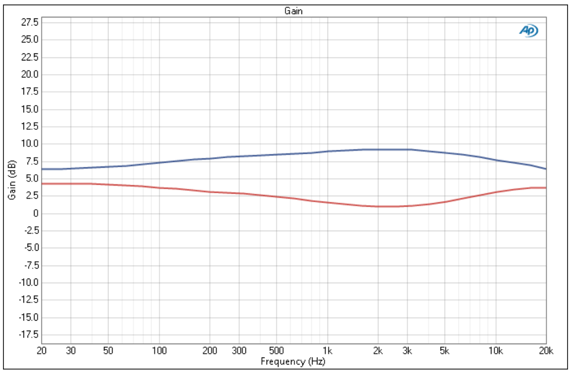
Bandpass Frequency Sweep: RMS Level results uses a sine wave stimulus signal that is moved across a range of frequencies in a specified number of points. This is the conventional frequency response sweep, bandpass filtered.

Units

Units available for Bandpass Frequency Sweep Level results are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*	*Jitter*
• Hz	• Vrms	• dBFS	• UI
• dHz	• dBV	• FS	• dBUI
• F/R	• dBu	• %FS	• s
• %Hz	• dBrA	• dBrA	
	• dBrB	• dBrB	
	• dBSPL1	• dBSPL1	
	• dBSPL2	• dBSPL2	
	• dBm		
	• W (watts)		

Gain



Gain is a Bandpass Frequency Sweep result that provides a graphical display of the voltage gain in each channel. In this result the DUT gain is plotted against frequency. The **Bandpass Frequency Sweep Gain** measurement is not available in **External Source** configuration.

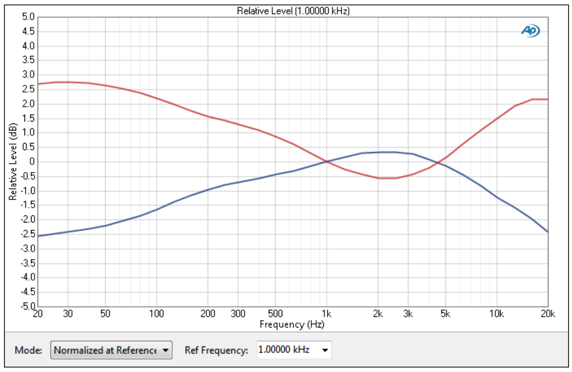
Gain results are not available when External Source is selected as the output configuration, when File is selected as the input configuration or when a generator waveform file is selected as the test signal.

Units

Units available for Bandpass Frequency Sweep: Gain are:

X-axis	*Y-axis same-domain*	*Y-axis cross-domain*
• Hz	• x/y	• FS/Vrms
• dHz	• %	• dB(FS/Vrms)
• F/R	• ppm	—or—
• %Hz	• dB	• Vrms/FS
		• dB(Vrms/FS)

Relative Level



The **Bandpass Frequency Sweep: Relative Level** result uses a sine wave stimulus signal that is moved across a range of frequencies in a specified number of points, providing a conventional frequency response sweep. Additionally, in this result the response curve is plotted in relation to the measured level at a selected frequency. This enables you to specify the frequency where the relative response will be unity and to view the response in relation to that frequency. A midband frequency such as 1 kHz is often set as the reference frequency, and is the default here.

Mode

Normalized at Reference

Enter a reference frequency for normalization in the **Ref Frequency** field. The measured level at the frequency you choose is set as 0 dB on the graph, and the DUT response is plotted in relation to this reference. You can change the **Ref Frequency** at any time (except after appending) and the graph will immediately be redrawn to reflect the new setting.

Auto-Centered in Limits

This control is only usable if upper and lower limits are set on this graph. In some applications, the general conformance of a trace to the shape of a set of limits is of importance, with absolute level less important or irrelevant. **Auto-Centered in Limits** centers the data trace relative to the upper and lower limits. To exclude possible exceptional data points at frequency extremes, the frequency range that is considered for the centering operation is constrained by the values in the **Min** and **Max** fields.

Ref Frequency

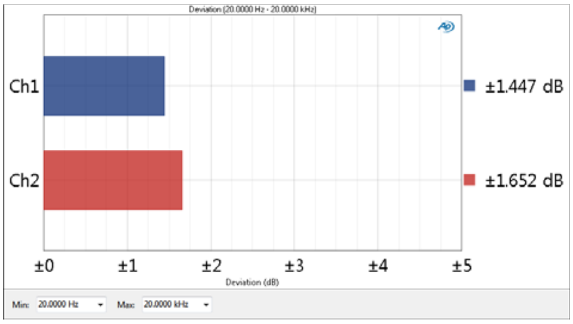
Choose a Reference Frequency here.

Units

Units available for Bandpass Frequency Sweep: Relative Level are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Deviation



The **Bandpass Frequency Sweep: Deviation** result uses a sine wave stimulus signal that is moved across a range of frequencies in a specified number of points. This result is a single value measurement computed from the stepped sweep that shows the frequency deviation (the total range of frequency variation) of each channel as a meter bar. You can specify a minimum and maximum frequency to define the range of the deviation measurement.

Min

Set the minimum frequency of the range of interest here.

Max

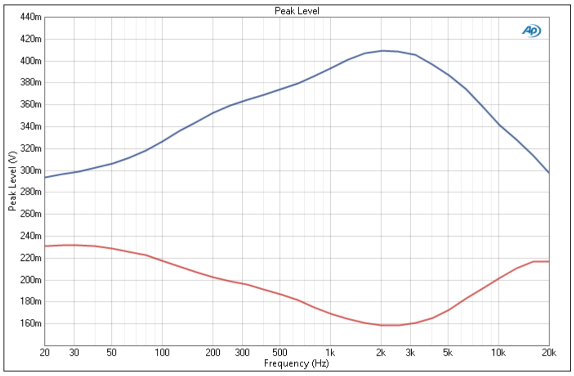
Set the maximum frequency of the range of interest here.

Units

Units available for Bandpass Frequency Sweep: Deviation are

- dB

Peak Level



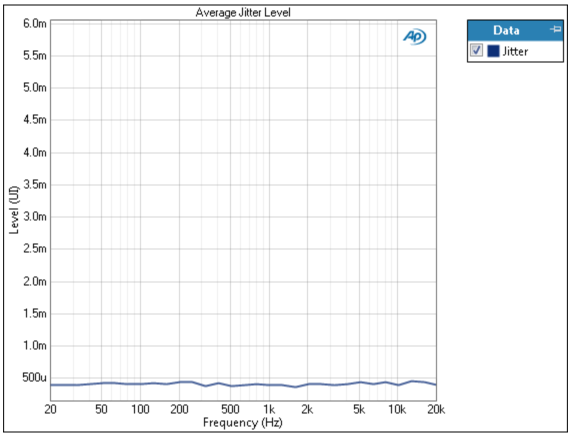
Bandpass Frequency Sweep: Peak Level results uses a sine wave stimulus signal that is moved across a range of frequencies in a specified number of points. This is a frequency response sweep, bandpass filtered, with peak-scaled results.

Units

Units available for Bandpass Frequency Sweep Peak Level results are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*	*Jitter*
• Hz	• V	• D	• UI
• dHz		• hex	• dBUI
• F/R			• s
• %Hz			

Average Jitter Level



Bandpass Frequency Sweep: Average Jitter Level results uses a sine wave stimulus signal that is moved across a range of frequencies in a specified number of points. The result is the bandpass filtered DUT jitter output, average scaled.

Units

Units available for Bandpass Frequency Sweep Average Jitter Level results are:

X-axis	***Jitter***
• Hz	• UI
• dHz	• dBUI
• F/R	• s
• %Hz	

23

Bandpass Level
(Sequence Mode)

Introduction

For the APx515, this measurement requires a software option key. See page 166 for more information about software options.

The Bandpass Level measurement provides a band-pass-filtered single-value result displaying the output level from each DUT channel, as measured at each analyzer input.

A bandpass filter is designed to pass a band of frequencies narrower than the system bandwidth. This range is called the *passband*. The *selectivity* of the filter determines the width of the passband; a highly selective filter will have a narrow passband.

Bandpass filters are often used for noise measurements, restricting the measurement to the frequency range of interest.

Bandpass Level results available in APx500 are:

- RMS Level
- Peak Level
- Average Jitter Level

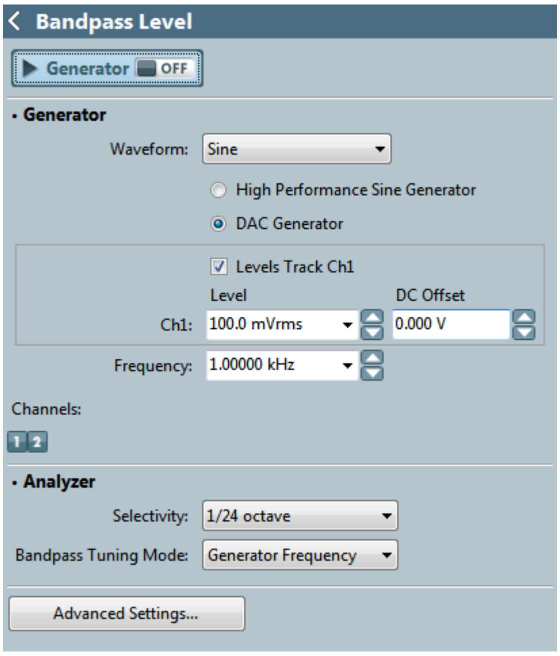
Average Jitter Level results are only available when Jitter is selected in Signal Path Setup > Input/Output > Measure.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 6 for information about Signal Path Setup.

Generator

The typical configuration for this measurement is the closed-loop configuration, using the APx generator as a stimulus. Alternatively, you can make open-loop measurements, using either the External Source or File Input configuration.



See Chapter 5 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 49 for a comparison of features.

Analyzer

This measurement can be configured to measure jitter in Signal Path Setup > Input > Measure. Read detailed information about jitter generation and measurement beginning on page 60.

Selectivity

In the Bandpass Level measurement you can choose from a number of bandpass filter widths using the **Selectivity** control. The list is ordered from narrowest at the top, to widest at the bottom.

- **Window width**
  This is the window width of the underlying FFT, typically only a few hertz wide. This selection has very steep skirts and a flat top.
- **1/24 octave**
  This selection has -3 dB points that are 1/24 octave apart.
- **1/12 octave**
  This selection has -3 dB points that are 1/12 octave apart.
- **1/9 octave**
  This selection has -3 dB points that are 1/9 octave apart.
- **1/6 octave**
  This selection has -3 dB points that are 1/6 octave apart.
- **1/3 octave**
  This selection has -3 dB points that are 1/3 octave apart.
- **1 octave**
  This selection has -3 dB points that are 1 octave apart.
- **Rectangular Band**
  This selection has very steep skirts and a flat top. The user defines the lower and upper frequency cutoff points. See **Min Freq: / Max Freq:** below.

All of the “x octave” filters are equivalent to second-order analog bandpass filters, with 6 dB/octave skirts. Q is chosen to achieve the specified bandwidth at the –3 dB points.

Bandpass Tuning Mode

The “Window width” and “x octave” filters are centered at some user-defined frequency for each measurement. This can be

- **Generator Frequency**
  The current APx audio generator frequency. When the generator channels are outputting different frequencies (Split Frequency generation), the bandpass filter center is set to the Frequency A. This mode is not available when using a generator waveform file.
- **Jitter Generator Frequency**
  The current APx jitter generator frequency, when jitter generation is available and enabled. Read detailed information about jitter generation and measurement beginning on page 60.
- **Measured Frequency**
  The current measured frequency.
  When the analyzer channels are receiving different frequencies, the bandpass filter for each channel is centered on the frequency in that channel.
- **Fixed Frequency**
  A fixed frequency selected by the user.
  When **Fixed Frequency** is selected, a **Filter Freq:** entry field becomes available beneath the **Bandpass Tuning Mode** control.

Min Freq: / Max Freq:

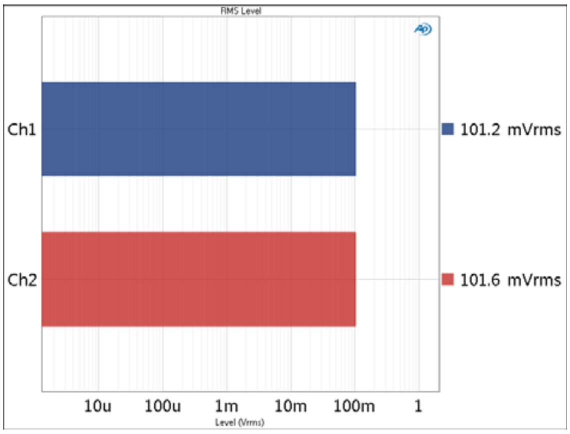
When **Rectangular Band** is the **Selectivity** choice, these fields become available for lower and upper frequency cutoff point entry.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, click **Advanced Settings**. See Advanced Settings for single value measurements on page 317.

See Chapter 98 for more information about units of measurement.

RMS Level



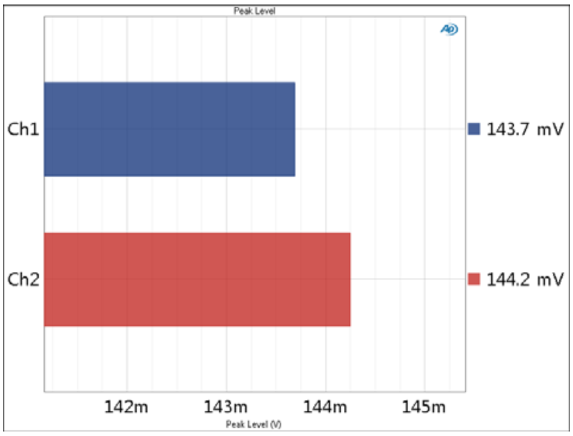
RMS Level provides a single-value meter result, displaying of the rms output level from each DUT channel, as measured at each analyzer input.

Units

Units available for RMS Level are

Analog Signals	*Digital Signals*	*Jitter signals*
• Vrms	• dBFS	• UI
• dBV	• FS	• dBUI
• dBu	• %FS	• s
• dBrA	• dBrA	
• dBrB	• dBrB	
• dBSPL1	• dBSPL1	
• dBSPL2	• dBSPL2	
• dBm		
• W (watts)		

Peak Level



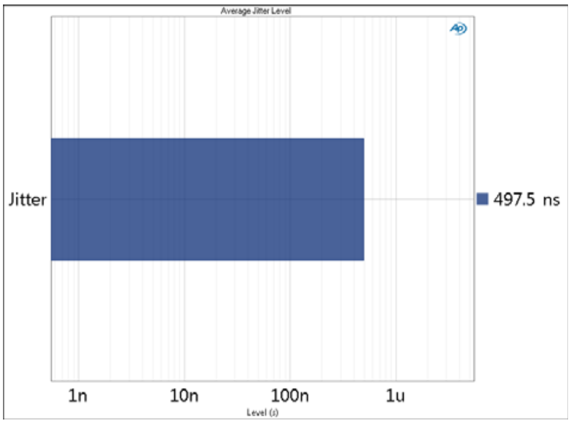
Peak Level provides a single-value meter result, displaying of the peak output level from each DUT channel, as measured at each analyzer input.

Units

Units available for Peak Level are

Analog Signals	*Digital Signals*	*Jitter signals*
• Vrms	• dBFS	• UI
• dBV	• FS	• dBUI
• dBu	• %FS	• s
• dBrA	• dBrA	
• dBrB	• dBrB	
• dBSPL1	• dBSPL1	
• dBSPL2	• dBSPL2	
• dBm		
• W (watts)		

Average Jitter Level



Average Jitter Level provides a single-value meter result, displaying the average jitter level from the digital receiver.

Units

Units available for Average Jitter Level are

Analog Signals	*Digital Signals*	*Jitter signals*
• Vrms	• dBFS	• UI
• dBV	• FS	• dBUI
• dBu	• %FS	• s
• dBrA	• dBrA	
• dBrB	• dBrB	
• dBSPL1	• dBSPL1	
• dBSPL2	• dBSPL2	
• dBm		
• W (watts)		

24

Bandpass Level Sweep
(Sequence Mode)

Introduction

For the APx515, this measurement requires a software option key. See page 166 for more information about software options.

The **Bandpass Level Sweep** measurement provides a sine wave stimulus signal that is moved across a range of levels in a series of points. The DUT output is acquired by the analyzer, and filtered using a bandpass filter. **Bandpass Level Sweep** measurements are not supported in the **External Source** configuration.

Bandpass Level Sweep results available in APx500 are:

- RMS Level
- Linearity
- Average Jitter Level
- Gain
- Peak Level

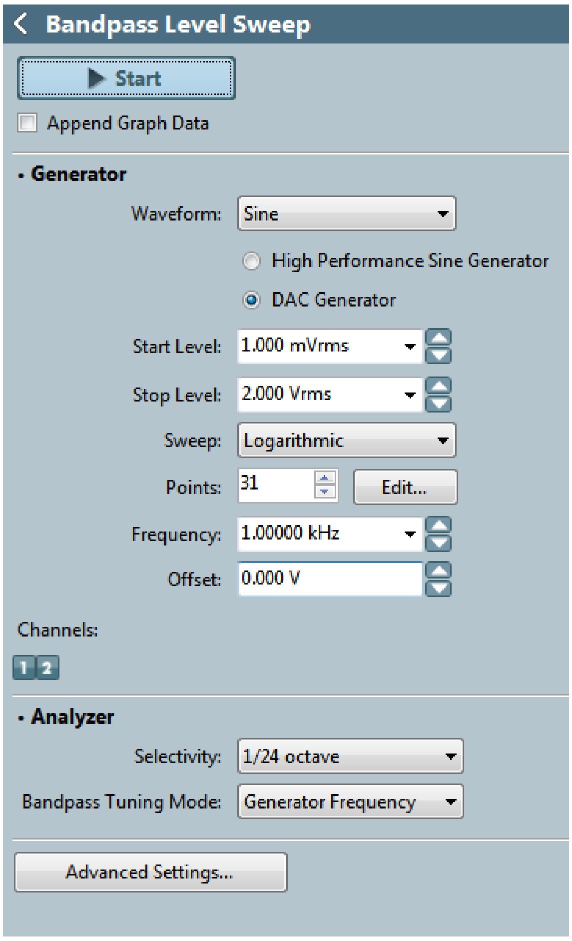
All of these results are available from a single measurement.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 6 for information about Signal Path Setup.

The **Bandpass Level Sweep** measurement uses a sine wave at the frequency and range of levels set in the Signal Generation panel as the test signal. This sine wave will be swept from the **Start Level** to the **Stop Level** in the set number of **Points**. The DUT output is acquired by the analyzer, and filtered using a bandpass filter.

Note: if Output Configuration is set to Bluetooth (A2DP Source profile), the Bluetooth hardware imposes a –54 dBFS cutoff. Signals below –54 dBFS are not transmitted. Please consider this when defining the range of levels for your sweep.



Start

For a default sweep of 1 mVrms to 2 Vrms in 31 logarithmic steps, simply click **Start**. If you want to specify other parameters for your sweep, review **Generator** and **Analyzer** settings, discussed below. Click **Start** when you have approved the settings.

Append Graph Data

Normally, the current graph data for all results is deleted each time you start a new measurement. If the **Append** box is checked, the current data is kept in memory as a **Data Set** (see page 165), and the new measurement data is appended as a new data set. You can append many **Data Sets**.

Generator

Waveform

Choose a **Special Sine** option (see page 158) to set a DC offset or to split frequency or phase across channels.

Start Level, Stop Level and Sweep Points

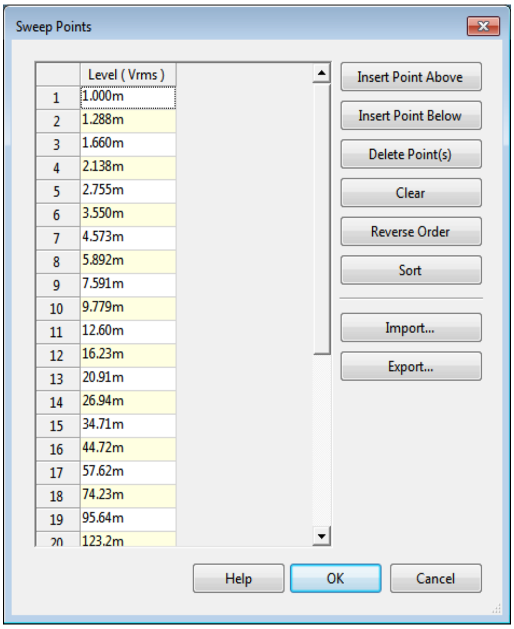
The sweep moves between two levels, set in the **Start Level** and **Stop Level** fields. The default levels are 1.000 mVrms and 2.000 Vrms.

The sweep moves in a specified number of step points, set in the **Points** field. The minimum is 2 points; maximum is 65,535. The default setting is 31.

The sweep point spacing is set by selecting one of the following choices in the Sweep field:

- **Logarithmic** (the default); use the **Points** field to set the number of logarithmically spaced points;
- **Linear**, which provides two methods of adjusting spacing: the **Points** field or the **Step Size** field; or
- **Custom**. Click **Edit** to open the **Sweep Points** dialog, where you can set points arbitrarily, or load or save sweep table files.

Viewing or Editing the Sweep Points table



You can view or edit the current sweep points at any time.

Click **Edit** to open the **Sweep Points** table. The table shows each sweep point and its corresponding level. You can edit this table to add or delete points, or to change the level of a point. Points can be sorted or reversed in order using the controls on the right.

A **Sweep Points** table can be saved as a *.csv file or as a Microsoft Excel *.xls file. A compatible *.csv or *.xls file can be opened and used as a **Sweep Points** table.

Analyzer

This measurement uses a tunable bandpass filter. No low pass, high pass or weighting filters are available for this measurement.
However, low pass, high pass and weighting filter settings made in Signal Path Setup > Input/Output will affect this measurement. Read more about APx Filters in Chapter 91.

Selectivity

In the **Bandpass Level Sweep** measurement you can choose from a number of bandpass filter widths using the **Selectivity** control. The list is ordered from narrowest at the top, to widest at the bottom. The **Window Width** and **x Octave** filters are tuned according to the **Bandpass Tuning Mode** setting.

- **Window width**
  This is the window width of the underlying FFT, typically only a few hertz wide. This selection has very steep skirts and a flat top.
- **1/24 octave**
  This selection has –3 dB points that are 1/24 octave apart.
- **1/12 octave**
  This selection has –3 dB points that are 1/12 octave apart.
- **1/9 octave**
  This selection has –3 dB points that are 1/9 octave apart.
- **1/6 octave**
  This selection has –3 dB points that are 1/6 octave apart.
- **1/3 octave**
  This selection has –3 dB points that are 1/3 octave apart.
- **1 octave**
  This selection has –3 dB points that are 1 octave apart.
- **Rectangular Band**
  This selection has very steep skirts and a flat top. The user defines the lower and upper frequency cutoff points. See **Min Freq: / Max Freq:** below.

All of the "x octave" filters are equivalent to second-order analog bandpass filters, with 6 dB/octave skirts. Q is chosen to achieve the specified bandwidth at the –3 dB points.

Min Freq: / Max Freq:

When **Rectangular Band** is the **Selectivity** choice, these fields become available for lower and upper frequency cutoff point entry.

Bandpass Tuning Mode

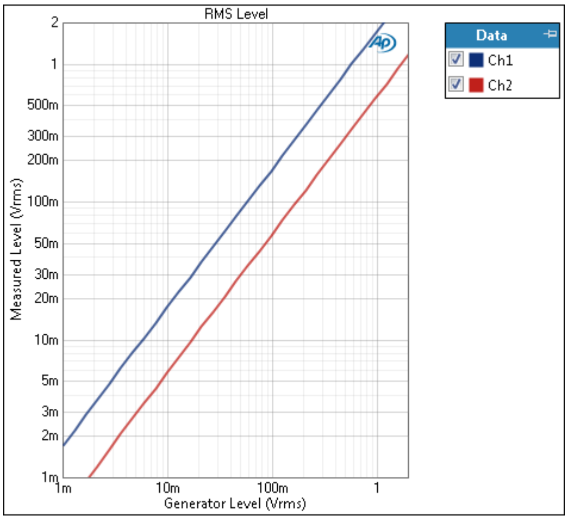
The "Window width" and "x octave" filters are centered at some user-defined frequency for each step of the measurement. This can be

- **Generator Frequency**
  The APx generator frequency for the current step. When the generator channels are outputting different frequencies (Split Frequency generation), the bandpass filter center is set to Frequency A.
- **Jitter Generator Frequency**
  The current APx jitter generator frequency, when jitter generation is available and enabled. Read detailed information about jitter generation and measurement beginning on page 60.
- **Measured Frequency**
  The measured frequency for the current step.
  When the analyzer channels are receiving different frequencies, the bandpass filter for each channel is centered on the frequency in that channel.
- **Fixed Frequency**
  A fixed frequency selected by the user. When **Fixed Frequency** is selected, a **Filter Freq:** entry field becomes available beneath the **Tune Mode** control.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, click **Advanced Settings**. See "Advanced Settings for Stepped Sweeps" on page 458. See Chapter 98 for more information about units of measurement.

RMS Level



The **Bandpass Level Sweep: RMS Level** result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is generator level; Y axis is the bandpass filtered DUT output level.

Units

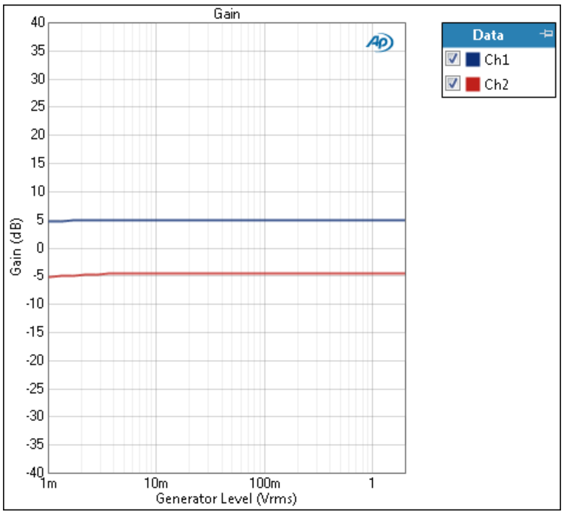
Units available for Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis (analog)*	*Y-axis (digital)*
• Vrms	• dBFS	• Vrms	• dBFS
• Vp	• FS	• dBV	• FS
• Vpp	• %FS	• dBu	• %FS
• dBV	• dBrG	• dBrA	• dBrA
• dBu		• dBrB	• dBrB
• dBrG		• dBSPL1	• dBSPL1
• dBm		• dBSPL2	• dBSPL2
• W (watts)		• dBm	
		• W (watts)	

Y-axis (jitter)

- UI
- dBUI
- s

Gain



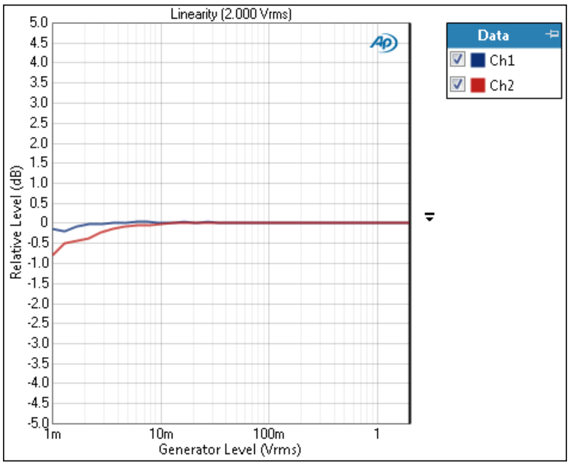
The **Bandpass Level Sweep: Gain** result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is generator level; Y axis is gain in the DUT. Gain is a ratio, calculated by dividing the bandpass filtered DUT output level by the generator level.

Units

Y-axis units available for Gain are:

X-axis (analog)	*X-axis (digital)*	*Y-axis same-domain*	*Y-axis cross-domain*
• Vrms	• dBFS	• x/y	• FS/Vrms
• Vp	• FS	• %	• dB(FS/Vrms)
• Vpp	• %FS	• ppm	—or—
• dBV	• dBrG	• dB	• Vrms/FS
• dBu			• dB(Vrms/FS)
• dBrG			
• dBm			
• W (watts)			

Linearity



The **Bandpass Level Sweep: Linearity** result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is generator level; Y axis is linearity in the DUT. Linearity is a ratio, calculated by dividing the bandpass filtered DUT output level by the generator level, normalized for a chosen relative level.

Mode

Normalized at Reference

Enter a reference level for normalization in the **Relative Level** field. To calculate linearity, each channel is normalized so that a linearity of unity (0 dB) is at the same measured DUT output level. Set this reference level in the **Relative Level** field. Default **Relative Level** is 1 Vrms (analog), 1 FS (digital). You can change the **Relative Level** at any time (except after appending) and the graph will immediately be redrawn to reflect the new setting.

Auto-Centered in Limits

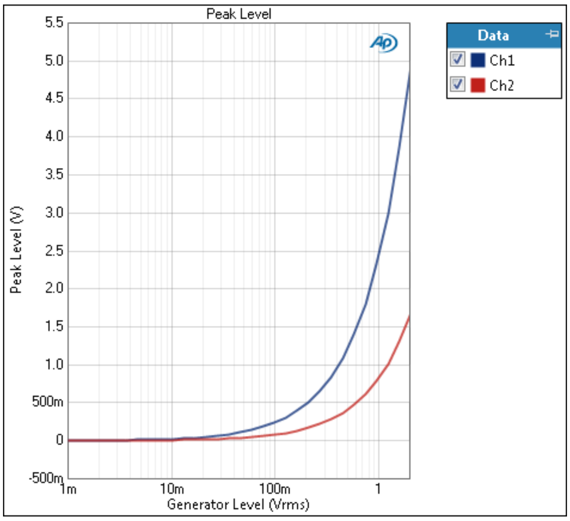
This control is only usable if upper and lower limits are set on this graph. In some applications, the general conformance of a trace to the shape of a set of limits is of importance, with absolute level less important or irrelevant. **Auto-Centered in Limits** centers the data trace relative to the upper and lower limits. To exclude possible exceptional data points at level extremes, the amplitude range that is considered for the centering operation is constrained by the values in the **Min** and **Max** fields.

Units

Units available for Linearity are:

X-axis (analog)	*X-axis (digital)*	*Y-axis*
• Vrms	• dBFS	• x/y
• Vp	• FS	• %
• Vpp	• %FS	• ppm
• dBV	• dBrG	• dB
• dBu		
• dBrG		
• dBm		
• W (watts)		

Peak Level



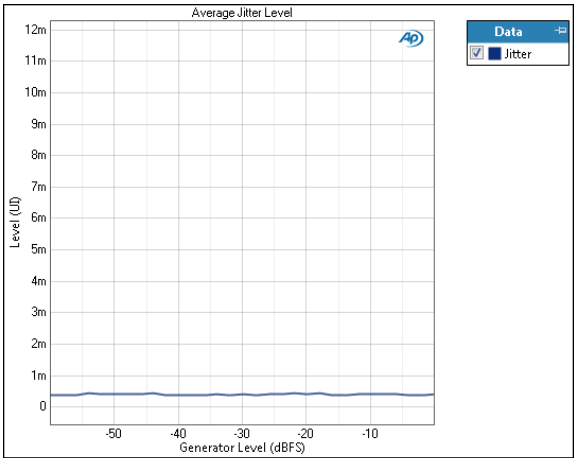
The **Bandpass Level Sweep: Peak Level** result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is generator level; Y axis is the bandpass filtered DUT output level, scaled in peak units.

Units

Units available for Peak Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis (analog)*	*Y-axis (digital)*
• Vrms	• dBFS	• V	• D
• Vp	• FS		
• Vpp	• %FS		
• dBV	• dBrG		
• dBu		*Y-axis (jitter)*	
• dBrG		• UI	
• dBm		• dBUI	
• W (watts)		• s	

Average Jitter Level



The **Bandpass Level Sweep: Average Jitter Level** result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is generator level; Y axis is the bandpass filtered DUT output jitter level, scaled in average units.

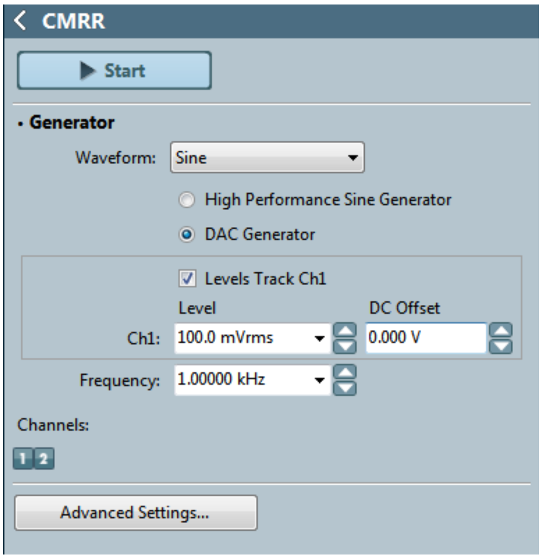
Units

Units available for Average Jitter Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis (jitter)*
• Vrms	• dBFS	• UI
• Vp	• FS	• dBUI
• Vpp	• %FS	• s
• dBV	• dBrG	
• dBu		
• dBrG		
• dBm		
• W (watts)		

25

CMRR (Sequence Mode)



CMRR is an analog measurement that uses hardware features found only in the 2-channel analog output circuitry fitted in an APx515, 525, 526, 555 or 582. CMRR measurements are not available in External Source configuration.

CMRR is an abbreviation of Common Mode Rejection Ratio. For a differential input such as a balanced audio input, “common mode” refers to a signal common to both the + and – inputs. Common mode signals are typically spurious signals picked up from the environment common to both leads, or sometimes a dc offset common to both leads. Common mode signals should be rejected to a high degree by differential amplifier inputs.

The CMRR measurement provides a single-value result, displaying the CMRR in each DUT channel.

Also see Chapter 26 for information about CMRR-IEC measurements.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 6 for information about Signal Path Setup.

Generator

This measurement must be made in the closed-loop configuration with balanced analog outputs, using the APx generator as a stimulus.

To measure CMRR, Click **Start**. The generator will output a sine wave to the DUT on the selected generator channels at the level and frequency set in the Signal Generation panel.

For the first half of the measurement, the generator is connected to the DUT in normal (differential) mode and a measurement of the normal mode rms level is made. The generator outputs are automatically switched to common mode using a pair of matched source resistors, one in each leg of the output circuit. A second measurement is made. The ratio of the two results is displayed as the Common Mode Rejection Ratio (CMRR).

See Chapter 5 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings.

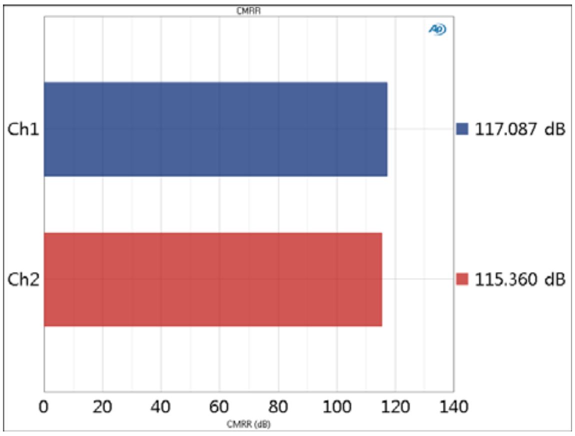
High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 49 for a comparison of features.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, click **Advanced Settings**. See Advanced Settings for single value measurements on page 317.

CMRR



Units

Units available for CMRR are

- x/y
- dB

See Chapter 98 for more information about units of measurement.

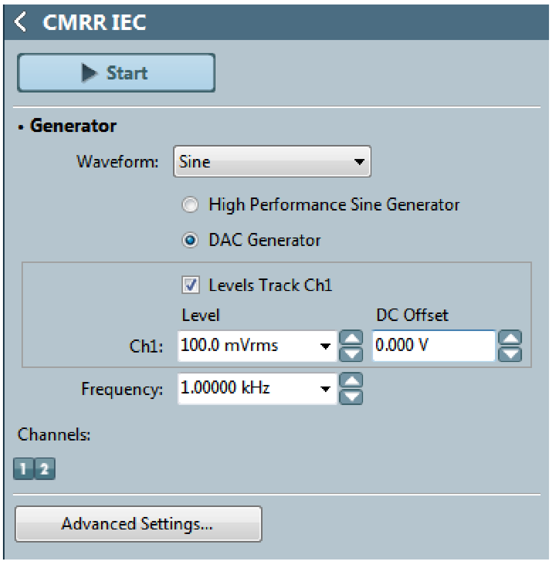
26

CMRR IEC
(Sequence Mode)

CMRR-IEC is an analog measurement that uses hardware features found only in the 2-channel analog output module fitted in an APx525, 526, 555 or 582. CMRR measurements are not available in External Source configuration.

CMRR IEC is a technique for measuring CMRR described in the International Electrotechnical Commission document IEC60268-3 that differs from the conventional technique used in the basic CMRR measurement discussed in Chapter 25. Use this method to produce CMRR results that satisfy IEC60268-3.

The CMRR IEC measurement provides single-value results showing the CMRR in each DUT channel by the IEC technique, which a unbalances the generator output circuit by first inserting a resistance in one leg (the Pin 2 measurement), and then the other leg (the Pin 3 measurement). The final CMRR IEC result is the worst-case of the pin 2 or pin 3 measurements.



Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 6 for information about Signal Path Setup.

Generator

This measurement must be made in the closed-loop configuration with balanced analog outputs, using the APx generator as a stimulus.

To measure CMRR IEC, Click **Start**. The generator will output a sine wave to the DUT on the selected generator channels at the level and frequency set in the Signal Generation panel.

For the first one-third of the measurement, the generator is connected to the DUT in normal (differential) mode and a measurement of the normal mode rms level is made. The generator outputs are automatically switched to use an IEC-defined common mode circuit, where one leg of the output circuit is connected directly to the generator output and the other leg is connected through an additional 10 ohm source resistor. A second measurement is made. Then the resistor and the direct connection are exchanged, and a third measurement is made. The worst ratio is reported as the CMRR IEC measurement result.

See Chapter 5 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings.

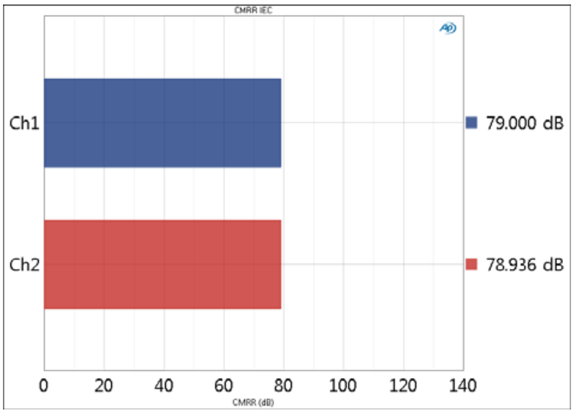
High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 49 for a comparison of features.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, click **Advanced Settings**. See Advanced Settings for single value measurements on page 317.

CMRR IEC



The CMRR Pin 2 and CMRR Pin 3 results are intermediate results, and are provided for diagnostic purposes. Report the data displayed in the CMRR IEC result.

Units

Units available for CMRR IEC are

- x/y
- dB

See Chapter 98 for more information about units of measurement.

27

Compare Encoded Bitstream to Reference (Sequence Mode)

This measurement is not available to the APx515.

This measurement allows you to compare the data in a coded audio bitstream to a reference file. It does not support comparison of linear PCM (IEC60958) bitstreams.

A typical use case is a coded audio (e.g. DTS, Dolby) compliance test, where a test disc (e.g., a Blu-ray Disc or DVD) and matching reference files are provided by a coded audio licensor to a licensee for verification of device compliance. The licensee plays the disc in bitstream mode via an HDMI or optical digital connection, for analysis in an APx analyzer whose HDMI sink EDID is configured to indicate support for such a bitstream. A bit-accurate comparison is made. If even one bit from the DUT does not match the reference file, the measurement indicates a failure.

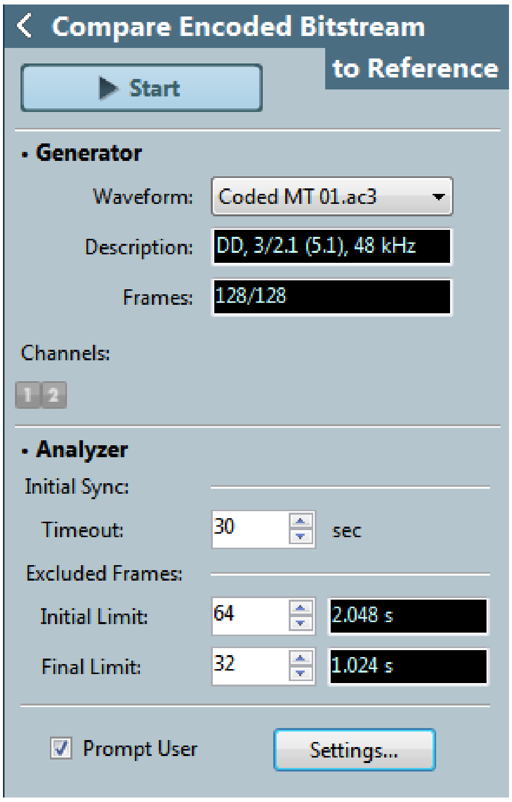
A number of frames at both the beginning and the end of the comparison can be excluded from triggering a measurement failure. These frames are received and examined, but any data mismatches are ignored. Use of these excluded areas prevents errors during disc player “spin up” and “spin down” times from causing the comparison to fail.

Received anomalies and their locations are reported, along with statistical summaries. For error frames, the hex data of both the reference file and the incoming data stream can be viewed.

Operation in Closed Loop mode

This measurement can be used in closed-loop mode to verify the compliance of a device such as an HDMI or IEC60958 repeater. If the APx DSIO module is fitted (for chip-level interface), HDMI or IEC60958 transmitter or receiver integrated circuits can be tested.

For Closed Loop operation, go to Signal Path Setup and select a digital output as your Output Configuration.



Signal Generation

Waveform

Choose a coded generator waveform file here. The same file will be used as the reference file for the comparison. The filename, description and frame count will be displayed in the associated fields.

Operation in Open Loop (External Source) mode

For Open Loop (External Source) operation such as disc player testing, go to Signal Path Setup and select None (External) as your Output Configuration.

Signal Acquisition and Analysis

Reference File:

Choose a reference file here. The file data must be bit-for-bit identical to the embedded coded data in the bitstream played from the DUT; typically, you will be using a test disc and matching test reference file provided by the coded audio licensor.

The **Description** and **Frame Count** will be displayed in the associated fields.

Initial Sync

Timeout

Many devices require a period of time to start, stabilize and synchronize after a play command has been issued. Set the maximum time allowed to wait for a valid frame here.

Excluded Frames

During startup, many devices issue invalid frames or skip frames altogether. You can exclude a range of frames at the beginning and the end of an acquisition, so that invalid frames in these regions will not cause the measurement to fail.

Initial Limit

Set the number of frames to exclude at the beginning of the acquisition in this field. The adjacent field shows the duration of the excluded range in seconds. This setting is also displayed as a gray band at the left side of the bar graph in the measurement panel.

Final Limit

Set the number of frames to exclude at the end of the acquisition in this field. The adjacent field shows the duration of the excluded range in seconds. This setting is also displayed as a gray band at the right side of the bar graph in the measurement panel.

Prompt User

You can optionally create an on-screen User Prompt for this measurement.

Settings

The optional User Prompt has Settings properties similar to the Pass/Fail Prompts, which are discussed on page 481.

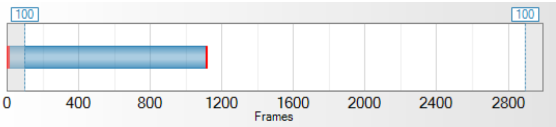
Results for Compare Encoded Bitstream to Reference

Overview

There are three possible results for this measurement.

- Failed due to timeout.
- Failed due to frame error.
- Passed.

Bar Graph Display



The top of the results area has a bar graph display that indicates received frames. Invalid frames are shown in red; valid frames are shown in blue. The excluded areas are represented by gray bands at the left and the right of the display.

Statistical and Descriptive Results

Reference File Frame Count:	2994
Received Frame Count:	1109
Valid Frame Range:	5 - 1111
Skipped Frame Range:	0 - 3
Excluded Error Frames:	4
Error Frame Number:	1112
Error Frame Description:	One or more data values in the fra...

The center of the results area reports statistical and descriptive information as text, for diagnostic purposes.

Reference File Frame Count

The number of frames in the reference file.

Received Frame Count

The number of valid frames received.

Valid Frame Range

The range(s) of valid frames.

Skipped Frame Range

The range(s) of skipped frames.

Excluded Error Frame Range

Error range(s) of error frames in the excluded areas.

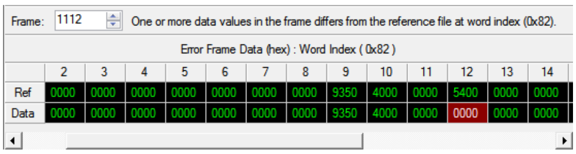
Error Frame Number

The number of the error frame found in the data area (between the excluded areas).

Error Frame Description

A description of the type of fault found in the error frame.

Data Grid viewer



The bottom of the results area displays specific information about invalid and skipped frames.

The Frame window allows you to select among the invalid or skipped frames for more information, which is shown in the data grid below.

The data grid displays hex values for 24 bytes of the selected frame, for both the reference file and the acquired data.

28

Continuous Sweep Measurements

Introduction

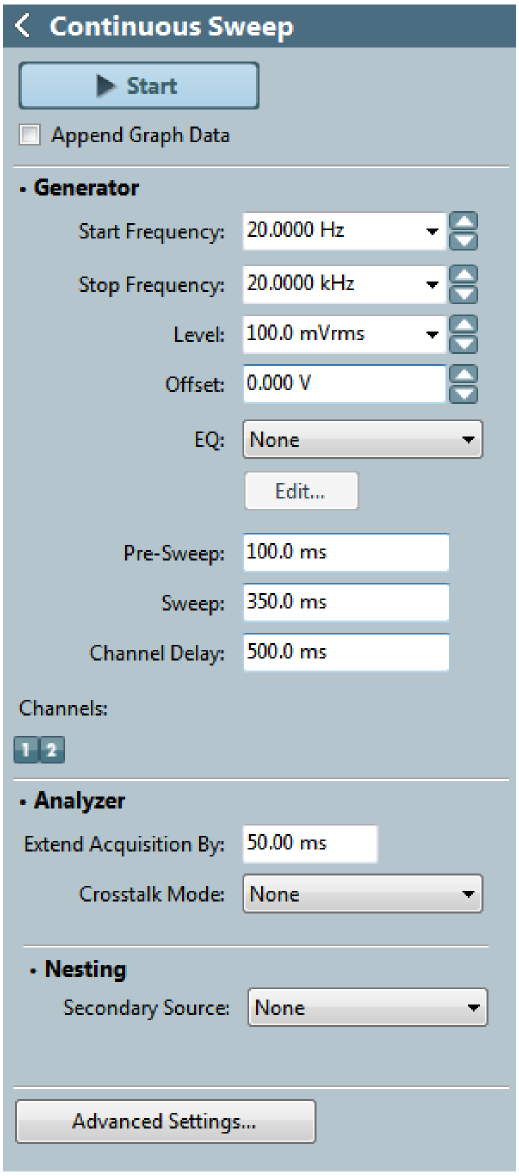
For the APx515, this measurement requires a software option key. See page 166 for more information about software options.

A Continuous Sweep is a brief log-swept sine wave (a Farina log chirp) that moves continuously across a specified range of frequencies. The DUT output is acquired by the analyzer and is mathematically processed to provide a number of results. For more information about continuous sweep measurements, go to page 220.

Continuous sweep based measurements, ire not supported in External Source or File Input configurations.

Continuous Sweep results available in APx500 are:

• Level	• Phase	• Distortion Product Ratio	• Crosstalk One Channel Driven
• Gain	• Group Delay	• Distortion Product Level	• Acquired Crosstalk Waveform One Channel Driven
• Relative Level	• THD Ratio	• Impulse Response	• Crosstalk, One Channel Undriven
• Deviation	• THD Level	• Acquired Waveform	• Acquired Crosstalk Waveform One Channel Undriven



Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 6 for information about Signal Path Setup.

Append Graph Data

Measured data are grouped in a Data Set, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the Append Graph Data box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 573.

Generator

The Continuous Sweep measurement uses a log-swept sine chirp stimulus signal, swept between the frequencies entered in Start Frequency and Stop Frequency, at the value set in Level.

Running the Measurement

To use Continuous Sweep, click Start. The generator will output the test signal to the DUT on the selected generator channels. Read the results in the selected result view.

Start Frequency and Stop Frequency

These controls set the frequency range of the sweep. See page 220 for more about continuous sweep ranges, which are dependent upon the instrument.

Level

Set the generator level here.

Offset

Set any DC offset to the generator signal here.

EQ

You can equalize the generator signal to make a pre-emphasized stimulus. See Generator Equalization (page 170) for a general discussion of this feature.

Choosing generator equalization

Select None, Relative or Absolute from the EQ drop-down menu.

- None applies no equalization to the generator.
- Relative applies the gain or loss specified in the EQ table to the current generator level for each channel.
- Absolute sets the generator level for all channels to the level specified in the EQ table. Note: when Absolute is selected, current channel level settings are lost.

Edit

If you have chosen to equalize the generator signal, you can edit the EQ table here. See Edit EQ Table (page 170).

Pre-Sweep and Sweep duration fields

A continuous sweep stimulus consists of a sine wave swept logarithmically across the sweep range in the period of time set in these fields.

To enable the DUT to stabilize before the beginning of the defined sweep measurement, there is variable Pre-Sweep time. The actual sweep stimulus signal begins below the defined Start Frequency and is swept upward, reaching the Start Frequency at the end of the Pre-Sweep time. The sum of the times in Pre-Sweep and Sweep is the total sweep length.

- The default for Pre-Sweep is 100.0 ms. Minimum is 0 s; maximum is 1.0 s.
- The default for Sweep is 350.0 ms. Minimum is 50.0 ms; maximum is 5 s.

Longer Pre-Sweep settings allow more time for DUT stabilization, but increase measurement time. Longer Sweep settings provide greater resolution and signal-to-noise ratio, but increase measurement time.

Channel Delay

The Channel Delay control allows you to specify a delay before the Start moments for each channel when in one of the High Accuracy Crosstalk modes.

If the Channel Delay is zero or very short, the slew rate limiting of a DUT or the time constant in AC Coupling can cause a transient when energy from the sweep in a previous channel appears in the analysis of the current channel. The default Channel Delay is 500 ms. If DC coupled and the DUT does not have appreciable slew rate limiting, Channel Delay can be set to a very low value, or 0.

Analyzer

The sweep is acquired and processed to provide the various results.

There are no local measurement filters available for continuous sweep measurements. However, low pass, high pass and weighting input filter settings made in Signal Path Setup > Input/Output > Filters will also affect this measurement. See Chapter 91 for more information about filtering.

Extend Acquisition By

By default, a continuous sweep measurement makes an acquisition slightly longer than the stimulus, to include possible time-delayed artifacts created in the DUT. By default, the acquisition is extended 50 ms longer than the stimulus.

In some cases, you may want to extend the acquisition further. Enter a new value in the **Extend Acquisition By:** field. Minimum extension is 0 s; maximum is 3 s.

Crosstalk Mode

Choose the mode of crosstalk measurement (or None). This selection changes the sweep alignment in time across channels, and may affect other continuous sweep results. For more information, see Continuous Sweep Crosstalk modes on page 222.

Choices are

- None
- **High speed** (default)
- High accuracy, one channel driven
- High accuracy, one channel undriven

Nesting

Continuous Sweeps can be nested. Read about Nested Sweeps beginning on page 161.

Secondary Source (nested sweeps)

Select the secondary sweep source here. This is the outside, nesting parameter that is varied to change the conditions for the primary sweep through a number of iterations.

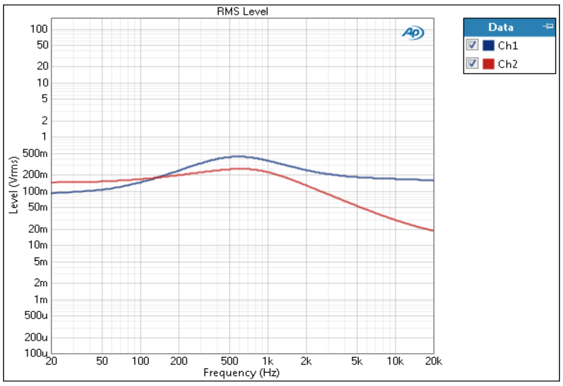
Available choices are dependent upon instrument hardware, options, Input/Output settings and Bench Mode Generator settings.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging, see Advanced Settings for Continuous Sweep on page 223.

See Chapter 98 for more information about units of measurement.

Continuous Sweep: Level



The Level result is a continuous sweep measurement that provides a graphical display of the frequency response of each channel. In this result the DUT output level is plotted against frequency.

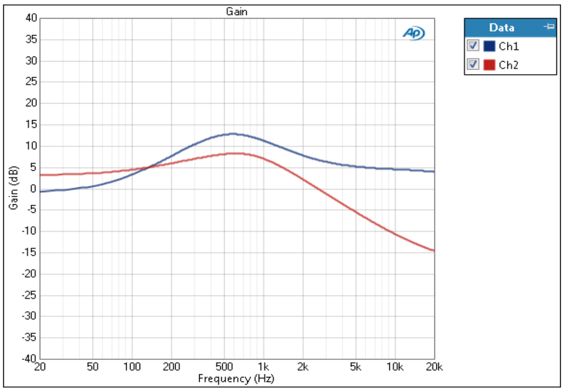
If the settings are identical, this will provide the same results as the **Frequency Response: Level** result.

Units

Units available for Continuous Sweep: Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz	• Vrms	• dBFS
• dHz	• dBV	• FS
• F/R	• dBu	• %FS
• %Hz	• dBrA	• dBrA
	• dBrB	• dBrB
	• dBSPL1	• dBSPL1
	• dBSPL2	• dBSPL2
	• dBm	
	• W (watts)	

Continuous Sweep: Gain



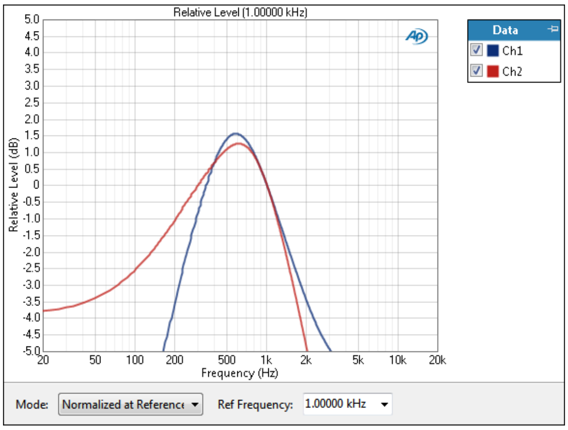
The Gain result is a continuous sweep measurement that provides a graphical display of the frequency response of each channel. In this result the DUT gain is plotted against frequency.

Units

Units available for Continuous Sweep: Gain are:

X-axis	*Y-axis same-domain*	*Y-axis cross-domain*
• Hz	• x/y	• FS/Vrms
• dHz	• %	• dB(FS/Vrms)
• F/R	• ppm	—or—
• %Hz	• dB	• Vrms/FS
		• dB(Vrms/FS)

Continuous Sweep: Relative Level



The Relative Level result is a continuous sweep measurement that provides a graphical display of the frequency response of each channel. In this result the DUT output level is plotted against frequency, relative to the level at a selected reference frequency. This enables you to specify the frequency that will be set as 0 dB and view the response in relation to that frequency. A midband frequency such as 1 kHz is often set as the reference frequency, and is the default here.

If the settings are identical, this will provide the same results as the Frequency Response: Relative Level result.

You can change the reference frequency at any time (except after appending) and the graph will immediately redraw around the new reference.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Mode

Normalized at Reference

Enter a reference frequency for normalization in the **Ref Frequency** field. The measured level at the frequency you choose is set as 0 dB on the graph, and the DUT response is plotted in relation to this reference. You can change the **Ref Frequency** at any time (except after appending) and the graph will immediately be redrawn to reflect the new setting.

Auto-Centered in Limits

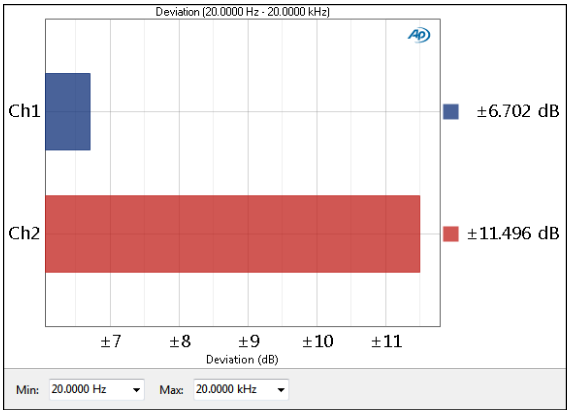
This control is only usable if upper and lower limits are set on this graph. In some applications, the general conformance of a trace to the shape of a set of limits is of importance, with absolute level less important or irrelevant. **Auto-Centered in Limits** centers the data trace relative to the upper and lower limits. To exclude possible exceptional data points at frequency extremes, the frequency range that is considered for the centering operation is constrained by the values in the **Min** and **Max** fields.

Units

Units available for Continuous Sweep: Relative Level are:

X-axis	***Y-axis***
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Continuous Sweep: Deviation



Continuous Sweep Deviation is a single value result computed from the continuous sweep acquisition. In this result the frequency deviation (the total range of frequency variation) of each channel is displayed as a meter bar. You can specify a minimum and maximum frequency to define the range be considered in the deviation measurement.

To measure Continuous Sweep Deviation, first enter a frequency range in the **Min Frequency** and **Max Frequency** fields below the graph; or, accept the default 20 Hz to 20 kHz range. Click **Start.** you can change the deviation **Min Frequency** or **Max Frequency** settings at any time and the meter bars will immediately redraw to reflect the new settings.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Min

Set the minimum frequency of the range of interest here.

Max

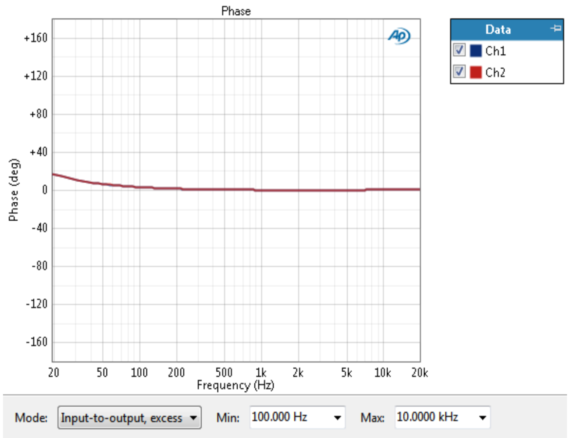
Set the maximum frequency of the range of interest here.

Units

Units available for Continuous Sweep: Deviation are

- dB

Continuous Sweep: Phase



The Continuous Sweep: Phase result displays the DUT output phase for each channel, plotted against frequency. See page 300 for more information about phase measurements.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Mode

Continuous Sweep Phase has four result modes:

Relative to Ch1

In this mode, the absolute phase of channel one is subtracted from the absolute phase of channels numbered greater than 1. The result is plotted against frequency for each channel numbered 2 and above, showing the phase difference (from channel 1) for each channel. Since channel 1 is used as the reference, it is not plotted in this result. This mode shows "unwrapped" phase differences.

Input-to-output

In the input-to-output mode, the absolute phase of each channel, from device input to device output, is plotted against frequency, "unwrapped." Input-to-output mode includes device delay.

Input-to-output, wrapped

This mode shows the same result as input-to-output phase, but "wrapped" within the range of –180° to +180°.

Input-to-output, excess

This mode shows the input-to-output phase (unwrapped), but removes the linear component (the average group delay of the system), leaving the "excess phase."

Some devices have linear phase only over a portion of their passband. For such devices, including frequency ranges where the phase is non-linear affects the excess phase result, which becomes dependent on the stop frequency of the measurement. Although the measurement is technically correct, the variation of the result with stop frequency is disconcerting and not particularly useful.

You may (optionally) use the **Min:** and **Max:** settings to set the expected linear phase range of the device under test. For this result, the device delay is computed as the best linear fit to the phase data between the **Min** frequency and the **Max** frequency. Excess phase is then computed as the difference from the linear fit.

Min and **Max** frequencies cannot extend beyond sweep **Start** and **Stop** frequencies.

See more information about wrapped and unwrapped phase in the next paragraphs.

Units

Units available for Continuous Sweep: Phase are:

X-axis	***Y-axis***
• Hz	• deg
• dHz	• rad
• F/R	
• %Hz	

More about Wrapped and Unwrapped Phase

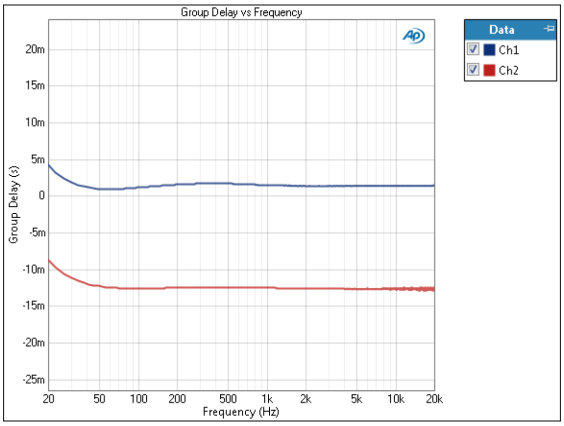
Wrapped phase results

Phase angles within a circle are typically expressed in degrees within the range of 0° to 360°. Using this frame of reference, if the phase of a signal has shifted by exactly 360°, it is considered to have "wrapped" back to 0° and is now "back in phase." Raw, unwrapped phase values are converted to wrapped values by subtracting 360 *n* times, until there is a remainder between 0° and 360°. This remainder is the wrapped value. For example, an unwrapped phase measurement of 742° would be displayed as a wrapped value of 22°. APx500 expresses wrapped phase using the range of –180° to +180°.

Unwrapped phase results

Phase angles of many thousands of degrees can be shown in an unwrapped phase mode.

Continuous Sweep: Group Delay



Group delay is a measure of the rate of change of phase shift as a function of frequency. Group delay can also be described as a time delay of a group of frequencies with respect to the generator.

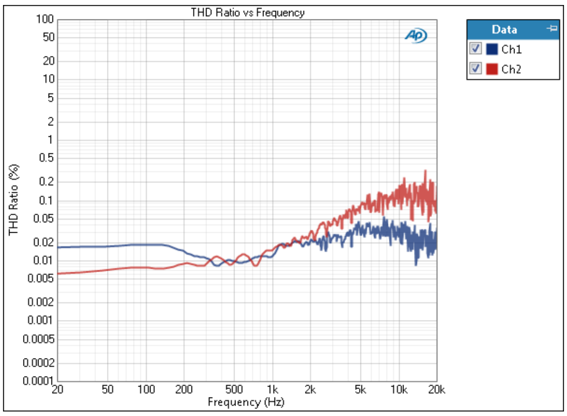
The Group Delay result is a continuous sweep measurement that displays the group delay measured in each channel, plotted against frequency.

Units

Units available for Continuous Sweep: Group Delay are:

X-axis	*Y-axis*
• Hz	• s (seconds)
• dHz	
• F/R	
• %Hz	

Continuous Sweep: THD Ratio



The THD Ratio result is a continuous sweep measurement that provides a graphical display of the harmonic distortion response of each channel. In this result the ratio of the level of the THD (total harmonic distortion) to the fundamental in the DUT output is plotted against frequency.

See more information about THD below.

Units

Units available for Continuous Sweep: THD Ratio are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

More about THD

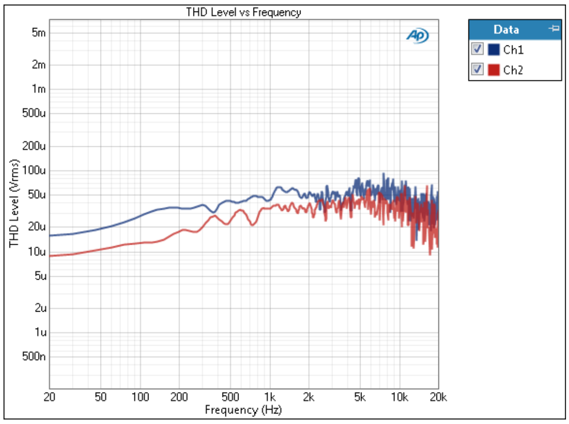
THD is an abbreviation for Total Harmonic Distortion. As above, this refers to the sum of all the harmonic distortion products in a measurement.

THD without the noise was not easily done until the advent of FFT measurement techniques. The APx500 implementation measures only the bins that contain harmonic distortion products, producing a result that is not influenced by DUT noise or spurious interfering signals. In a very low-noise system, the THD result will be the same as the THD+N result.

The FFT method that provides THD results also brings the capability to selectively measure discrete harmonic distortion products.

Also see More about THD+N on page 475.

Continuous Sweep: THD Level



The THD Level result is a continuous sweep measurement that provides a graphical display of the harmonic distortion response of each channel. In this result the level of the THD (total harmonic distortion) in the DUT output is plotted against frequency.

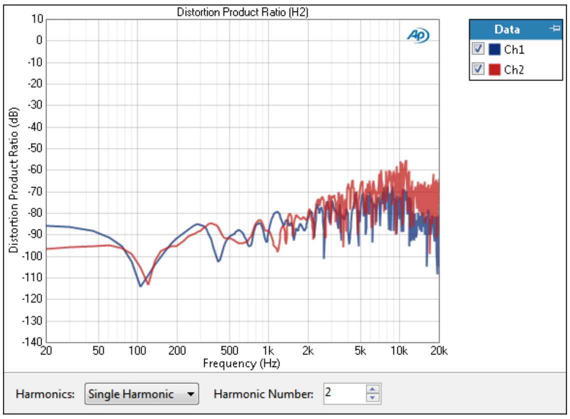
See the previous topic for more information about THD.

Units

Units available for Continuous Sweep: THD Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz	• Vrms	• dBFS
• dHz	• dBV	• FS
• F/R	• dBu	• %FS
• %Hz	• dBrA	• dBrA
	• dBrB	• dBrB
	• dBSPL1	• dBSPL1
	• dBSPL2	• dBSPL2
	• dBm	
	• W (watts)	

Continuous Sweep: Distortion Product Ratio



The Continuous Sweep Distortion Product Ratio result is a continuous sweep measurement that provides a graphical display of the selected harmonic distortion products present in each channel. In this result the ratio of the level of the selected harmonic distortion product to the fundamental in the DUT output is plotted against frequency.

Select the harmonic distortion product(s) you wish to view using the **Harmonics** controls, described below.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Harmonics

For a graph of the ratio of one specific harmonic product to the signal level, select **Single Harmonic** and set the **Harmonic Number** to your choice, from 2nd harmonic to 20th harmonic.

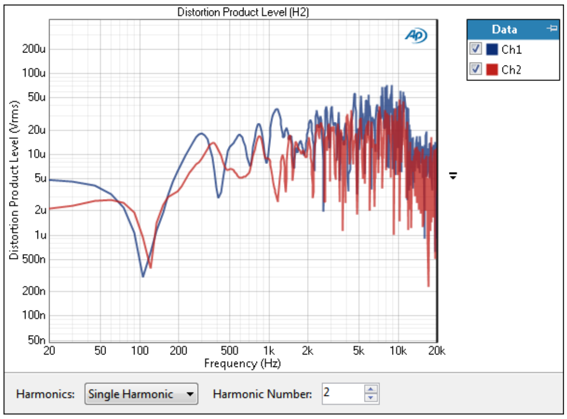
For a graph of the sum of the levels of any combination of harmonic products (from **H2** through **H20**), divided by the signal level, select **Sum of Harmonics**. Then click **Select Harmonics...** to open a result definition dialog.

Units

Units available for Continuous Sweep Distortion Product Ratio are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Continuous Sweep: Distortion Product Level



The Continuous Sweep Distortion Product Level result is a continuous sweep measurement that provides a graphical display of the selected harmonic distortion products present in each channel. In this result the level of the selected harmonic distortion product in the DUT output is plotted against frequency.

Select the harmonic distortion product(s) you wish to view using the **Harmonics** controls, described below.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Harmonics

For a graph of the level of one specific harmonic product, select **Single Harmonic** and set the **Harmonic Number** to your choice, from 2nd harmonic to 20th harmonic.

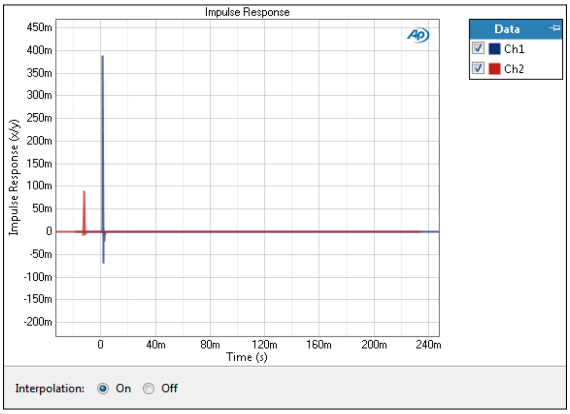
For a graph of the level of the sum of several or all harmonic products (from **H2** through **H20**), select **Sum of Harmonics**, then click **Select Harmonics...** to open a result definition dialog.

Units

Units available for Continuous Sweep Distortion Product Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz	• Vrms	• dBFS
• dHz	• dBV	• FS
• F/R	• dBu	• %FS
• %Hz	• dBrA	• dBrA
	• dBrB	• dBrB
	• dBSPL1	• dBSPL1
	• dBSPL2	• dBSPL2
	• dBm	
	• W (watts)	

Continuous Sweep: Impulse Response



The Impulse Response result is a continuous sweep measurement that displays the impulse response for each channel. Impulse response is plotted against time.

See page 218 for more information about impulse response.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Interpolation

For time-domain displays, the APx500 graphs are normally plotted using sinc function interpolation. This is the default setting. However, digital domain signals are sometimes best understood when viewing actual samples, with no interpolation. Turn display interpolation ON or OFF here.

Units

Units available for Impulse Response are:

X-axis	*Y-axis same-domain*	*Y-axis cross-domain*
• s (seconds)	• x/y	• D/V
		—or—
		• V/D

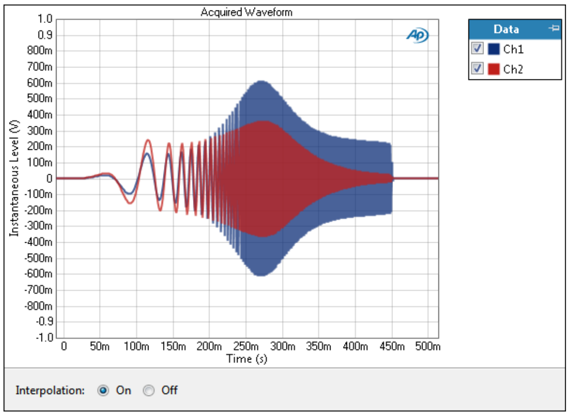
More about Impulse Response

The impulse response of the DUT is derived from the APx500 log-swept continuous sweep measurement. This gives the same result (although with better signal to noise ratio) as would be produced by stimulating the DUT with an impulse and observing the output in the time domain. This result is useful to study artifacts that move or spread the stimulus in the time domain, such as delay, reverberation, echo or reflection, etc.

Impulse response is a property of the device or system under test, and for same-domain measurements (analog to analog, for example) the impulse response is unitless, which we represent in APx500 as “x/y”. For cross-domain measurements, the impulse response is expressed as the relationship between output and input. A DAC, for example, has an impulse response expressed in units of V/D. See Chapter 98 for more information about units of measurement.

An ideal impulse response does not change with stimulus level.

Continuous Sweep: Acquired Waveform



The Acquired Waveform result is a continuous sweep measurement that displays the Acquired Waveform for each channel. See page 220 for more information about acquired waveform.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Interpolation

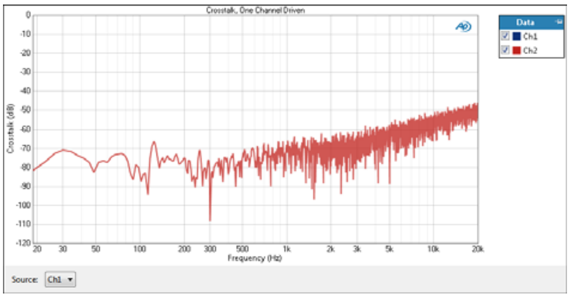
For time-domain displays, the APx500 graphs are normally plotted using sinc function interpolation. Interpolation adds points to the displayed trace that do not exist in the measured data, to make data trends more easily visualized. The default in APx is to have Interpolation **On**. However, digital domain signals are sometimes best understood when viewing actual samples, with interpolation switched **Off**. Also see “Interpolation and limit failure markers” on page 581.

Units

Units available for Acquired Waveform are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• s (seconds)	• V	• D • hex

Continuous Sweep: Crosstalk, One Channel Driven



The Crosstalk, One Channel Driven result provides a graphical display of the crosstalk from a selected source channel into each of the other channels. See page 238 for more information about crosstalk.

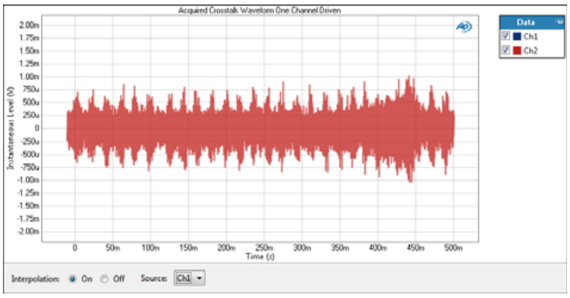
Units

Crosstalk is expressed as a ratio, with the crosstalk in the measured channel divided by the actual signal level in that channel. The result is typically stated in decibels.

Units available for Continuous Sweep: Crosstalk, One Channel Driven are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Continuous Sweep: Acquired Crosstalk Waveform, One Channel Driven



The Acquired Crosstalk Waveform, One Channel Driven result is a continuous sweep measurement that displays the Acquired Crosstalk Waveform for each undriven channel. This acquisition is only made when Crosstalk Mode is set to **High Accuracy, One Channel Driven**. See page 220 for more information about acquired waveform.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Interpolation

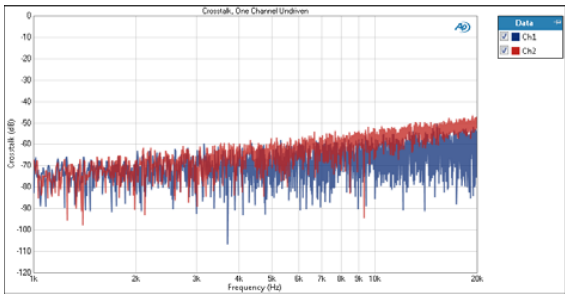
For time-domain displays, the APx500 graphs are normally plotted using sinc function interpolation. Interpolation adds points to the displayed trace that do not exist in the measured data, to make data trends more easily visualized. The default in APx is to have Interpolation **On**. However, digital domain signals are sometimes best understood when viewing actual samples, with interpolation switched **Off**. Also see “Interpolation and limit failure markers” on page 581.

Units

Units available for Acquired Crosstalk Waveform are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• s (seconds)	• V	• D • hex

Continuous Sweep: Crosstalk, One Channel Undriven



The Crosstalk, One Channel Undriven result is a continuous sweep measurement that provides a graphical display of the combined crosstalk into each channel from all the other channels. See page 238 for more information about crosstalk.

To measure make a continuous sweep Crosstalk, One Channel Undriven measurement, first select a source channel the **Source** list beneath the graph. Click **Start**. You can select another Source channel, and repeat the test by clicking Start again.

You can change the Source channel selection at any time (except after appending) and the graph will immediately redraw to show crosstalk from that new source.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Source

Designate a channel as the Source for crosstalk results.

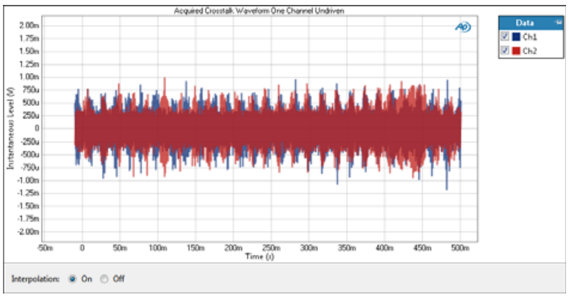
Units

Crosstalk is expressed as a ratio, with the crosstalk in the measured channel divided by the fundamental signal level in that channel. The result is typically stated in decibels.

Units available for Continuous Sweep: Crosstalk, One Channel Undriven are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Continuous Sweep: Acquired Crosstalk Waveform, One Channel Undriven



The Acquired Crosstalk Waveform, One Channel Undriven result is a continuous sweep measurement that displays the Acquired Crosstalk Waveform for each channel. This acquisition is only made when Crosstalk Mode is set to **High Accuracy, One Channel Undriven**. See page 220 for more information about acquired waveform.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Interpolation

For time-domain displays, the APx500 graphs are normally plotted using sinc function interpolation. Interpolation adds points to the displayed trace that do not exist in the measured data, to make data trends more easily visualized. The default in APx is to have Interpolation **On**. However, digital domain signals are sometimes best understood when viewing actual samples, with interpolation switched **Off**. Also see “Interpolation and limit failure markers” on page 581.

Units

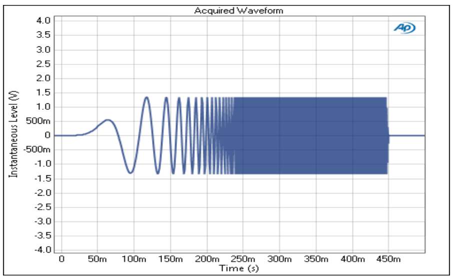
Units available for Acquired Crosstalk Waveform are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• s (seconds)	• V	• D • hex

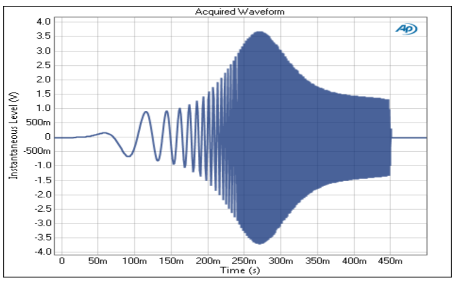
More about Acquired Waveforms

After having passed through the DUT, the continuous sweep waveform is acquired into memory for analysis.

The Acquired Waveform result is a time-domain display of the waveform as acquired.



Acquired Waveform, flat DUT.



Acquired Waveform, EQ’d DUT.

More about Continuous Sweep

The APx Continuous Sweep is an implementation of impulse response measurement methods using log-swept sines. The Continuous Sweep method is used in the APx500 Continuous Sweep, Frequency Response and Acoustic Response measurements.

A continuous sweep provides a brief broadband stimulus signal to the DUT. This stimulus consists of a log-swept sine wave (often called a *log chirp* or a *chirp*) that moves continuously across a specified range of frequencies. The DUT output is acquired by the analyzer and is mathematically processed to provide a number of results.

Advantages

Continuous sweep is very fast and can provide many high-resolution results from a single acquisition. A typical continuous sweep measurement can return 15 separate results, each made up of many thousands of data points across multiple channels through a range of 20 Hz to 20 kHz, in about 4 seconds.

Limitations

1. When Crosstalk Type is set to High Speed, the Continuous Sweep measurement generates log chirps that are delayed from channel to channel, with channel 2 slightly later than 1, channel 3 slightly later than 2, and so on. These staggered sweeps enable the processing algorithms to determine the interchannel crosstalk for frequencies above 1 kHz, all from the same acquisition. In some circumstances, the staggered sweeps can produce unexpected results. See “Continuous Sweep Crosstalk modes” on page 222 and “Solutions for Unexpected Continuous Sweep Crosstalk Results” on page 223.
   Frequency Response and Acoustic Response measurements do not provide crosstalk results, and do not use staggered sweeps.
2. Continuous sweep based measurements must be used in closed loop configuration. Open loop (External Source) configurations are not supported.
3. Total delay through device or system under test must be less than 3 seconds.
4. Sweeps with Stop frequencies less than 1 kHz are not supported. See “Sweep range,” below.
5. Harmonic Distortion is measured using the THD method. This method may produce different results from the traditional THD+N method. See “More about THD” on page 216 and “More about THD+N” on page 475.

Sweep range

In the Audio Precision implementation, the sine can be swept from the system minimum (0.1 Hz for the APx525 family, 2.0 Hz for the APx515, and 5.0 Hz for the APx585 family) to a maximum of 80 kHz. The default range is 20 Hz to 20 kHz.

Start Frequency can be set to any frequency within the system’s range, up to a frequency equal to 1/2 the current Stop Frequency.

Stop Frequency can be set to any frequency within the range of 1 kHz to 80 kHz (depending upon system bandwidth setting).

Sweep duration

The duration of the sweep can be adjusted from a minimum of 50 ms to a maximum of 5 s, with a default of 350 ms. A shorter duration, of course, provides for a faster test: stimulus, acquisition and processing time are all reduced. Longer sweeps provide greater resolution and a better signal to noise ratio. The 350 ms default length is a good compromise between speed and resolution and provides excellent results.

Pre-sweep duration

The total length of the stimulus includes a pre-sweep time in addition to the sweep time. The pre-sweep is an extension of the sweep that begins slightly below the specified start frequency, reaching the start frequency at the end of the pre-sweep time. The amplitude of the pre-sweep is modulated from zero to the sweep level to avoid an initial transient “pop.” Pre-sweep times can be set from 0 s to 1 s, with a default of 100 ms.

Extend Acquisition

The DUT output is acquired, digitized and processed in DSP. The length of the acquisition is usually somewhat longer than the stimulus sweep to allow for time-delayed artifacts created in the DUT. In APx500, the acquisition can be extended up to 3.00 s after end of stimulus. The default is 50 ms.

Input Range

The analyzer inputs must be ranged to provide an optimal acquisition level for a continuous sweep. The APx500 implementation uses autoranging, with the first range set to the lowest level for the best noise performance. If this range is too low, data from the out-of-range sweep are discarded. The input range is moved up and the sweep is repeated. This process may result in several sweep attempts before the correct range is determined; however, since the sweeps are very fast (typically one second) the total acquisition time is short.

If your sweep is a step in an automated sequence that you would like to run as fast as possible, you can optimize the speed of the ranging process by setting the range floor to the correct range for the measurement.

Ranging is set on the Advanced Settings panel (page 223).

Continuous Sweep Distortion Product results

Similar to the meter bar results, the continuous sweep harmonic distortion product results provide great detail about the makeup of the harmonic distortion. For each channel, a selected harmonic product is plotted against frequency. You can choose to view the plot of any harmonic from 2nd to 20th.

In the Distortion Product Level result, the graph displays rms level. In the Distortion Product Ratio result, the graph displays the ratio of the rms level of the distortion product to the rms level of the fundamental tone.

Troubleshooting Unexpected Continuous Sweep Results

Continuous sweep measurements can quickly gather data for many results across multiple channels. However, adverse conditions in the DUT or system under test can produce unexpected results when using Continuous Sweep. These conditions include high distortion (such as clipping), presence of spurious signals (hum, buzz or other signals), or high crosstalk, including crosstalk of distortion products within the DUT.

The solution generally is to configure or adjust the DUT or system under test to minimize these conditions. In some cases, you may have to choose another APx measurement for your test.

Continuous Sweep Crosstalk modes

Overview

APx500 now offers a selection of Crosstalk modes for Continuous Sweep. These modes affect the time-alignment of the sweep stimulus across channels, and the results available for Continuous Sweep Crosstalk, One Channel Driven and Continuous Sweep Crosstalk, One Channel Undriven.

These choices offer an alternative to the original Continuous Sweep crosstalk mode (High Speed), which required that the continuous sweep stimulus signal be delayed across output channels.

Here’s the explanation for this change:

The High Speed crosstalk method relies on starting the sweep at different times, across channels. For example, the sweep in channel 1 starts at T0, the sweep in channel 2 starts at T1, and so on. In analysis, the signals “crosstalking” from other channels can be identified by their time offsets. This method enables the measurement of multichannel crosstalk very quickly, in one sweep cycle.

The time-offset stimulus and the analysis required to separate the crosstalk signals can create confusing artifacts in the results; it returns no results at all below 1 kHz; in some circumstances, it can affect other continuous sweep results. The Crosstalk Type selection feature allows you to use the High Speed method or choose alternatives that do not use time-offset sweep signals. The High Accuracy crosstalk modes provide excellent results, but require multiple sweep cycles and more test time.

Crosstalk Type

None

Sweeps are aligned in time across channels. There is only one sweep cycle, and no crosstalk results are reported.

High speed (default)

Sweeps are offset in time across channels. There is one sweep cycle, and high-speed crosstalk results are reported for both Crosstalk, One Channel Undriven and Crosstalk, One Channel Driven results.

High accuracy, one channel driven

Sweeps are aligned in time across channels. There are n sweep cycles, where n = channel count. Only one channel is driven per sweep cycle. With each sweep cycle, the driven channel is advanced to the next channel, until all channels have been tested. Crosstalk, One Channel Driven are reported.

High accuracy, one channel undriven

Sweeps are aligned in time across channels. There are n sweep cycles, where n = channel count. All channels but one are driven per sweep cycle. With each sweep cycle, the undriven channel is advanced to the next channel, until all channels have been tested. Crosstalk, One Channel Undriven results are reported.

Comparison of measurement results

As mentioned above, a key difference in the crosstalk method is the type of stimulus signal used, whether the log chirp is aligned in time across channels, or offset.

High-speed results

No results are returned below 1 kHz. Results between 2 kHz and the maximum sweep frequency agree well with the high-accuracy results. There is a small artifact at the high-frequency end of the sweep.

High-accuracy results

Results are returned across the entire sweep spectrum.

Noise in continuous sweep crosstalk results

Since crosstalk results are often very low in amplitude, the DUT noise or system noise is a measurable component of a crosstalk result when no bandpass filtering is used. Since bandpass filtering is unavailable for continuous sweep measurements, we recommend using a relatively high stimulus level for continuous sweep crosstalk measurements.

Solutions for Unexpected Continuous Sweep Crosstalk Results

Continuous Sweep High Speed crosstalk method (staggered sweeps across channels)

When Continuous Sweep **Crosstalk Type** is set to **High Speed**, the Continuous Sweep measurement generates log chirps that are delayed from channel to channel, with channel 2 slightly later than 1, channel 3 slightly later than 2, and so on. These “staggered sweeps” enable the analysis software to identify crosstalk signals for all channels within one stimulus/response cycle.

For most testing, this method provides fast, accurate results. However, staggered sweeps and the analysis necessary to interpret the data produce some unexpected conditions and results, which can be complicated by errors in cable connection or DUT characteristics.

- The FFT windowing required to focus on the crosstalk data limits the range of the crosstalk measurements to frequencies above 1 kHz.
- A DUT with different delays across channels will produce erroneous crosstalk results.
- Mis-connection or swapping of channel connections will produce erroneous crosstalk results.
- Similarly, mixing of channels within the DUT will produce erroneous crosstalk results.
- The heuristic required in the crosstalk analysis method creates a measurement floor that is higher than the true instrument floor. This measurement floor is also dependent upon the measurement setup, in particular the relationship between sweep range and sample rate/channel bandwidth. Crosstalk above the measurement floor is correctly measured.

Note: The Frequency Response and Acoustic Response measurements do not provide crosstalk results at all, and thus do not use staggered sweeps.

Alternative crosstalk methods using Continuous Sweep (simultaneous sweeps across channels)

Your measurement requirements or DUT characteristics may make the staggered sweep method undesirable.

The Continuous Sweep measurement provides crosstalk by sequential stimulus/response cycles in these two **Crosstalk Type** selections:

- High accuracy, one channel driven
- High accuracy, one channel undriven

Alternative crosstalk methods using other methods

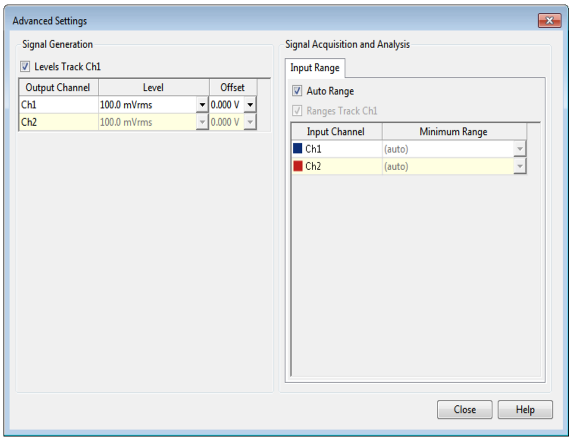
The alternative to using staggered sweeps is to run multiple stimulus/response cycles (one for each channel of interest), with the generator switched to OFF for the channel of interest for each cycle, requiring longer testing times.

For crosstalk meter measurements, you can use the **Crosstalk, One Channel Driven, Crosstalk, One Channel Undriven,** or **Crosstalk, Custom** measurements.

For crosstalk versus frequency stepped sweeps, you can use the **Crosstalk Sweep, One Channel Driven, Crosstalk Sweep, One Channel Undriven, Crosstalk Sweep, Custom** or **Multitone** crosstalk measurements.

Advanced Settings for Continuous Sweep, Frequency Response and Acoustic Response

The default settings here are appropriate for most measurements. You may want to make adjustments for special situations.



Signal Generation

Output levels

If **Track first channel level** is checked (the default), the generator output level values for channel 1 are copied to the other channels, and output levels for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels.

To set individual generator output channel levels, uncheck the **Track first channel level** checkbox and enter values in the output channel Level fields.

Signal Acquisition and Analysis

Input Range

Digital inputs do not require ranging, and Input Range settings are unavailable when inputs are set to a digital format.

Continuous Sweep, Frequency Response and Acoustic Response use a modified autoranging process to find the correct input range.

The sweep acquisition begins in a particular range called the range floor, and moves up from there if necessary. By default, the range floor is the lowest range (0 Vrms to 320 mVrms), but you can specify a higher range by entering the value in the **Min Range** field.

If the initial range is correct for the signal, the acquisition is made and the data are processed and displayed. If this range is too low, data from the out-of-range sweep are discarded. The input range is moved up and the sweep is repeated. This process may result in several sweep attempts before the correct range is determined; however, since the sweeps are very fast (typically one second) the total acquisition time is short.

If your sweep is a step in an automated sequence that you would like to run as fast as possible, you can optimize the speed of the ranging process by setting the range floor to the correct range for the measurement.

29

Crosstalk Sweep, Custom
(Sequence Mode)

Introduction

For the APx515, this measurement requires a software option. See page 166 for more information about software options.

The Crosstalk Sweep, Custom measurement applies a stepped frequency sweep signal to the DUT. You must take steps to prevent the signal from stimulating the channel of interest, as described later in this chapter. The level measured at each frequency point in the undriven channel is compared to the level measured in a driven channel. The ratio is the crosstalk for that channel. X axis is generator frequency; Y axis is crosstalk ratio.

Read more about crosstalk beginning on page 238.

The Crosstalk Sweep, One Channel Driven measurement is not available in External Source or in File Input configurations.

Operation

If you have not yet set up your test, first go to Signal Path Setup (Chapter 6).

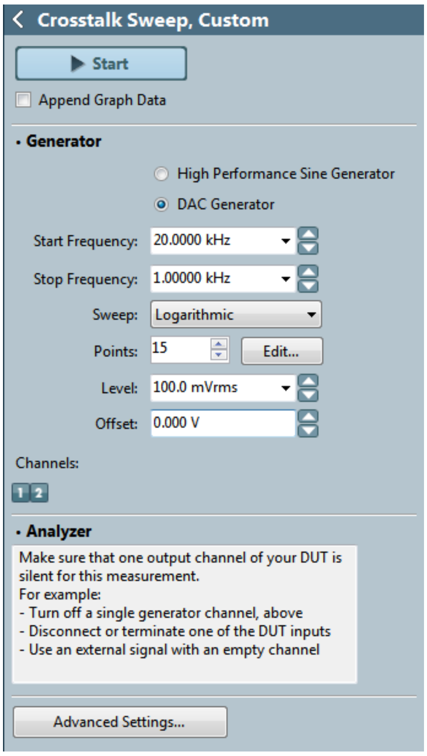
Running the Measurement

The generator outputs the test signal on all channels. You must choose a channel of interest to examine, and remove the generator signal from that channel. The crosstalk (signal leakage) into the channel of interest from all of the remaining channels is measured.

Before you run the sweep, you must prevent the signal from stimulating the channel of interest, as discussed next.

Remove the generator signal from the channel of interest using one of the following methods:

1. Turn OFF a generator channel.
   For many DUTs or systems under test, simply turning a channel of interest **OFF** at the generator will



   provide the most useful Crosstalk Seep, Custom results. Use one of the generator channel selection buttons.
2. Interrupt a channel in the DUT or system under test.
   For some DUTs or systems, interrupting the signal (by switching or disconnection) for the channel of interest may provide more useful results. In the special case of a DUT that has fewer inputs than outputs and where an input is routed within the DUT to more than one output, interrupting the signal within the DUT is the only way to measure crosstalk across some channel combinations.

You can select another channel of interest without turning off the generator. First, reconnect the current channel of interest, then disconnect the new channel of interest. Read the updated results immediately.

For a default sweep of 20 kHz to 1 kHz in 15 logarithmic steps, simply click **Start**. The generator will output the sweep signal to the DUT on all available channels. Next, you must remove the generator signal from the channel of interest (as described above). Read the crosstalk measured in each the remaining channel from the graph display.

Append Graph Data

Measured data are grouped in a **Data Set**, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 573.

Generator

The configuration for this measurement must be the closed-loop configuration, using the APx generator as a stimulus.

See Chapter 5 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 49 for a comparison of features.

Start

Set the sweep **Start Frequency** here.

Stop

Set the sweep **Stop Frequency** here.

Sweep

Choose

- **Logarithmic** point spacing
- **Linear** point spacing
- **Custom**
  Edit sweep spacing and number of **Points** to create a Custom sweep.

Points

Set the number of sweep points here.

Points / Edit

Open the **Sweep Points** dialog to edit import or export the generator **Sweep Points** table. See page 457.

Step Size

For linear sweeps, enter the **Step Size** here.

Level

Set the signal level here. This level may be changed if generator equalization is used, and can be adjusted on a per-channel basis in **Advanced Settings** (see page 457).

Channels

Toggle specific output channels on or off by clicking on the Channel number button.

Analyzer

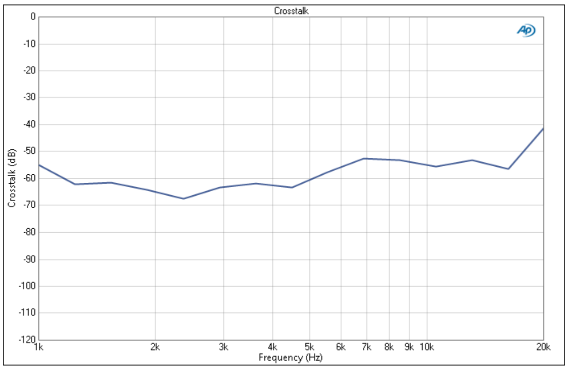
No local low pass, high pass or weighting filters are available for this measurement. However, low pass, high pass and weighting filter settings made in Signal Path Setup > Input/Output will affect this measurement. Read more about APx Filters in Chapter 91.

Advanced Settings

If your test required special adjustments or settings, click **Advanced Settings**. See page 458.

See Chapter 98 for more information about units of measurement.

Crosstalk result



This result shows the crosstalk introduced into Channel 1 from Channel 2, with both channels driven but the Channel 1 connection between the generator and the DUT disconnected.

Units

Units available for Crosstalk Sweep, One Channel Undriven: Crosstalk are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

30

Crosstalk Sweep, One Channel Driven (Sequence Mode)

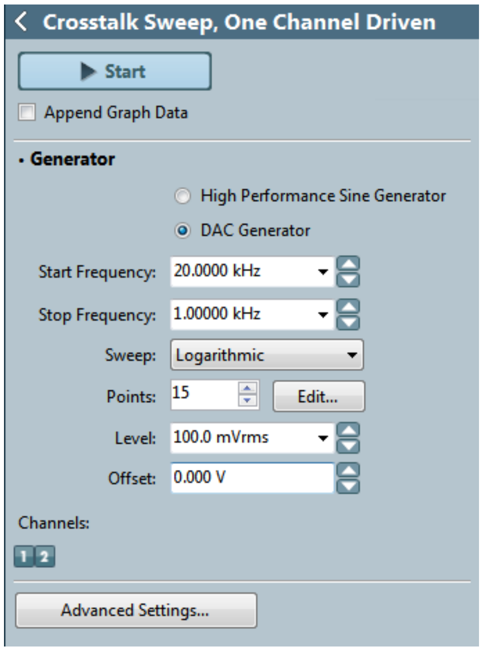
Introduction

For the APx515, this measurement requires a software option. See page 166 for more information about software options.

The **Crosstalk, One Channel Driven** performs a series of stepped frequency sweeps where only one channel is driven at a time. Each sweep applies the stimulus signal to one channel only (initially, Channel 1). Each undriven channel is measured for crosstalk. Then the sweep is run again, with the next channel (Channel 2) driven. This cycle is repeated through the available output channels. The level measured at each frequency point in the undriven channel(s) is compared to the level measured in the driven channel. The ratio of the two is the crosstalk result for that channel. X axis is generator frequency; Y axis is crosstalk ratio.

Read more about crosstalk beginning on page 238.

The Crosstalk Sweep, One Channel Driven measurement is not available in External Source or in File Input configurations.



Operation

If you have not yet set up your test, first go to Signal Path Setup (Chapter 6).

Running the Measurement

For a default sweep of 20 kHz to 1 kHz in 15 logarithmic steps, simply click **Start**. The generator will begin a series of sweeps, driving first Channel 1, then the next channel until the last available channel has been driven. If you want to specify other parameters for your sweep, review **Generator** and **Analyzer** settings, discussed below. Click **Start** when you have approved the settings.

Append Graph Data

Measured data are grouped in a **Data Set**, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 573.

Generator

The configuration for this measurement must be the closed-loop configuration, using the APx generator as a stimulus.

See Chapter 5 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 49 for a comparison of features.

Start

Set the sweep **Start Frequency** here.

Stop

Set the sweep **Stop Frequency** here.

Sweep

Choose

- **Logarithmic** point spacing
- **Linear** point spacing
- **Custom**
  Edit sweep spacing and number of **Points** to create a Custom sweep.

Points

Set the number of sweep points here.

Points / Edit

Open the **Sweep Points** dialog to edit import or export the generator **Sweep Points** table. See page 457.

Step Size

For linear sweeps, enter the **Step Size** here.

Level

Set the signal level here. This level may be changed if generator equalization is used, and can be adjusted on a per-channel basis in **Advanced Settings** (see page 457).

Channels

Toggle specific output channels on or off by clicking on the Channel number button.

Analyzer

No local low pass, high pass or weighting filters are available for this measurement. However, low pass, high pass and weighting filter settings made in Signal Path Setup > Input/Output will affect this measurement. Read more about APx Filters in Chapter 91.

Source

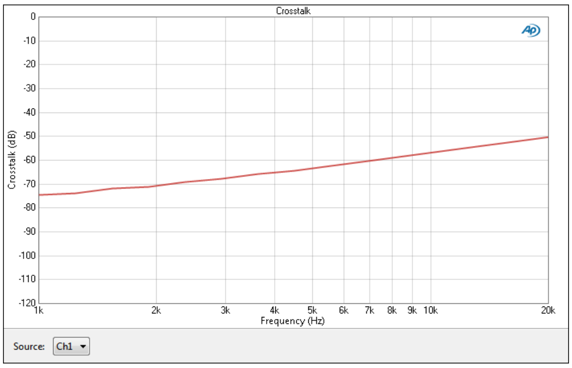
Set the **Source** control beneath the graph in the result panel to view the crosstalk signal(s) introduced into the non-source channel(s).

Advanced Settings

If your test required special adjustments or settings, click **Advanced Settings**. See page 458.

See Chapter 98 for more information about units of measurement.

Crosstalk result



This result shows the crosstalk introduced into Channel 2, while Channel 1 is set as the Source.

Units

Units available for **Crosstalk Sweep, One Channel Driven: Crosstalk ratio** are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

31

Crosstalk Sweep, One Channel Undriven (Sequence Mode)

Introduction

For the APx515, this measurement requires a software option. See page 166 for more information about software options.

The **Crosstalk, One Channel Undriven** measurement performs a series of stepped frequency sweeps where all channels but one are driven. Each sweep applies the stimulus signal to all channels except the undriven channel (initially, Channel 1). The undriven channel is measured for crosstalk. Then the sweep is run again, with the next channel (Channel 2) undriven. This cycle is repeated through the available output channels. The level measured at each frequency point in the undriven channel(s) is compared to the level measured in a driven channel. The ratio of the two is the crosstalk result for that channel. X axis is generator frequency; Y axis is crosstalk ratio.

Read more about crosstalk beginning on page 238.

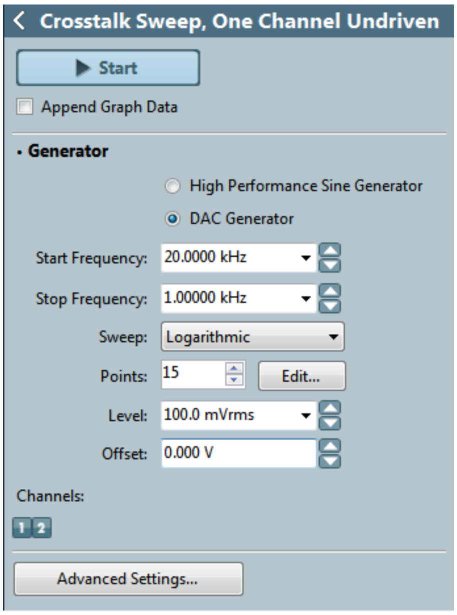
The Crosstalk Sweep, One Channel Driven measurement is not available in External Source or in File Input configurations.

Operation

If you have not yet set up your test, first go to Signal Path Setup (Chapter 6).

Running the Measurement

For a default sweep of 20 kHz to 1 kHz in 15 logarithmic steps, simply click **Start**. The generator will begin a series of sweeps, driving first Channel 1, then the next channel until the last available channel has been driven. If you want to specify other parameters for your sweep, review **Generator** and **Analyzer** settings, discussed below. Click **Start** when you have approved the settings.



Append Graph Data

Measured data are grouped in a **Data Set**, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 573.

Generator

The configuration for this measurement must be the closed-loop configuration, using the APx generator as a stimulus.

See Chapter 5 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 49 for a comparison of features.

Start

Set the sweep **Start Frequency** here.

Stop

Set the sweep **Stop Frequency** here.

Sweep

Choose

- **Logarithmic** point spacing
- **Linear** point spacing
- **Custom**
  Edit sweep spacing and number of **Points** to create a Custom sweep.

Points

Set the number of sweep points here.

Points / Edit

Open the **Sweep Points** dialog to edit import or export the generator **Sweep Points** table. See page 457.

Step Size

For linear sweeps, enter the **Step Size** here.

Level

Set the signal level here. This level may be changed if generator equalization is used, and can be adjusted on a per-channel basis in **Advanced Settings** (see page 457).

Channels

Toggle specific output channels on or off by clicking on the Channel number button.

Analyzer

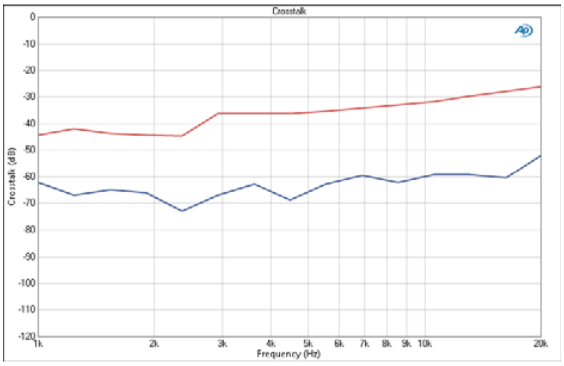
No local low pass, high pass or weighting filters are available for this measurement. However, low pass, high pass and weighting filter settings made in Signal Path Setup > Input/Output will affect this measurement. Read more about APx Filters in Chapter 91.

Advanced Settings

If your test required special adjustments or settings, click **Advanced Settings**. See page 458.

See Chapter 98 for more information about units of measurement.

Crosstalk result



This result shows the crosstalk introduced into Channel 1 from Channel 2, and the crosstalk introduced into Channel 2 from Channel 1.

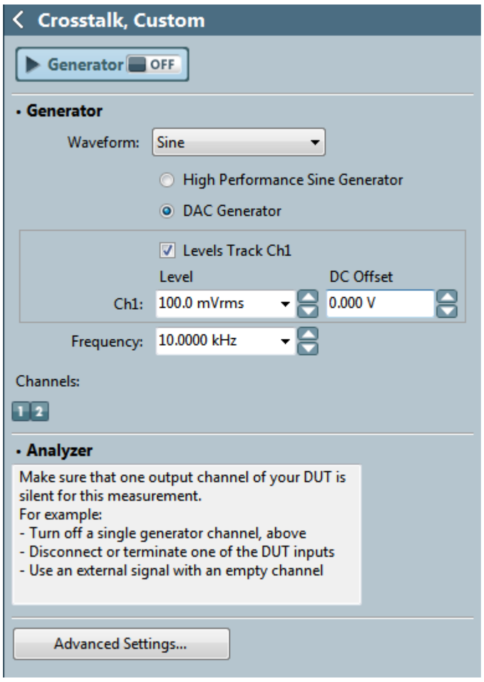
Units

Units available for **Crosstalk Sweep, One Channel Undriven: Crosstalk** are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

32

Crosstalk, Custom
(Sequence Mode)



Crosstalk is unwanted leakage or bleed of a signal from one or more channels to other channels within a device.

The Crosstalk, Custom measurement result provides a measurement of the crosstalk into one undriven DUT channel, while the other channels are driven. For this measurement, the user must manually turn OFF or interrupt the signal to the undriven channel.

More Information

For more information about crosstalk, see page 238. Also see the Crosstalk, One Channel Drive measurement in Chapter 33 and Crosstalk, One Channel Undriven measurement in Chapter 34.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 6 for information about Signal Path Setup.

Generator

The typical configuration for this measurement is the closed-loop configuration, using the APx generator as a stimulus. Alternatively, you can make open-loop measurements, using either the External Source or File Input configuration.

See Chapter 5 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

Be sure that external sources or input/generator files correspond with the Crosstalk “Driven Channel” setting.

By default, the Crosstalk, Custom measurement uses a sine wave at the frequency and level set in the Signal Generation panel as the test signal. Since crosstalk typically increases with frequency, 10 kHz has been selected as the default frequency.

Alternatively, you can select a generator waveform file as the test signal, using the file browser in the Waveform drop-down menu. See Chapter 14 for more information about using Generator Waveform files.

The generator outputs the test signal on all channels. You must choose a channel of interest to examine, and remove the generator signal from that channel. The crosstalk (signal leakage) into the channel of interest from all of the remaining channels is measured.

To measure Crosstalk, Custom, turn the Generator **On**. The generator will output the test signal to the DUT on all channels. Next, you must remove the generator signal form the channel of interest (see below). Read the crosstalk measured in each remaining channel from the meter bar display.

Removing the generator signal from the channel of interest

Remove the generator signal from the channel of interest using one of the following methods:

1. Turn OFF a generator channel.
   For many DUTs or systems under test, simply turning a channel of interest OFF at the generator will provide the most useful Crosstalk, Custom results. Use one of the generator channel selection buttons. When in External Source mode, playing a waveform with the channel of interest silent will provide the same results.
2. Interrupt a channel in the DUT or system under test.
   For some DUTs or systems, interrupting the signal (by switching or disconnection) for the channel of interest may provide more useful results. In the special case of a DUT that has fewer inputs than outputs and where an input is routed within the DUT to more than one output, interrupting the signal within the DUT is the only way to measure crosstalk across some channel combinations.

You can select another channel of interest without turning off the generator. First, reconnect the current channel of interest, then disconnect the new channel of interest. Read the updated results immediately.

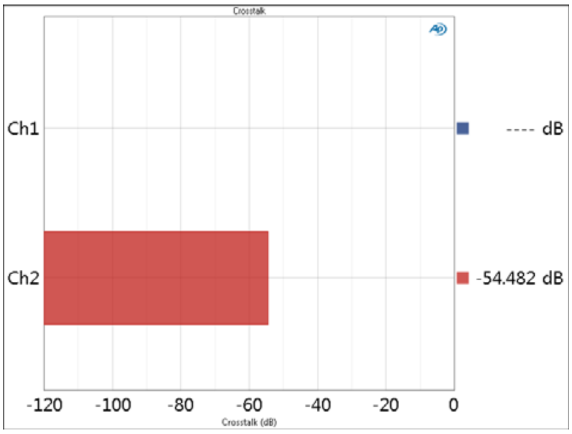
High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 49 for a comparison of features.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, click Advanced Settings. See Advanced Settings on page 317.

Crosstalk



The Crosstalk measurement result provides a single-value measurement crosstalk in the DUT. In this case, the Channel 2 drive signal has been interrupted in the DUT, so crosstalk results only appear in Channel 2.

Crosstalk is usually expressed as a ratio, with the measured crosstalk in the unstimulated channel divided by the full signal level measured in a stimulated channel. The result is typically stated in decibels or as a percentage.

Units

Crosstalk is usually expressed as a ratio, with the measured crosstalk in the unstimulated channel divided by the full signal level measured in a stimulated channel. The result is typically stated in decibels or as a percentage.

Units available for crosstalk are

- x/y
- %
- ppm
- dB

See Chapter 98 for more information about units of measurement.

33

Crosstalk, One Channel Driven
(Sequence Mode)

Crosstalk is unwanted leakage or bleed of a signal from one or more channels to other channels within a device.

The Crosstalk, One Channel Driven measurement result provides a measurement of the crosstalk into the unstimulated DUT channel(s), when one channel is stimulated.

More Information

For more information about crosstalk, see page 238. Also see the Crosstalk, Custom measurement in Chapter 32 and Crosstalk, One Channel Undriven measurement in Chapter 34.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 6 for information about Signal Path Setup.

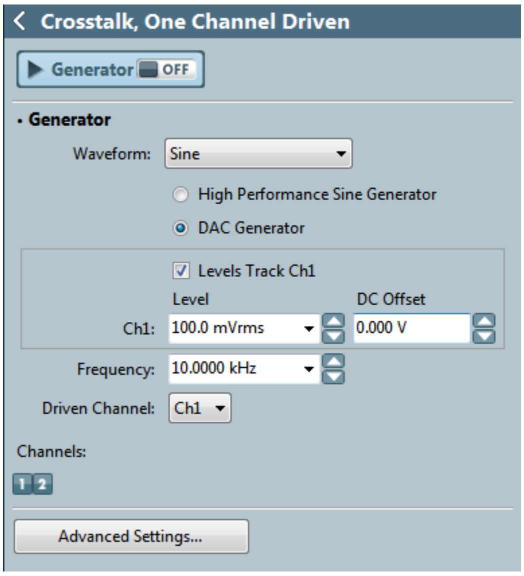
Generator

The typical configuration for this measurement is the closed-loop configuration, using the APx generator as a stimulus. Alternatively, you can make open-loop measurements, using either the External Source or File Input configuration.

See Chapter 5 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

Be sure that external sources or input/generator files correspond with the Crosstalk “Driven Channel” setting.

By default, the Crosstalk, One Channel Driven measurement uses a sine wave at the frequency and level set in the Signal Generation panel as the test signal. Since crosstalk typically increases with frequency, 10 kHz has been selected as the default frequency.



The generator outputs the test signal on one channel only, called the “Driven Channel.” The crosstalk (signal leakage) in each of the remaining channels is measured.

To measure Crosstalk, One Channel Driven, turn the Generator ON. The generator will output the test signal to the DUT on the selected Driven Channel.

Since the Crosstalk, One Channel Driven measurement relies on your choice of a Driven Channel, as described above, deselecting other generator channels can produce ambiguous results. We recommend that for Crosstalk measurements all the generator channel selection buttons remain set to ON, the default condition.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 49 for a comparison of features.

Analyzer

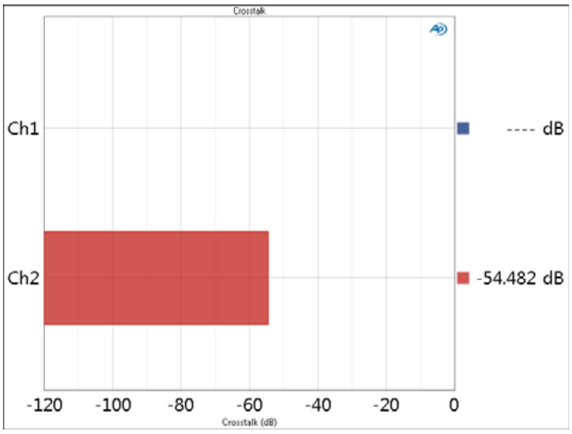
The meter bar display excludes data from the Driven Channel; read the crosstalk measured in each remaining channel from the active meter bars.

You can select another channel as the Driven Channel with the Generator **ON**, and read the updated results immediately.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, click Advanced Settings. See Advanced Settings on page 317.

Crosstalk



The Crosstalk measurement result provides a single-value measurement crosstalk in the DUT. In this case, the Driven Channel is Channel 1, so crosstalk results only appear in Channel 2.

Crosstalk is usually expressed as a ratio, with the measured crosstalk in the unstimulated channel divided by the full signal level measured in a stimulated channel. The result is typically stated in decibels or as a percentage.

Units

Units available for crosstalk are

- x/y
- %
- ppm
- dB

See Chapter 98 for more information about units of measurement.

34

Crosstalk, One Channel Undriven (Sequence Mode)

Crosstalk is unwanted leakage or bleed of a signal from one or more channels to other channels within a device.

The Crosstalk, One Channel Undriven measurement result provides a measurement of the crosstalk into one unstimulated DUT channel (the *undriven* channel), while the other channels are stimulated. This Crosstalk measurement automatically steps the measurement through the channels.

Note: The Crosstalk, One Channel Undriven measurement is not available in External Source configuration.

More Information

For more information about crosstalk, see page 238. Also see the Crosstalk, Custom measurement in Chapter 32 and Crosstalk, One Channel Driven measurement in Chapter 33.

Operation

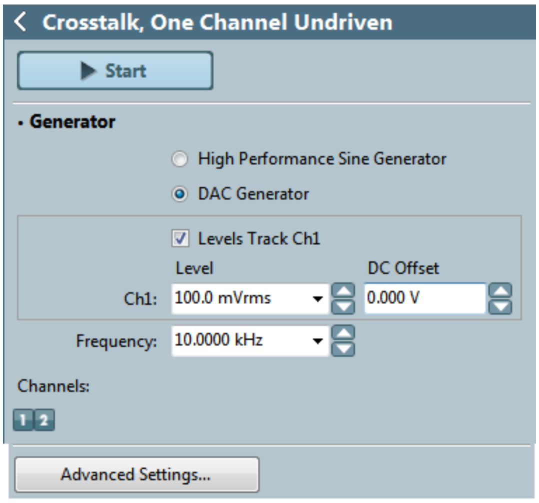
If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 6 for information about Signal Path Setup.

Generator

This measurement must be made in the closed-loop configuration with balanced analog outputs, using the APx generator as a stimulus.

See Chapter 5 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings.

The Crosstalk, One Channel Undriven measurement uses a sine wave at the frequency and level set in the Signal Generation panel as the test signal. Since crosstalk typically increases with frequency, 10 kHz has been selected as the default frequency.



To run Crosstalk, One Channel Undriven, click **Start**. The generator will output the test signal to the DUT on all channels except Channel 1. After the first result is displayed, the measurement runs again with Channel 1 turned **ON**, and Channel 2 **OFF**. This process repeats through all of the available channels. Read the crosstalk measured in each remaining channel from the meter bar display.

Since the Crosstalk, One Channel Undriven measurement relies on stimulating one channel at a time as described above, deselecting generator channel(s) will produce ambiguous results. We recommend that for Crosstalk, One Channel Undriven measurements all generator channel selection buttons remain set to **ON**, the default condition.

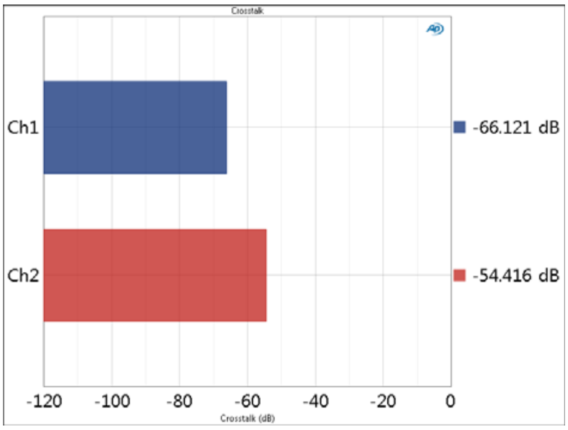
High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 49 for a comparison of features.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, click Advanced Settings. See Advanced Settings on page 317.

Crosstalk



Units

Crosstalk is usually expressed as a ratio, with the measured crosstalk in the unstimulated channel divided by the full signal level measured in a stimulated channel. The result is typically stated in decibels or as a percentage.

Units available for Crosstalk, One Channel Undriven are

- x/y
- %
- ppm
- dB

See Chapter 98 for more information about units of measurement.

More about Crosstalk

Crosstalk is expressed as a ratio. The measured crosstalk level in a channel is divided by the measured level of the stimulus tone in the source channel. Crosstalk is typically stated in decibels (dB).

Crosstalk should normally be measured as a function of frequency, since the values are likely to vary strongly with frequency. The most common circuit mechanisms causing crosstalk are stray capacitive and inductive coupling. If circuit impedances are approximately constant with frequency, crosstalk caused by a single instance of capacitive coupling will increase with increasing frequency at a 6 dB per octave rate. That is, crosstalk due to single pole capacitive coupling will be 6 dB worse at 4 kHz than at 2 kHz, etc. Crosstalk can also be caused by inductive coupling, shared power supplies, shared ground returns, etc., so the relationship is often not the simple capacitive coupling model.

For traditional techniques, crosstalk is measured selectively. That is, a narrow bandpass filter in the analyzer is tuned to the generator frequency, in order to measure crosstalk at or below the wideband noise level. This is not merely an academic concern; the human ear is able to distinguish coherent signals such as sine waves even when the signal amplitude is 10 dB to 20 dB below the wideband noise level.

Crosstalk from “Channel X” to “Channel Y” is often not identical to crosstalk in the other direction, from “Channel Y” into “Channel X.” Circuit layout and complex stray coupling mechanisms often result in somewhat different values for crosstalk in the two directions. Good values for crosstalk in electronic devices range from –50 dB and below, with crosstalk in better than –100 dB achievable by sufficient isolation. When the application is stereo or surround sound, such values are far higher than really necessary for a full effect. However, if the two devices are carrying independent program material, values of crosstalk of –60 dB to –70 dB and more are desirable.

35

DC Level
(Sequence Mode)

The DC Level view provides a single-value result, the DC voltage present at the output of each DUT channel.

Input coupling is forced to DC coupling during this measurement.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 6 for information about Signal Path Setup.

Running the measurement with no stimulus signal

No stimulus is required to measure DC Level. Simply open the measurement and read the DC Level present at the analyzer inputs.

Generator

Running the measurement with a stimulus signal

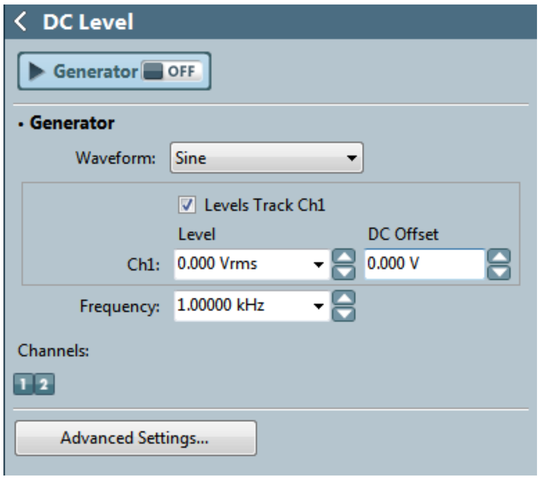
However, you can measure DC Level in the presence of signal, if desired. To measure DC Level in the presence of signal, click the Generator button to **On**. The generator will output the test signal to the DUT on the selected generator channels. Read the DC Level of each channel from the meter bar display.

Selecting the generator waveform

By default, the DC Level measurement uses a sine wave at the frequency and level set in the Signal Generation panel as the test signal. You can add a DC offset to the signal.

Alternatively, you can select a generator waveform file as the test signal, using the file browser in the Waveform drop-down menu.

See Chapter 5 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

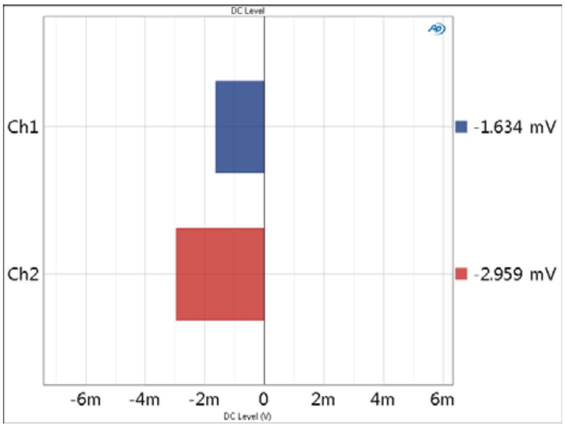


Advanced Settings

Advanced Settings makes available individual channel output level and offset controls, and input ranging and settling controls. See Advanced Settings for single value measurements on page 317.

See Chapter 98 for more information about units of measurement.

DC Level



The DC Level measurement provides a single-value result showing the DC voltage present at the output of each DUT channel.

The analog measurement range for DC Level depends upon the instrument model; for an APx515 it is from approximately ±10 μV to ±120 V. For an APx525 family analyzer it is from approximately ±10 μV to ±160 V. For an APx582, 585 or 586 it is from approximately ±10 μV to ±115 V.

Units

Analog	***Digital***
• V	• D • hex

See Chapter 98 for more information about units of measurement.

More about DC in APx Analyzers

APx can measure DC Level using these measurements:

- DC Level
- Multitone: DC Level
- Signal Analyzer: FFT Spectrum
- Monitor: FFT Spectrum

DC offset

Unless blocked by a capacitor or a transformer-coupled output, analog audio devices will have some (usually very small) dc component in their output. Typical dc levels (often called dc offset) are in the range of –20 mV to +20 mV.

The embedded audio in a digital signal can have a dc component as well. This is usually very low in level, and may vary with the applied dither.

The Constant Value digital signal is “digital dc” of an arbitrary value.

APx can generate sine signals with an arbitrary dc offset, if desired. This facilitates “dc in the presence of signal” measurements.

DC in the presence of signal measurements

APx can generate stimulus signals with AC and DC components, and can measure the dc component when it is an offset in an ac signal.

Coupling: AC or DC

Input signals can be analyzed and displayed in one of two coupling modes:

- AC
  AC coupling blocks any dc signal component from analysis, displaying results for only the ac components. As a consequence, AC coupling introduces a rolloff below 10 Hz. AC coupling is the default setting. For the DC Level measurement only, DC coupling is forced.

Note: Some measurements may require the insertion of a small delay before analysis, to allow for blocking filter settling after a transient (such as generator start).

- DC
  DC coupling includes the dc signal component in analysis and display.

For the DC Level measurement, the APx system is DC coupled regardless of user settings.

Advanced Settings for DC Level measurements

The default settings here are appropriate for most measurements. You may want to make minor adjustments for special situations.

Signal Generation

If **Track first channel level** is checked (the default), the generator output level value (and DC offset level value, if any) for channel 1 is copied to the other channels, and output levels for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels.

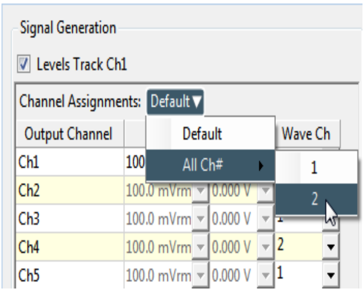
To set individual generator output or DC offset channel levels, uncheck the **Track first channel level** checkbox and enter values in the output channel **Level** or **Offset** fields.

Set Channel Assignments for special waveforms

For Sine + Offset special waveform generation, you can specify the offset by channel. Change individual channel settings in the **Offset** column.

Set Channel Assignments for generator waveforms

For stereo or multichannel generator waveform files, you can set channel mapping. By default, channel 1 audio is mapped to channel 1 output, channel 2 to channel 2, and so on. If the number of channels in the waveform file is less than the number of output channels, the waveform channels resume numbering at 1 and wrap to the next available output channel. These assignments can be remapped by changing individual settings in the **Wave Ch** column, or by selection one of several presets from the **Set Channel Assignments menu.**



File Playback

For generator waveform file playback, you can view the length of the file, and you **can** adjust the playback start position. Select **Seconds** or Samples for the **Length** and **Start Position** units. Enter a new **Start Position** if desired.

Signal Acquisition and Analysis

Input Range tab

Digital inputs do not require ranging, and the Input Range tab is unavailable when inputs are set to a digital format.

If **Auto Range** is **checked** (the default), each analog channel determines its input range automatically, based on the level of the input signal. If the input signal level changes beyond a ranging threshold, **Auto Range** will cause the input ranging circuits to move up or down for proper ranging.

See page 551 for more information about ranging and autoranging.

If **Auto Range** is **unchecked**, you can set a fixed range for each analog input channel. If **Track first channel range** is **checked** (the default), the fixed input range setting for channel 1 is copied to the other channels, and range settings for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels.

To set individual analog input channel ranges, **uncheck** the **Track first channel range** checkbox and enter values in the input channel Range fields.

Delay Time

DC Level results can vary with noise-like characteristics. When a noise measurement (or a DC Level measurement) is run from the navigator or as part of a sequence, the acquisition is delayed for the time set here. This pause can avoid signal disruptions that may occur in the DUT when changing measurements. The **Delay Time** range is 0.0 s to 10.0 s; the default setting is 300.0 ms.

Acquisition Time:

DC Level results can vary with noise-like characteristics. Since noise is not periodic, a very short acquisition may display a result that is well above or below the average noise level. In a noise measurement the signal is acquired over a period of time between 100 ms and 10 s, set in Acquisition Time. Longer acquisitions provide more repeatable results. The default acquisition time is 250 ms.

36

DC Level (DCX)
(Sequence Mode)

DCX measurements require that a DCX-127 Multifunction Module be connected to the APx analyzer system, typically using an Audio Precision USB/APIB adapter connected to the PC.

The DCX-127 brings additional measurement and control features to an Audio Precision analyzer. The APx DC Level (DCX) measurement uses the DCX DMM (Digital Multimeter) mode to implement a precise DC voltage meter. Download the DCX-127 Multifunction Module User's Guide from the Audio Precision Web site at ap.com for complete operational information for the DCX-127.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 6 for information about Signal Path Setup.

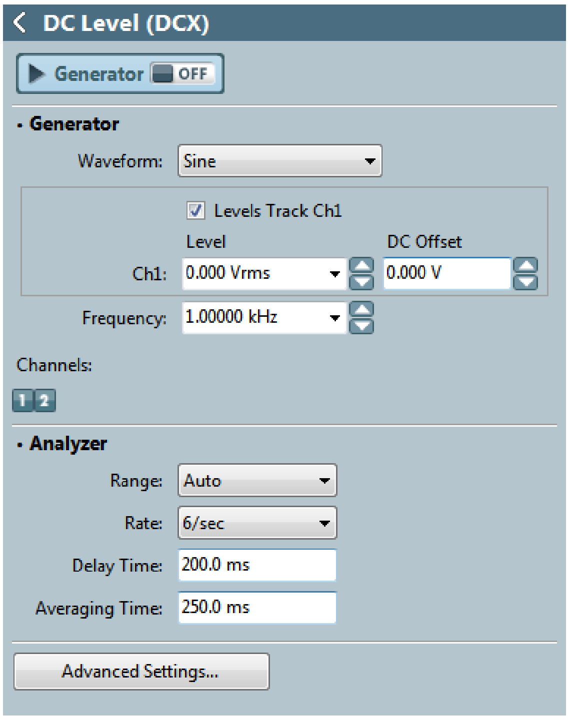
The DCX-127 must be powered on and connected to the measurement PC using an Audio Precision USB/APIB adapter.

Selecting the DC Level (DCX) measurement places the DCX in "DMM" mode. Note that you can configure the DCX DC Outputs or Auxiliary outputs in Signal Path Setup > DCX, independent of any settings or operation in the DC Level (DCX) measurement.

You can make a DC Level measurement with no stimulus signal, or you can use the generator to provide a stimulus to investigate how an audio signal affects the DC Level at the output of a DUT.

Running the measurement with no stimulus signal

No stimulus is required to measure DC Level. Simply open the measurement and read the DC Level present at the analyzer inputs.



Generator

Running the measurement with a stimulus signal

You can also run this measurement in the closed-loop configuration, using the APx generator as a stimulus. See Chapter 5 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

Analyzer

The DCX has a digital multimeter (DMM) that is configured as DC voltmeter for this measurement. The APx analyzer is not used; all the data comes from the DCX.

Range

Select a fixed range for the DC Level measurement, or select **Auto**.

- Auto
- 500 V
- 200 V
- 20 V
- 2 V
- 200 mV

Rate

The Rate control allows you to select one of two meter reading rates.

- **6/sec**
  Approximately 6 meter readings are taken per second.
- **25/sec**
  Approximately 25 meter readings are taken per second.

At 6/sec, DCX meter resolution is a full 4 1/2 digits. At 25/sec, the resolution is reduced. The number of digits displayed is the same but the least significant digit is always 5 or 0.

Delay Time

A Delay Time interval is started when the DC Level (DCX) measurement is started within a sequence. APx ignores DCX data until the time set in Delay Time has passed. Minimum Delay Time is 200 ms, maximum is 10 s, 200 ms is the default.

Averaging Time

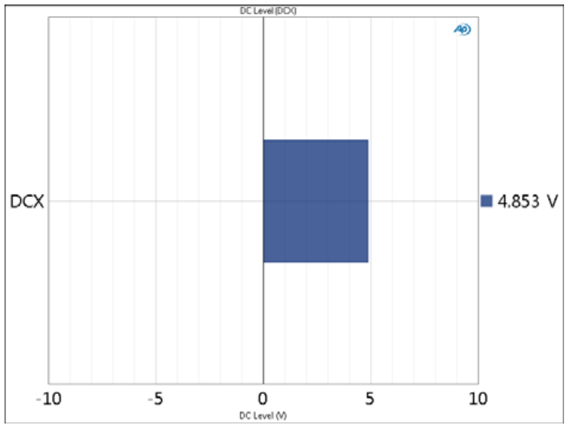
APx performs a rolling average of the DCX readings across a specified time before display. Minimum Averaging Time is 200 ms, maximum is 10 s, 250 ms is the default.

Advanced Settings

Advanced Settings makes available individual channel output level and offset controls, and input ranging and settling controls. See Advanced Settings for single value measurements on page 317.

See Chapter 98 for more information about units of measurement.

DC Level (DCX)



Units

Unit available for DC Level (DCX) is

- V

37

DC Level Sweep
(Sequence Mode)

For the APx515, this measurement requires a software option key. See page 166 for more information about software options.

The DC Level Sweep measurement uses a DC stimulus that is moved across a range of values in a series of points. The DUT output is acquired by the analyzer. DC Level and Noise results are plotted. The Noise result can be filtered.

The DC Level Sweep measurement is not available in External Source or File Input configurations.

Input coupling is forced to DC Coupling during this measurement.

DC Level Sweep results available in APx500 are:

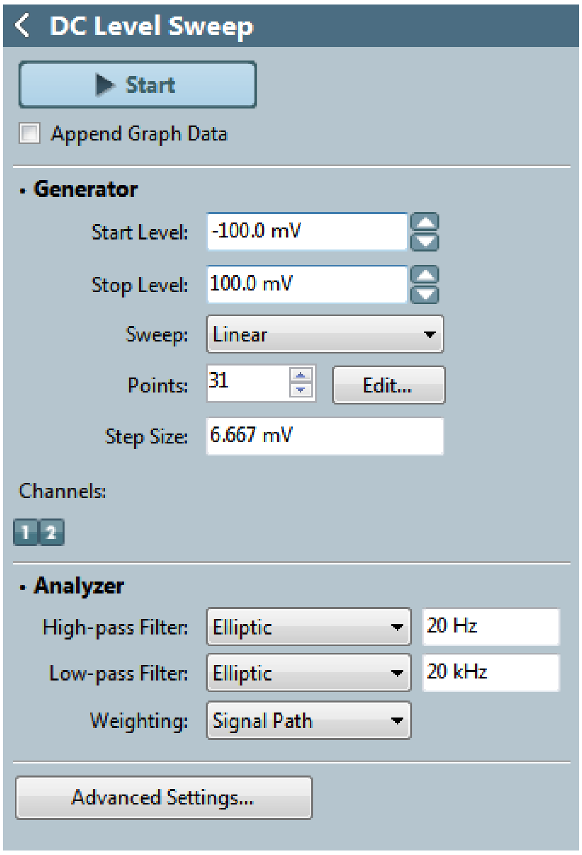
- DC Level
- Noise

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 6 for information about Signal Path Setup.

The DC Level Sweep measurement does not use an audio signal for a stimulus, but instead sweeps a DC signal across a range. For analog outputs, this is a DC voltage; for digital outputs, it is digital DC (constant value).

To make a DC Level Sweep measurement, click **Start**. The generator will output the DC signal to the DUT on the selected generator channels. The signal will be swept from the **Start Level** to the **Stop Level** in the set number of points. You can select **Logarithmic** or **Linear** point spacing, or choose **Edit...** to customize the **Sweep Points** table.



Append Graph Data

Normally, the current graph data for all results is deleted each time you start a new measurement. If the **Append** box is checked, the current data is kept in memory as a **Data Set** (see page 165), and the new measurement data is appended as a new data set. You can append many **Data Sets**.

Generator

Start Level, Stop Level and Sweep Points

The sweep moves between two levels, set in the **Start Level** and **Stop Level** fields. The default levels are –100.0 mV and +100.0 mV.

The sweep moves in a specified number of step points, set in the **Points** field. The minimum is 2 points; maximum is 65,535. The default setting is 31.

The sweep point spacing is set by selecting one of the following choices in the Sweep field:

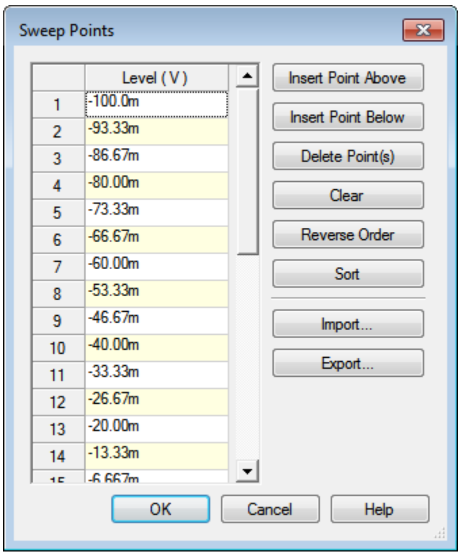
- **Logarithmic**; use the **Points** field to set the number of logarithmically spaced points;
- **Linear** (the default), which provides two methods of adjusting spacing: the **Points** field or the **Step Size** field; or
- **Custom**. Click **Edit** to open the **Sweep Points** dialog, where you can set points arbitrarily, or load or save sweep table files.

Viewing or Editing the Sweep Points table

You can view or edit the current generator sweep points at any time.

Click **Edit** to open the generator **Sweep Points** table. The table shows each sweep point and its corresponding level. You can edit this table to add or delete points, or to change the level of a point. Points can be sorted or reversed in order using the controls on the right.

A **Sweep Points** table can be saved as a *.csv file or as a Microsoft Excel *.xls file. A compatible *.csv or *.xls file can be opened and used as a **Sweep Points** table.



A Linear sweep can include a range that extends into negative values, but a Logarithmic sweep can only include positive values.

Each point is displayed as it is acquired. In a short period of time, the entire sweep is processed, and the small graphs in the Selector will be populated with results. Simply click the result view you want in the Navigator, or choose the graph thumbnail result view in the Selector.

Analyzer

Filters

Local high-pass, low-pass and weighting filters can be applied to the **Noise Level** result. The **DC Level** result is unfiltered.

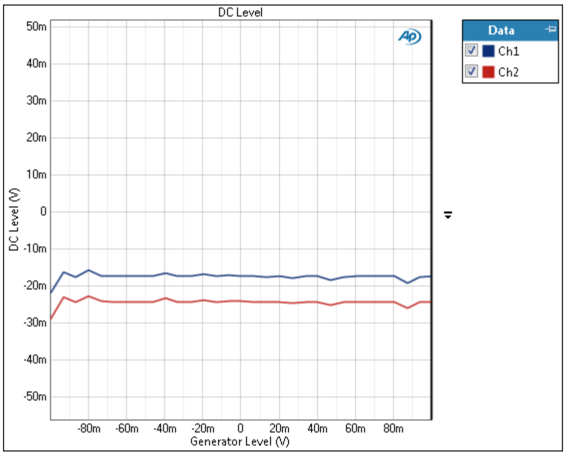
Noise measurements are typically made within a limited passband. The default setting for **Noise Level** measurements uses a 20 kHz low-pass filter and a 20 Hz high-pass filter. See Measurement Filters on page 547 for detailed information about the filters available locally for some measurements.

Low pass, high pass and weighting Signal Path Filters settings (see page 545) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass or high-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active. A local weighting filter setting does not override any Signal Path filter settings; the local weighting filter is applied in addition to any active Signal Path filters.

Advanced Settings

If you'd like to adjust analyzer ranging or settling parameters, click **Advanced Settings**. Read more about Advanced Settings for DC measurements on page 240.

DC Level Result



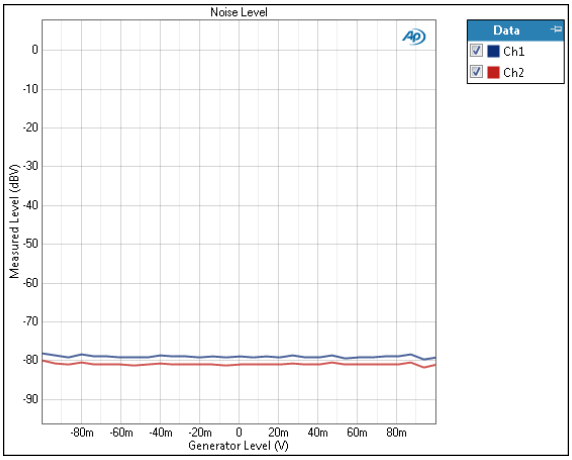
The DC Level Sweep measurement uses a DC stimulus that is moved across a range of values in a series of points. The DUT output is acquired by the analyzer. This view shows DC Level result.

Units

Units available for DC Level Sweep: DC Level are:

X-axis (analog signals)	*Y-axis (analog signals)*
• V	• V
X-axis (digital signals)	***Y-axis (digital signals)***
• D • hex	• D • hex

Noise Level Result



The DC Level Sweep measurement uses a DC stimulus that is moved across a range of values in a series of points. The DUT output is acquired by the analyzer. This view shows the Noise result. The Noise result can be filtered.

Units

Units available for DC Level Sweep: Noise Level are:

X-axis (analog signals)	*Y-axis (analog signals)*
• V	• Vrms • dBV • dBu • dBrA • dBrB • dBSPL1 • sBSPL2 • dBm • W (watts)
X-axis (digital signals)	***Y-axis (digital signals)***
• D • hex	• dBFS • FS • %FS • dBrA • dBrB • dBSPL1 • dBSPL2

38

Digital Error Rate
(Sequence Mode)

Introduction

For the APx515, this measurement requires a software option key. See page 166 for more information about software options.

The Digital Error Rate measurements uses a bit-accurate technique to provide results showing both the number of errors and the error rate in a digital audio system.

For meaningful results, an Audio Precision Bit test waveform should be used. Walking Ones, Walking Zeros and Constant waveforms can also be used.

Go to page 251 for more information about bit-accurate measurements and Bit test, Walking Zeros, Walking Ones and Constant Value waveforms.

Digital Error Rate measurements are not available in analog input configurations.

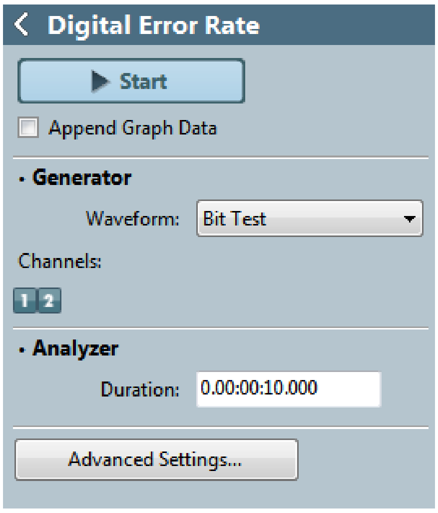
Digital Error Rate measurements available in APx500 are:

- Average Error Rate
- Instantaneous Error Rate
- Total Errors
- Cumulative Errors

Operation

Append Graph Data

Measured data are grouped in a Data Set, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox and a Notes field for each data set. Read more about Data Sets beginning on page 573.



Running the measurement

Click **Start**. The generator will output the test signal to the DUT on the selected generator channels. The acquisition will begin and the graph will display the error rate as the signal is acquired. The acquisition will continue until the time set in **Duration**.

Generator

Selecting the generator waveform

By default, the Average Error Rate measurement uses the Bittest Random waveform (see page 48) at a fixed level.

Alternatively, you can select a generator waveform file as the test signal, using the file browser in the Waveform drop-down menu. See Chapter 14 for more information about using Generator Waveform files. For useful results, generator waveforms for Digital Error Rate measurements must be Audio Precision bit-accurate waveforms.

Analyzer

For Digital Error Rate, the analyzer automatically determines whether or not the acquired signal is a compatible bit-accurate signal. If so, the signal is analyzed for error rate, regardless of the current generator setting. Compatible bit-accurate signals are

- Audio Precision Bit test
- Constant Value
- Walking ones
- Walking zeros

Note: Constant Value signals of all ones (1s) or all zeros (0s) are treated as invalid Constant Value signals, and will return errors. This implementation serves to flag digital connection errors, which can incorrectly produce a stream of constant ones or zeros. When using Constant Value signals, set the signal to any value except ones or zeros.

Duration

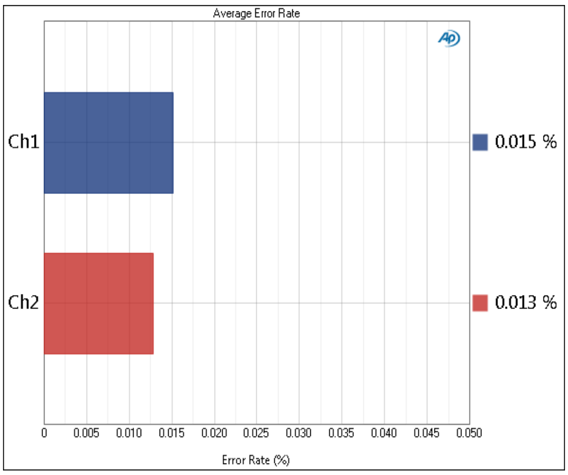
Duration sets the length of the sample acquisition. Minimum time is 0.1 s, maximum time is 7 days.

Enter the Duration following this pattern:

d:hh:mm:ss.s

where d = days, hh = hours, mm = minutes, ss = seconds. Days and fractional seconds are optional.

Average Error Rate measurement



The Digital Error Rate: Average Error Rate measurement result provides a single-value measurement of the average audio bit error rate. The average error rate is calculated by dividing the errors received by the total number of samples received.

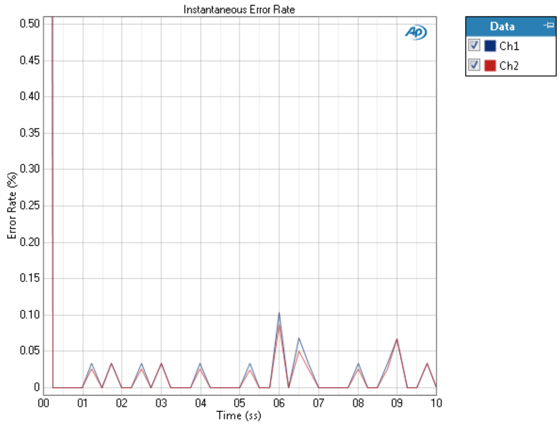
Units

Units available for Average Error Rate are

X-axis

- %
  Error percentage.
- Err/S
  Errors per sample, expressed in exponential notation (also called scientific notation).

Instantaneous Error Rate measurement



The Digital Error Rate: Instantaneous Error Rate measurement result provides a record of the rate of digital errors versus elapsed time.

Units

Units available for Instantaneous Error Rate are

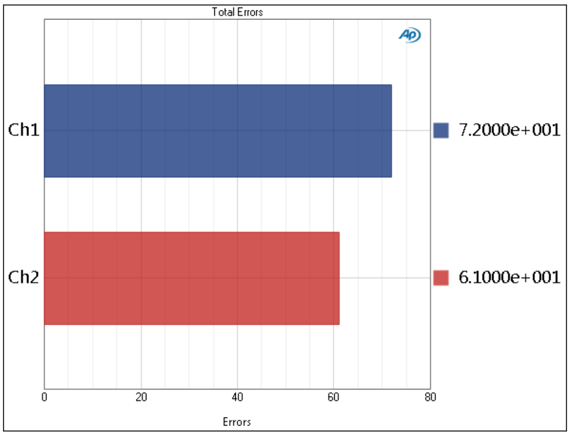
X-axis

- s (seconds)

Y-axis

- %
  Error percentage.
- Err/S
  Errors per sample, expressed in exponential notation (also called scientific notation).

Total Errors measurement



The Digital Error Rate: Total Errors measurement result provides a single-value measurement of the total digital errors acquired.

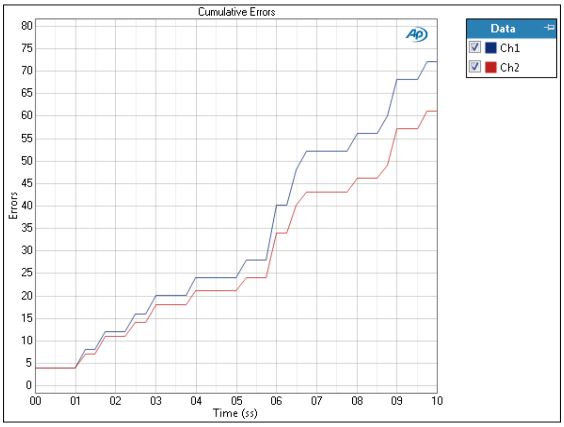
Units

Units available for Total Errors are

X-axis

- The X-axis is unitless. The total number of errors is displayed, using exponential notation (also called scientific notation).

Cumulative Errors measurement



The Digital Error Rate: Cumulative Errors result provides a record of the accumulation of digital errors versus elapsed time.

Units

Units available for Cumulative Errors are

X-axis

- s (seconds)

Y-axis

- %
  Error percentage.
- Err/S
  Errors per sample, expressed in exponential notation.

Generator Units

Generator units available for Constant Value are

- hex
- D

More about Bit-accurate measurements

Bit-accurate measurements are digital domain measurements that use test signals in which the value of every audio sample is known. In analysis, each sample that does not have the correct value is counted as an error. Measurement results may be total errors, error rate, and so on. A digital device or system that can pass such a waveform with no errors is said to be exhibiting bit-accurate performance.

APx500 generates and analyzes four signals used for bit-accurate measurements:

- Bit test
- Walking Zeros
- Walking Ones
- Constant Value

Read more about these waveforms on page 48.

39

DIM IMD
(Sequence Mode)

Note: DIM IMD measurements are supported only in the APx555 or in instruments fitted with the AG52 option, when in analog output configurations. Digital output configurations or External Source configuration are not supported. See page 5 for information about the AG52 Option and page 6 for information about the APx555.

DIM stands for Dynamic Intermodulation Distortion. DIM is a technique for measuring IMD under rapidly-changing dynamic conditions, which typically stress analog power amplifiers.

For more information about the DIM method, see page 255. For information about IMD in general, see page 273.

The DIM IMD measurement results available in APx500 are:

- DIM Ratio
- DIM Distortion Product Ratio

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 6 for information about Signal Path Setup.

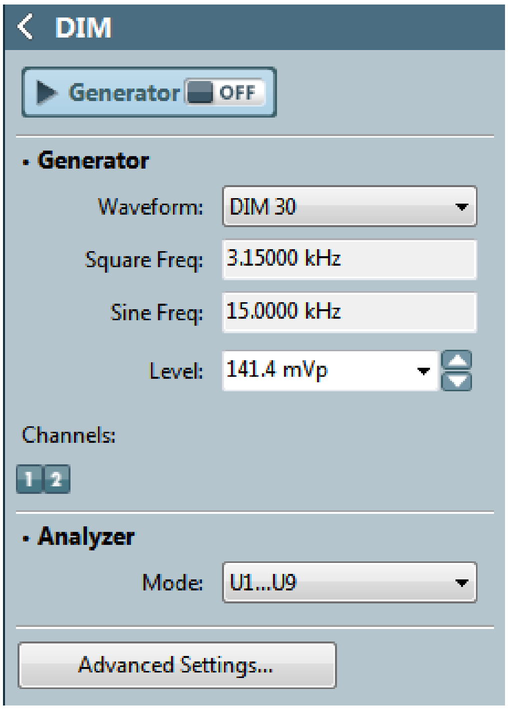
Generator

This measurement must be performed in a closed-loop analog-output configuration, using the APx generator as a stimulus. See Chapter 5 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings.

Selecting the DIM generator waveform

The DIM Ratio measurement uses a DIM test waveform at the level set in the Generator panel.

The DIM test waveforms are as follows:



- DIM 30
  The DIM 30 signal is the linear sum of a square wave of 3.15 kHz and a sine wave of 15.00 kHz. The square wave is filtered by a single-pole low-pass filter at 30 kHz.
- DIM 100
  The DIM 100 signal is the linear sum of a square wave of 3.15 kHz and a sine wave of 15.00 kHz. The square wave is filtered by a single-pole low-pass filter at 100 kHz.
- DIM B
  The DIM B (broadcast) signal is the linear sum of a square wave of 2.96 kHz and a sine wave of 14.00 kHz. The square wave is filtered by a single-pole low-pass filter at 30 kHz.

- DIM B8
  The DIM B8 signal is the linear sum of a square wave of 2.96 kHz and a sine wave of 8.00 kHz. The square wave is filtered by a single-pole low-pass filter at 30 kHz.

In all cases, the ratio of the peak voltage of the square wave to the peak voltage of the sine wave is 4:1.

Analyzer

No local low pass, high pass or weighting filters are available for this measurement. However, low pass, high pass and weighting filter settings made in Signal Path Setup > Input/Output will affect this measurement.

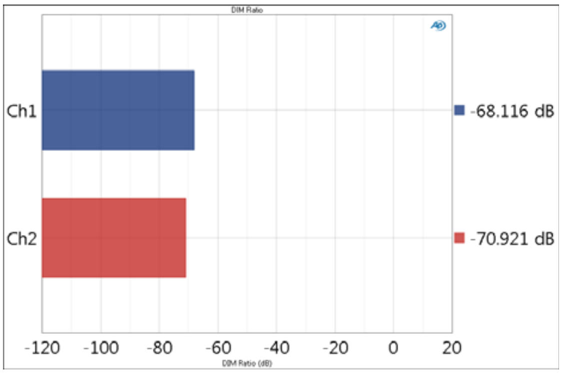
Mode

- U1...U9
  The IEC 60268-3 standard defines the DIM measurement, listing nine specific IM components and identifying the corresponding output voltages as U1, U2, U3, U4, U5, U6, U7, U8 and U9. The U1...U9 selection is the APx500 default, and displays the rms summation of all nine voltages expressed as a ratio to the amplitude of the test signal sine wave component.
- U4+U5
  This mode includes only the two IM components that fall below the test signal square wave frequency, U4 and U5.
- Emulation
  Audio Precision System One, System Two, Cascade and 2700 series instruments measured DIM differently, using a combination of analog filters and rms meters. Choose **Emulation** to view DIM results compatible with results from earlier Audio Precision instruments.

Advanced Settings

If your test requires special settings, click **Advanced Settings**. Advanced Settings for IMD measurements include individual channel generator levels, custom IMD frequencies, individual channel analyzer ranging and custom analyzer settling. See Advanced Settings for IMD on page 274. See Chapter 98 for more information about units of measurement.

DIM Ratio results



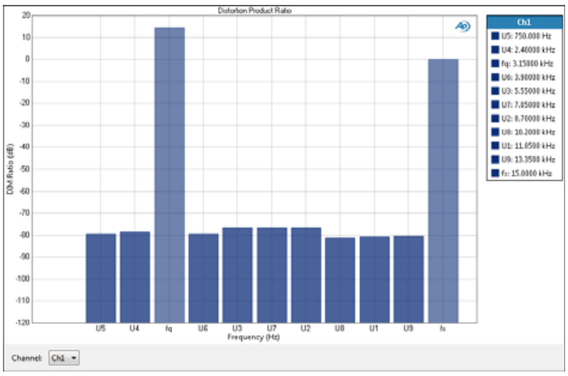
The DIM Ratio result provides single-value measurement results of the IMD (intermodulation distortion) ratio in the output signal from each DUT channel.

Units

Units available for DIM Ratio are

- x/y
- %
- ppm
- dB

DIM Distortion Product Ratio results



The DIM Distortion Product Ratio result provides measurement results of the IMD (intermodulation distortion) ratio in the output signal from each DUT channel.

Read the DIM Distortion Product Ratio for the selected channel from the meter bar display. The fundamental square wave fq, the fundamental sine wave fs and the distortion products U1 through U9 (as chosen by the Mode selector) will be shown in the meter bar display. The graph legend shows the frequency of each distortion product.

Result Settings: Channel

This setting is made in the Result Settings bar, beneath the graph display.

The Distortion Product Ratio result shows only one channel at a time. Select the channel to be viewed from the Channel drop-down list.

Units

Units available for DIM Distortion Product Ratio are

- x/y
- %
- ppm
- dB

See Chapter 98 for more information about units of measurement.

More About DIM IMD

DIM stands for Dynamic Intermodulation Distortion. It is a technique to measure the nonlinearity of a device, designed to be particularly sensitive to distortions produced during transient conditions typical of program material. DIM measurements typically use a square wave of about 3 kHz summed with a lower-amplitude sine wave of 14 kHz to 15 kHz. The DIM measurement is defined in the standard IEC 60268-3, sec. 14.12.9.

DIM is also referred to as TIM—Transient Intermodulation Distortion.

For information about IMD in general, see page 273.

40

DIM Level Sweep
(Sequence Mode)

Note: DIM IMD measurements are supported only in the APx555 or in instruments fitted with the AG52 option, when in analog output configurations. Digital output configurations or External Source configuration are not supported. See page 5 for information about the AG52 Option and page 6 for information about the APx555.

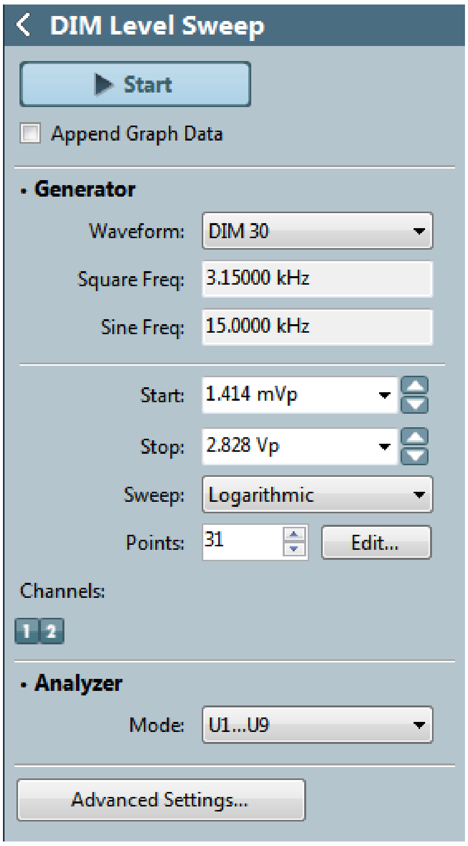
DIM stands for Dynamic Intermodulation Distortion. DIM is a technique for measuring IMD under rapidly-changing dynamic conditions, which typically stress analog power amplifiers.

The DIM Level Sweep measurement provides a DIM stimulus signal (square wave mixed with sine wave, as described on pages 253 and 255) that is moved across a range of levels in a series of points. The DUT output is acquired by the analyzer and processed for display.

For more information about the DIM method, see page 255. For information about IMD in general, see page 273.

DIM Level Sweep results available in APx500 are:

- DIM Ratio
- DIM Ratio vs. Measured Level



Operation

Append Graph Data

Normally, the current graph data for all results is deleted each time you start a new measurement. If the **Append** box is checked, the current data is kept in memory as a **Data Set** (see page 165), and the new measurement data is appended as a new data set. You can append many **Data Sets.**

If you have not yet set up your test, first go to Signal Path Setup (Chapter 6).

Generator

This signal will be swept from the **Start Level** to the **Stop Level** in the set number of **Points.**

Level Settings for DIM IMD

The DIM Level Sweep measurement uses a DIM test waveform at the level set in the Signal Generation panel as the test signal.

The DIM test waveforms are as follows:

- **DIM 30**
  The DIM 30 signal is the linear sum of a square

wave of 3.15 kHz and a sine wave of 15.00 kHz. The square wave is filtered by a single-pole low-pass filter at 30 kHz.

- DIM 100
  The DIM 100 signal is the linear sum of a square wave of 3.15 kHz and a sine wave of 15.00 kHz. The square wave is filtered by a single-pole low-pass filter at 100 kHz.
- DIM B
  The DIM B (broadcast) signal is the linear sum of a square wave of 2.96 kHz and a sine wave of 14.00 kHz. The square wave is filtered by a single-pole low-pass filter at 30 kHz.
- DIM B8
  The DIM B8 signal is the linear sum of a square wave of 2.96 kHz and a sine wave of 8.00 kHz. The square wave is filtered by a single-pole low-pass filter at 30 kHz.

In all cases, the ratio of the peak voltage of the square wave to the peak voltage of the sine wave is 4:1.

Start Level, Stop Level and Sweep Points

The sweep moves between two levels, set in the **Start Level** and **Stop Level** fields. The default levels are 1.000 mVrms and 2.000 Vrms.

The sweep moves in a specified number of step points, set in the **Points** field. The minimum is 2 points; maximum is 65,535. The default setting is 31.

The sweep point spacing is set by selecting one of the following choices in the Sweep field:

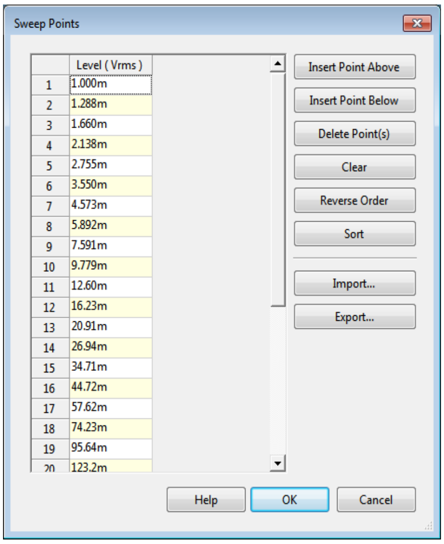
- **Logarithmic** (the default); use the **Points** field to set the number of logarithmically spaced points;
- **Linear**, which provides two methods of adjusting spacing: the **Points** field or the **Step Size** field; or
- **Custom**. Click **Edit** to open the **Sweep Points** dialog, where you can set points arbitrarily, or load or save sweep table files.

Viewing or Editing the Sweep Points table

You can view or edit the current sweep points at any time.

Click **Edit** to open the **Sweep Points** table. The table shows each sweep point and its corresponding level. You can edit this table to add or delete points, or to change the level of a point. Points can be sorted or reversed in order using the controls on the right.

A **Sweep Points** table can be saved as a *.csv file or as a Microsoft Excel *.xls file. A compatible *.csv or *.xls file can be opened and used as a **Sweep Points** table.



Analyzer

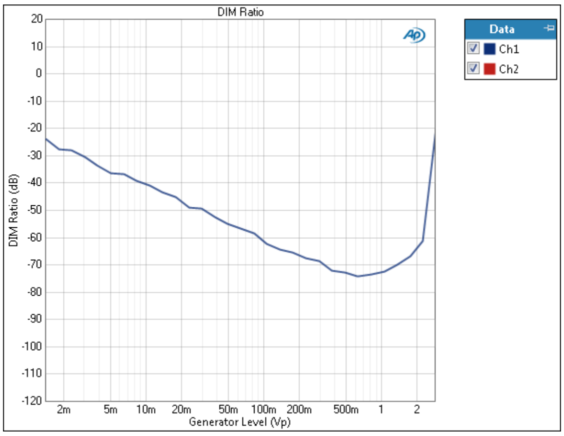
Mode

- U1...U9
  The IEC 60268-3 standard defines the DIM measurement, listing nine specific IM components and identifying the corresponding output voltages as U1, U2, U3, U4, U5, U6, U7, U8 and U9. The **U1...U9** selection is the APx500 default, and displays the rms summation of all nine voltages expressed as a ratio to the amplitude of the test signal sine wave component.
- U4+U5
  This mode includes only the two IM components that fall below the test signal square wave frequency, U4 and U5.
- Emulation
  Audio Precision System One, System Two, Cascade and 2700 series instruments measured DIM differently, using a combination of analog filters and rms meters. Choose Emulation to view DIM results compatible with results from earlier Audio Precision instruments.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, click **Advanced Settings**. See Advanced Settings for Stepped Sweeps on page 458. See Chapter 98 for more information about units of measurement.

DIM Ratio (vs. Generator Level)



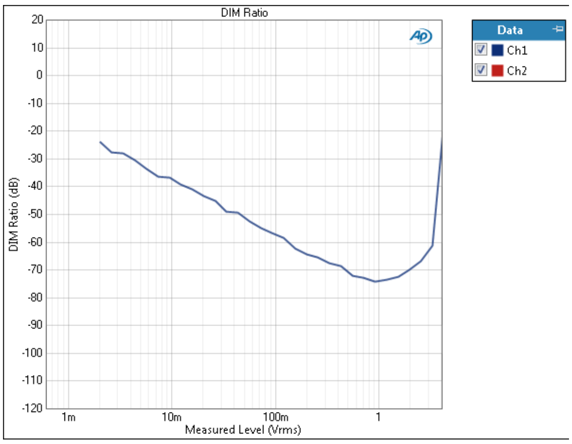
DIM Ratio plots the distortion ratio on the Y axis against the Generator level on the X axis.

Units

Units available for the DIM Ratio vs. Generator Level result are

X-axis	*Y-axis*
• Vrms	• x/y
• Vp	• %
• Vpp	• ppm
• dBV	• dB
• dBu	
• dBrG	
• dBm	
• W (watts)	

DIM Ratio vs. Measured Level



DIM Ratio vs. Measured Level plots the distortion ratio on the Y axis against the DUT output level on the X axis.

Units

Units available for the DIM Ratio vs. Measured Level result are

X-axis (analog)	*X-axis (digital)*	*Y-axis*
• Vrms	• FS	• x/y
• Vp	• %FS	• %
• Vpp	• dBFS	• ppm
• dBV	• dBrA	• dB
• dBu	• dBrB	
• dBrG	• dBSPL1	
• dBm	• dBSPL2	
• W (watts)		

See Chapter 98 for more information about units of measurement.

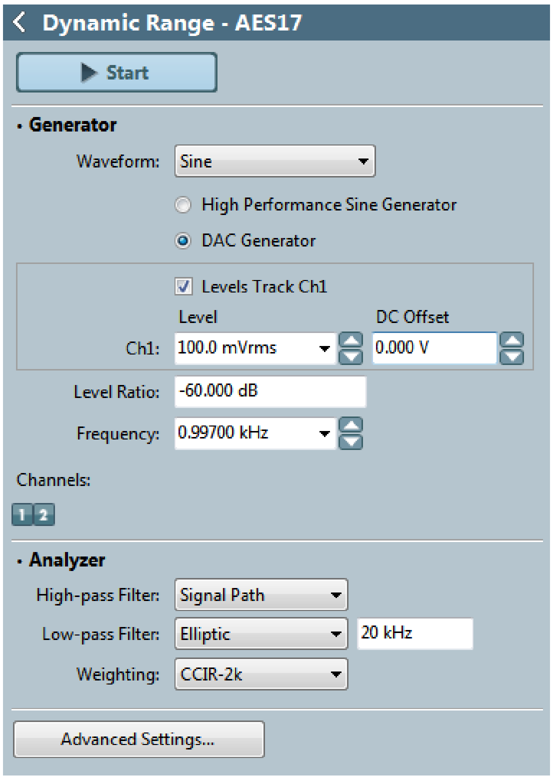
41

Dynamic Range (AES17) (Sequence Mode)

Application

For the APx515, this measurement requires a software option key. See page 166 for more information about software options.

We recommend using this AES17 dynamic range measurement for digital converters. When testing other devices, we recommend using the Signal-to-Noise Measurement described in Chapter 74.



Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 6 for information about Signal Path Setup.

Running the measurement

To measure Dynamic Range, click **Start**. The generator is turned ON briefly at the specified full level on the selected generator channels, and a measurement is made. Then the generator is set to the lower level (set in Level Ratio), the signal is notched out, and a second measurement is made. The ratio between the two results is computed for each channel and displayed as meter bars.

Generator

The typical configuration for this measurement is the closed-loop configuration, using the APx generator as a stimulus. Alternatively, you can make open-loop measurements, using either the External Source or File Input configuration.

See Chapter 5 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 49 for a comparison of features.

Level

For dynamic range, you must choose a generator level that represents the maximum level for the device under test (DUT).

DAC testing

For DACs, the maximum level is usually 0 dBFS.

ADC testing

For ADCs, it is usually the analog signal that just produces 0 dBFS at the ADC output. You can set the analog generator to produce a target digital output level in Reference Levels, and then set that generator level as dBrG. Return to the Dynamic Range measurement and choose 0 dBrG as the Generator level.

Level Ratio

AES17 recommends the lower signal be set to 60 dB below the full level signal (the APx500 default). Other ratios can be selected.

Running the measurement in External Source configuration (Open Loop)

When configured for External Source, use a signal source that varies between your desired maximum value, and your desired minimum value (usually 60 dB below the maximum). APx500 uses an algorithm that monitors the incoming signal, waiting for readings that indicate a level change of at least 20 dB. When this can be reliably determined, a tuned notch filter is applied to the lower level acquisition and the ratio between the two levels is displayed as the Dynamic Range result.

The Audio Precision utility APxWfmGenerator.exe (available from www.ap.com) provides a suitable waveform called “Dynamic Range (1 kHz)”. This waveform is a 0.99700 kHz tone at a level of 0 dBFS, alternating with periods of 0.99700 kHz at –60 dBFS.

Analyzer

Filters

Local high-pass, low-pass and weighting filters are available for the lower (typically -60 dB) acquisition of this measurement. A low-pass filter set just above the audio passband (at 20 kHz) is recommended for noise measurements. This is the APx default.

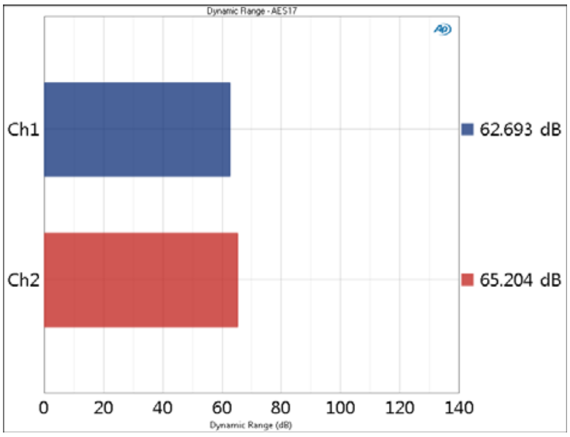
AES17 recommends the CCIR-2k weighting filter for this measurement; this is the APx default. See Measurement Filters on page 547 for detailed information about the filters available locally for some measurements.

Low pass, high pass and weighting Signal Path Filters settings (see page 545) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass or high-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active. A local weighting filter setting does not override any Signal Path filter settings; the local weighting filter is applied in addition to any active Signal Path filters.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, click **Advanced Settings**. See Advanced Settings for single value measurement on page 317.

Dynamic Range results



The Dynamic Range (AES17) measurement provides a single-value result showing the AES17 Dynamic Range for each DUT channel.

Units

Units available for Dynamic Range are

- x/y
- dB

See Chapter 98 for more information about units of measurement.

More about Dynamic Range

Overview

Dynamic Range is an expression of the ratio of the largest signal a device can pass to the device’s noise floor. “Largest signal” usually refers to a signal at a specified degree of distortion, often 1%. Signal-to-Noise Ratio and Dynamic Range are essentially the same measurement, except that the signal in SNR is arbitrary (and should be stated in the results), and the signal in Dynamic Range is at the maximum (details of which should also be stated in the results).

For digital converter measurements...

We recommend using this AES17 dynamic range measurement. It is intended specifically for ADC (analog-to-digital converter) and DAC (digital-to-analog converter) dynamic range and “noise in the presence of signal” measurements, as described in Section 9.3 of AES17. A similar method is defined in IEC61606.

This method differs from standard signal-to-noise and dynamic range measurements in that it uses a –60 dBFS stimulus during the noise measurement. This method is used for two reasons:

In both ADCs and DACs, “idle tones” can be produced within the converter in the absence of applied signal. In the method here, a low-level tone is applied to the converter to avoid production of idle channel noise. The low-level tone is removed by a notch filter before measurement.

In some DACs, the output of the device is switched off when there is no signal, providing an unrealistically quiet measurement. The low-level tone (again, notched out before measurement) defeats this muting mechanism.

At –60 dBFS, the tone is so low that any distortion products created are below the noise floor.

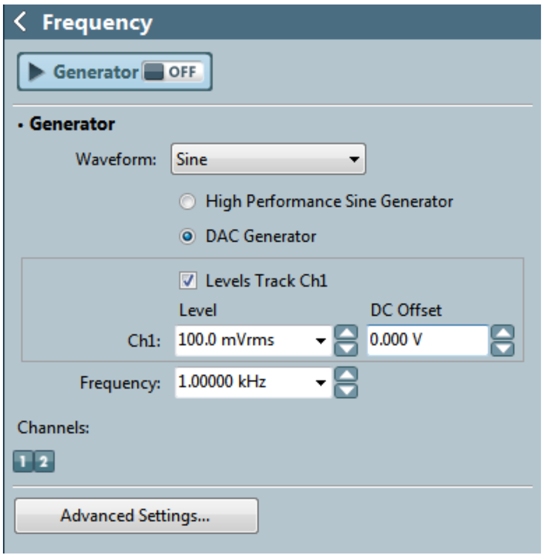
For dynamic range measurements of other devices...

We recommend using the Signal-to-Noise measurement configured for dynamic range when testing other devices. It is described in Chapter 74. The low-level signal is not required for non-converter devices.

42

Frequency
(Sequence Mode)

The Frequency measurement provides a single-value result showing the frequency of the strongest component in the output signal of each DUT channel, as measured at each analyzer input.



Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 6 for information about Signal Path Setup.

Generator

The typical configuration for this measurement is the closed-loop configuration, using the APx generator as a stimulus. Alternatively, you can make open-loop measurements, using either the External Source or File Input configuration.

See Chapter 5 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 49 for a comparison of features.

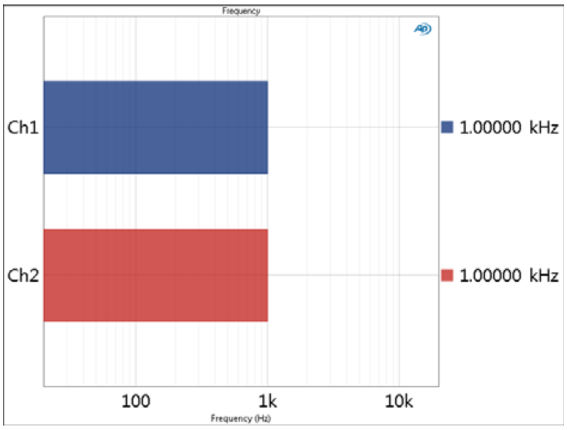
Filters

There are no local filters available for this measurement. However, low pass, high pass and weighting filter settings made in Signal Path Setup > Input/Output will affect this measurement.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, click Advanced Settings. See Advanced Settings for single value measurements on page 52.

Frequency Measurement



Units

Units available for Frequency are

- Hz
- dHz
- F/R
- %Hz

See Chapter 98 for more information about units of measurement.

43

Frequency Response
(Sequence Mode)

Frequency Response is a continuous sweep measurement that displays the frequency response of each channel, plotted against frequency. A broadband single value deviation result is also computed from the continuous sweep acquisition.

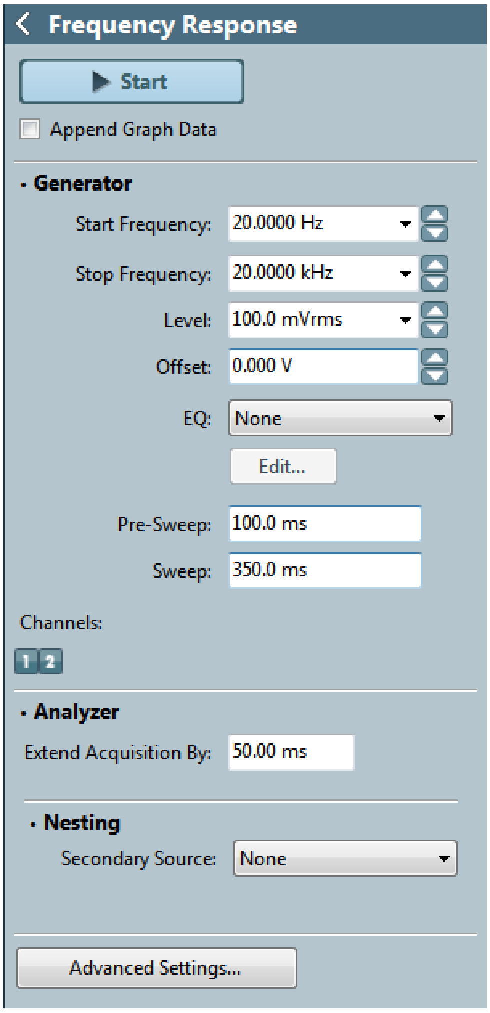
A continuous sweep is a brief log-swept sine wave (a Farina log chirp) that moves continuously across a specified range of frequencies. The DUT output is acquired by the analyzer and is mathematically processed to provide a number of results. For more information about the continuous sweep measurements used in Frequency Response, go to page 220.

Continuous sweep based measurements, including Frequency Response, are not supported in External Source or File Input configurations.

Frequency Response results available in APx500 are:

- Level
- Gain
- Relative Level
- Deviation

All of these results are available from a single acquisition.



Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 6 for information about Signal Path Setup.

Append Graph Data

Measured data are grouped in a **Data Set**, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 573.

Generator

The Frequency Response measurement uses a continuous sweep (log-swept sine chirp) stimulus signal, swept between the frequencies entered in Start Frequency and Stop Frequency, at the value set in Level.

Running the Measurement

To use Frequency Response, click Start. The generator will output the test signal to the DUT on the selected generator channels. Read the results in the selected result view.

Start Frequency and Stop Frequency

These controls set the frequency range of the sweep. See page 220 for more about continuous sweep ranges, which are dependent upon the instrument.

Level

Set the generator level here.

Offset

Set any DC offset to the generator signal here.

EQ

You can equalize the generator signal to make a pre-emphasized stimulus. See Generator Equalization (page 170) for a general discussion of this feature.

Choosing generator equalization

Select None, Relative or Absolute from the EQ drop-down menu.

- None applies no equalization to the generator.
- Relative applies the gain or loss specified in the EQ table to the current generator level for each channel.
- Absolute sets the generator level for all channels to the level specified in the EQ table. Note: when Absolute is selected, current channel level settings are lost.

Edit

If you have chosen to equalize the generator signal, you can edit the EQ table here. See Edit EQ Table (page 170).

Pre-Sweep and Sweep duration fields

A continuous sweep stimulus consists of a sine wave swept logarithmically across the sweep range in the period of time set in these fields.

To enable the DUT to stabilize before the beginning of the defined sweep measurement, there is variable Pre-Sweep time. The actual sweep stimulus signal begins below the defined Start Frequency and is swept upward, reaching the Start Frequency at the end of the Pre-Sweep time. The sum of the times in Pre-Sweep and Sweep is the total sweep length.

- The default for Pre-Sweep is 100.0 ms. Minimum is 0 s; maximum is 1.0 s.
- The default for Sweep is 350.0 ms. Minimum is 50.0 ms; maximum is 5 s.

Longer Pre-Sweep settings allow more time for DUT stabilization, but increase measurement time. Longer Sweep settings provide greater resolution and signal-to-noise ratio, but increase measurement time.

Analyzer

The sweep is acquired and processed to provide the various results.

There are no local measurement filters available for continuous sweep measurements. However, low pass, high pass and weighting input filter settings made in Signal Path Setup > Input/Output > Filters will also affect this measurement. See Chapter 91 for more information about filtering.

Extend Acquisition By

By default, a continuous sweep measurement makes an acquisition slightly longer than the stimulus, to include possible time-delayed artifacts created in the DUT. By default, the acquisition is extended 50 ms longer than the stimulus.

In some cases, you may want to extend the acquisition further. Enter a new value in the **Extend Acquisition By:** field. Minimum extension is 0 s; maximum is 3 s.

Nesting

Frequency Response sweeps can be nested. Read about Nested Sweeps beginning on page 161.

Secondary Source

Select the secondary sweep source here. This is the outside, nesting parameter that is varied to change the conditions for the primary sweep through a number of iterations.

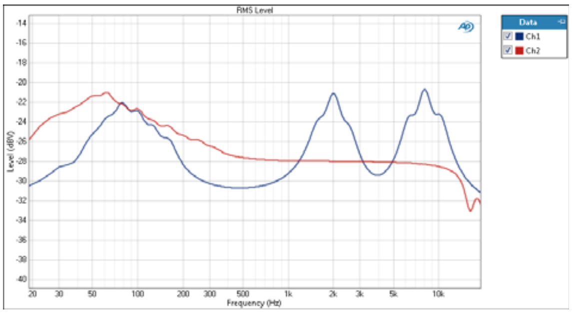
Available choices are dependent upon instrument hardware, options, Input/Output settings and Bench Mode Generator settings. Read about Nested Sweeps beginning on page 161.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging, see Advanced Settings for Continuous Sweep on page 223.

See Chapter 98 for more information about units of measurement.

Frequency Response: Level



The Frequency Response: Level result is a continuous sweep measurement that provides a graphical display of the frequency response of each channel. In this result the DUT output level is plotted against frequency.

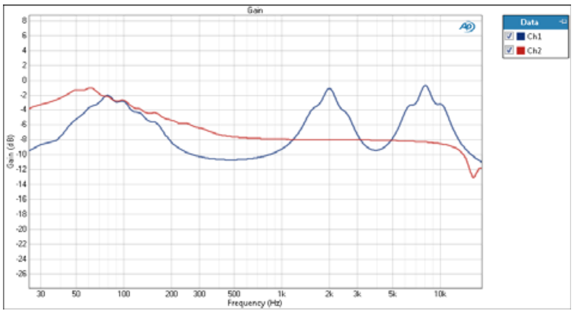
If the settings are identical, this will provide the same results as the Continuous Sweep: Level result.

Units

Units available for Frequency Response: Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz	• Vrms	• dBFS
• dHz	• dBV	• FS
• F/R	• dBu	• %FS
• %Hz	• dBrA	• dBrA
	• dBrB	• dBrB
	• dBSPL1	• dBSPL1
	• dBSPL2	• dBSPL2
	• dBm	
	• W (watts)	

Frequency Response: Gain



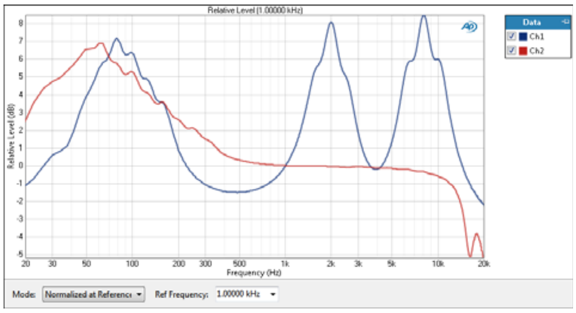
The Frequency Response: Gain result is a continuous sweep measurement that provides a graphical display of the frequency response of each channel. In this result the DUT gain is plotted against frequency.

Units

Units available for Frequency Response: Gain are:

X-axis	*Y-axis same-domain*	*Y-axis cross-domain*
• Hz	• x/y	• FS/Vrms
• dHz	• %	• dB(FS/Vrms)
• F/R	• ppm	—or—
• %Hz	• dB	• Vrms/FS
		• dB(Vrms/FS)

Frequency Response: Relative Level



The Frequency Response: Relative Level result is a continuous sweep measurement that provides a graphical display of the frequency response of each channel. In this result the DUT output level is plotted against frequency, relative to the level at a selected frequency. This enables you to specify the frequency that will be set as 0 dB and view the response in relation to that frequency. A midband frequency such as 1 kHz is often set as the reference frequency, and is the default here.

If the settings are identical, this will provide the same results as the Continuous Sweep: Relative Level result.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Mode

Normalized at Reference

Enter a reference frequency for normalization in the Ref Frequency field. The measured level at the frequency you choose is set as 0 dB on the graph, and the DUT response is plotted in relation to this reference. You can change the Ref Frequency at any time (except after appending) and the graph will immediately be redrawn to reflect the new setting.

Auto-Centered in Limits

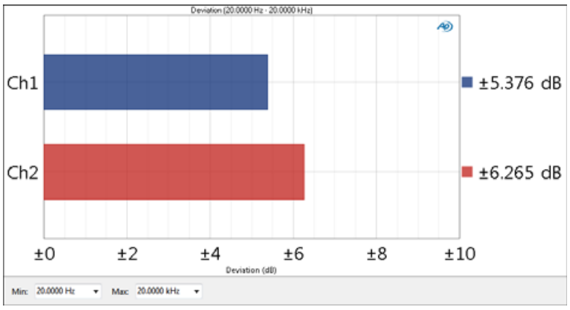
This control is only usable if upper and lower limits are set on this graph. In some applications, the general conformance of a trace to the shape of a set of limits is of importance, with absolute level less important or irrelevant. Auto-Centered in Limits centers the data trace relative to the upper and lower limits. To exclude possible exceptional data points at frequency extremes, the frequency range that is considered for the centering operation is constrained by the values in the **Min** and **Max** fields.

Units

Units available for Frequency Response: Relative Level are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Frequency Response: Deviation



Frequency Response: Deviation is a single value result computed from the continuous sweep acquisition. In this result the frequency deviation (the total range of frequency variation) of each channel is displayed as a meter bar. You can specify a minimum and maximum frequency to define the range to be considered in the deviation measurement.

To measure Frequency Response: Deviation, first enter a frequency range in the **Min Frequency** and **Max Frequency** fields below the graph; or, accept the default 20 Hz to 20 kHz range.

You can change the **Deviation Min Frequency** or **Max Frequency** settings at any time and the meter bars will immediately redraw to reflect the new settings.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Min

Set the minimum frequency of the range of interest here.

Max

Set the maximum frequency of the range of interest here.

Units

Units available for Frequency Response: Deviation are

• dB

44

IMD
(Sequence Mode)

Introduction

For the APx515, these measurements require a software option key. See page 166 for more information about software options.

IMD (intermodulation distortion) measurements use two tones of different frequencies summed into a stimulus signal. This measurement provides single-value IMD ratio and IMD distortion product results.

The IMD measurement provides stimulus tones and analysis for four important methods of measuring intermodulation distortion: SMPTE, DFD, CCIF and MOD.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 6 for information about Signal Path Setup.

Measurement

IMD Type

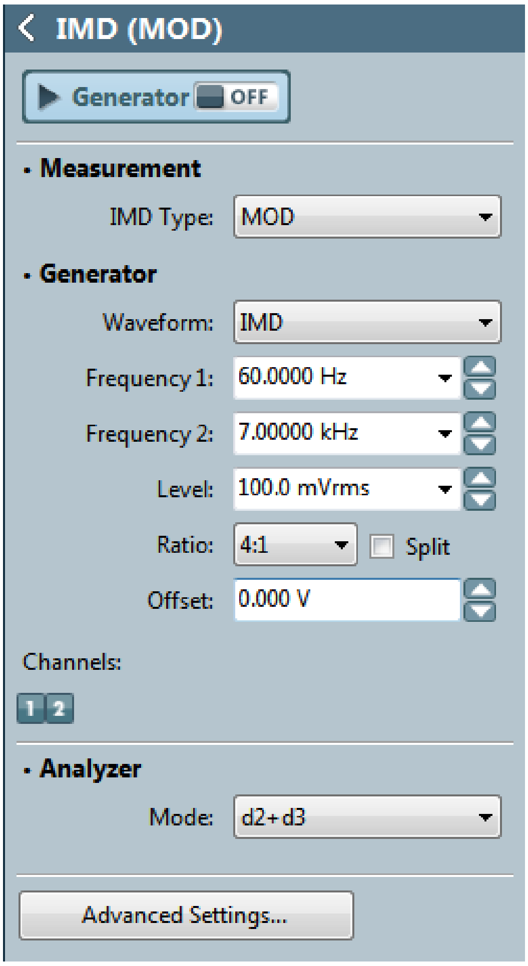
Select the IMD type here. Read more about IMD beginning on page 273 for descriptions of the IMD types and stimulus frequencies.

- SMPTE
- DFD
- CCIF
- MOD

Generator

Waveform

The typical configuration for this measurement is the closed-loop configuration, using the APx generator as



a stimulus. Alternatively, you can make open-loop measurements, using either the External Source or File Input configuration.

See Chapter 5 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

If you have selected None (External Source) in Signal Path Setup Output Configuration, the Signal Generation controls listed below will be not be shown. For generator waveform files or external source use, the selected waveform file must be a correctly formed IMD stimulus corresponding to the selected IMD type. See Chapter 5, External Source, page 150 for information about Stimulus Signal Resources, and page 163 for information about Generator Waveforms.

Frequency

Set the IMD tone to be generated here.

- For **SMPTE** or **MOD**, set **Frequency 1** or **Frequency 2**.
  These measurements use dual-tone stimulus, with a lower, dominant frequency (**Frequency 1**) and a higher frequency of interest (**Frequency 2**). In a nonlinear device, the tones modulate each other. You can choose different frequency values here. The SMPTE and MOD stimulus frequency constraints are listed in the More on IMD discussion beginning on page 273.
- For **DFD** or **CCIF**, set **Mean Frequency** or **Diff Frequency**.
  DFD and **CCIF** use a dual-tone stimulus, with two high frequency tones separated by a **Difference Frequency**, centered around a **Mean Frequency**. You can choose different frequency values here. The DFD and CCIF stimulus frequency constraints are listed in the More on IMD discussion beginning on page 273.

Level

Set the Generator Level here.

Ratio

For **SMPTE**, select a **1:1** or **4:1** ratio (level ratio of **Freq 1** to **Freq 2**). For **MOD**, select **1:1, 4:1** or **10:1**.

Split

The **Split** setting defeats the summing of the two IMD frequencies within the generator. Instead, the signals are "split" and routed independently to channel outputs. **Frequency 1** or the **Mean Frequency** is routed to odd-numbered channels, and **Frequency 2** or the **Diff Frequency** is routed to even-numbered channels.

Acoustic testing of microphones sometimes uses two stimulus loudspeakers, each with a single frequency component of an IMD stimulus. The stimulus signals are summed acoustically at the microphone element. This method provides a distortion result for the microphone while minimizing the distortion contributions of the loudspeakers.

Offset

Set any DC offset here.

Analyzer

Filters

No local low pass, high pass or weighting filters are available for this measurement. However, low pass, high pass and weighting filter settings made in Signal Path Setup > Input/Output will affect this measurement. Read more about APx Filters in Chapter 91.

Mode

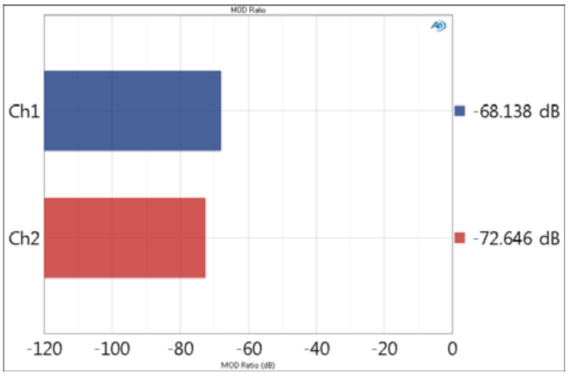
The Mode selector is not available for the SMPTE IMD measurement.

For **DFD**, **CCIF** and **MOD** sweeps, the **Mode** selector allows you to choose the distortion products or combinations of distortion products to be displayed. The default is d2 + d3 (2nd order plus 3rd order products).

Advanced Settings

If your test requires special settings, click **Advanced Settings**. Advanced Settings for IMD measurements include individual channel generator levels, custom IMD frequencies, individual channel analyzer ranging and custom analyzer settling. See Advanced Settings for IMD on page 274.

IMD Ratio



IMD Ratio provides a single-value result showing the IMD (intermodulation distortion) ratio in the output signal from each DUT channel, using the SMPTE, DFD, CCIF or MOD method.

Units

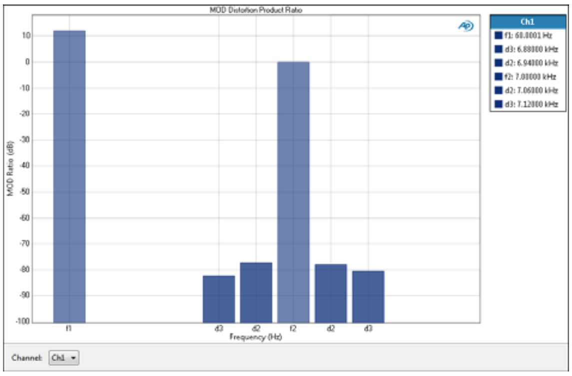
Units available for IMD Ratio are

Y-axis

- x/y
- %
- ppm
- dB

See Chapter 98 for more information about units of measurement.

IMD Distortion Product Ratio



IMD Distortion Product Ratio provides a single-value result that shows the individual second, third, fourth and fifth-order modulation products (d2 through d5) as a ratio to the level of the high frequency stimulus tone (f2), for a selected channel.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Channel

The Distortion Product Ratio result shows only one channel at a time. Select the channel to be viewed from the Channel drop-down list.

Units

Units available for IMD Distortion Product Ratio are

X-axis	***Y-axis***
• Hz	• Stimulus and distortion product tone bars
• dHz	
• F/R	
• %Hz	

See Chapter 98 for more information about units of measurement.

More about Intermodulation Distortion

IMD is an abbreviation for Intermodulation Distortion. IMD is created when two or more audio tones beat with one another (intermodulate) in a non-linear device to produce undesired new tones. The primary mechanism producing IM in most devices is AM (amplitude modulation), which creates sidebands that are at the sum and difference of the frequencies of the original audio tones.

For example, if a 2 kHz audio tone and a 7 kHz audio tone pass through a non-linear device and undergo AM intermodulation distortion, the output signal will contain new tones (modulation products) at 9 kHz (the sum of the tones) and at 5 kHz (the difference of the tones).

The modulation products may also beat with each other and with the original audio signal, creating more modulation products.

Measurements that have two tones in the stimulus are used to measure IMD. IMD measurements in APx500 include DFD, DIM, MOD, SMPTE and CCIF.

SMPTE IMD

SMPTE IMD is a technique for measuring IMD (intermodulation distortion) according to the SMPTE RP120-1983 standard. The DIN intermodulation distortion technique uses a similar method.

The stimulus is a strong low-frequency interfering signal (F1) combined with a weaker high frequency signal of interest (F2). F1 is usually 60 Hz and F2 is usually 7 kHz, at a ratio of F1:F2 = 4:1. The stimulus signal is the sum of the two sine waves. In a distorting DUT, this stimulus results in an AM (amplitude modulated) waveform, with F2 as the “carrier” and F1 as the modulation.

In analysis, F1 is removed, and the residual is bandpass filtered and then demodulated to reveal the AM modulation products. The rms level of the modulation products is measured and expressed as a ratio to the rms level of F2. The SMPTE IMD measurement includes noise within the passband, and is insensitive to FM (frequency modulation) distortion.

The APx500 implementation of SMPTE IMD provides the capability to vary the stimulus frequencies. F1 can be in the range of 40 Hz to 1 kHz; F2 can be in the range of 2 kHz to 60 kHz. However, F2 must always be at least 6 times greater than F1.

The level ratio F1:F2 can be set to the standard 4:1, or to 1:1.

Distortion products out to the 5th order are measured and reported in the distortion product result.

MOD IMD

MOD IMD is similar to SMPTE IMD, and in some cases produces the same results. The MOD stimulus is the same as the SMPTE stimulus, but instead of using AM demodulation for analysis MOD selectively measures the 2nd and 3rd order intermodulation products and combines their values arithmetically. This method reduces the influence of noise in the result, and is sensitive to any distortion mechanism. When only amplitude modulation IMD exists, and when noise is very low, the MOD results will be identical to the SMPTE results.

The APx500 implementation of MOD IMD provides the capability to vary the stimulus frequencies. F1 can be in the range of 40 Hz to 1 kHz; F2 can be in the range of 2 kHz to 60 kHz. However, F2 must always be at least 6 times greater than F1.

The level ratio F1:F2 can be set to the standard 4:1, or to 10:1 or 1:1.

The 4th and 5th order products are also measured and can be reported in the distortion product result.

DFD IMD

DFD stands for Difference Frequency Distortion. DFD is described in the standards IEC60118 and IEC60268.

The DFD stimulus is two equal-level high-frequency tones f1 and f2, centered around a frequency called the mean frequency, (f1+f2)/2. The tones are separated by a frequency offset called the difference frequency. The two tones intermodulate in a nonlinear DUT to produce sum and difference frequencies.

The APx500 implementation of DFD IMD provides the capability to vary the stimulus frequencies. The mean frequency can be in the range of 250 Hz to 60 kHz; and the difference frequency can be in the range of 80 Hz to 2 kHz.

For analysis DFD selectively measures the 2nd and 3rd order intermodulation products, combines their values arithmetically and provides a result that is the ratio of the sum of the products to a reference voltage defined as 2x the voltage of f2 (effectively, the sum of f1 and f2). In the APx500 implementation, the 4th and 5th order products are also measured and can be reported in the distortion product result.

Because the stimulus tones are high in frequency, DFD is a useful measurement for observing distortion in devices that exhibit distortion that rises with frequency. Since the tones by default are only 80 Hz apart, much of the energy contained in the distortion products will fall near or below the stimulus tones. This makes DFD a good choice for measuring distortion at higher frequencies in band limited devices, where harmonic distortion products from high-frequency stimulus tones would fall out of band.

DFD measurements are made in the same way as CCIF measurements, differing only in amplitude calibration. DFD results are expressed as values 6.02 dB lower than CCIF.

CCIF IMD

The CCIF IMD method is described in document no. 11 of the Commission Mixte, CCIF/UIR, March 1937, issued by the International Telephonic Consultative Committee (CCIF). CCIF no longer exists as an organization, having become the ITU-R division of the International Telecommunications Union (ITU). This method is also referred to as IMD (ITU-R).

The CCIF stimulus is two equal-level high-frequency tones f1 and f2, centered around a frequency called the mean frequency, (f1+f2)/2. The tones are separated by a frequency offset called the difference frequency. The two tones intermodulate in a distorting DUT to produce sum and difference frequencies.

The APx500 implementation of CCIF IMD provides the capability to vary the stimulus frequencies. The mean frequency can be in the range of 250 Hz to 60 kHz; and the difference frequency can be in the range of 80 Hz to 2 kHz.

For analysis CCIF selectively measures the 2nd and 3rd order intermodulation products, combines their values arithmetically and provides a result that is the ratio of the sum of the products to a reference voltage defined as 2x the voltage of f2 (effectively, the sum of f1 and f2). In the APx500 implementation, the 4th and 5th order products are also measured and can be reported in the distortion product view.

Because the stimulus tones are high in frequency, CCIF is a useful measurement for observing distortion in devices that exhibit distortion that rises with frequency.

Since the tones by default are only 80 Hz apart, much of the energy contained in the distortion products will fall near or below the stimulus tones. This makes CCIF a good choice for measuring distortion at higher frequencies in band limited devices, where harmonic distortion products from high-frequency stimulus tones would fall out of band.

CCIF measurements are made in the same way as DFD measurements, differing only in amplitude calibration. CCIF results are expressed as values 6.02 dB higher than DFD.

Filters

Filters are not provided with IMD measurements, but APx Signal Path > Input > Filters can be applied. Any filtering in the range of the fundamentals or the distortion products would give non-standard results of questionable usefulness.

Advanced Settings for IMD

The default settings here are appropriate for most measurements. You may want to make minor adjustments for special situations.

Signal Generation

If **Track first channel level** is checked (the default), the generator output level values for channel 1 are copied to the other channels, and output levels for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels.

To set individual generator output channel levels, uncheck the **Track first channel level** checkbox and enter values in the output channel Level fields.

Signal Acquisition and Analysis

Input Range tab

Digital inputs do not require ranging, and the Input Range tab is unavailable when inputs are set to a digital format.

If Auto Range is checked (the default), each analog channel determines its input range automatically, based on the level of the input signal. If the input signal level changes beyond a ranging threshold, Auto Range will cause the input ranging circuits to move up or down for proper ranging.

See page 551 for more information about ranging and autoranging.

If Auto Range is unchecked, you can set a fixed range for each analog input channel. If Track first channel range is checked (the default), the fixed input range setting for channel 1 is copied to the other channels, and range settings for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels.

To set individual analog input channel ranges, uncheck the Track first channel range checkbox and enter values in the input channel Range fields.

Settling tab

A Settling tab is available for IMD measurement results. See page 552 for more information about settling.

45

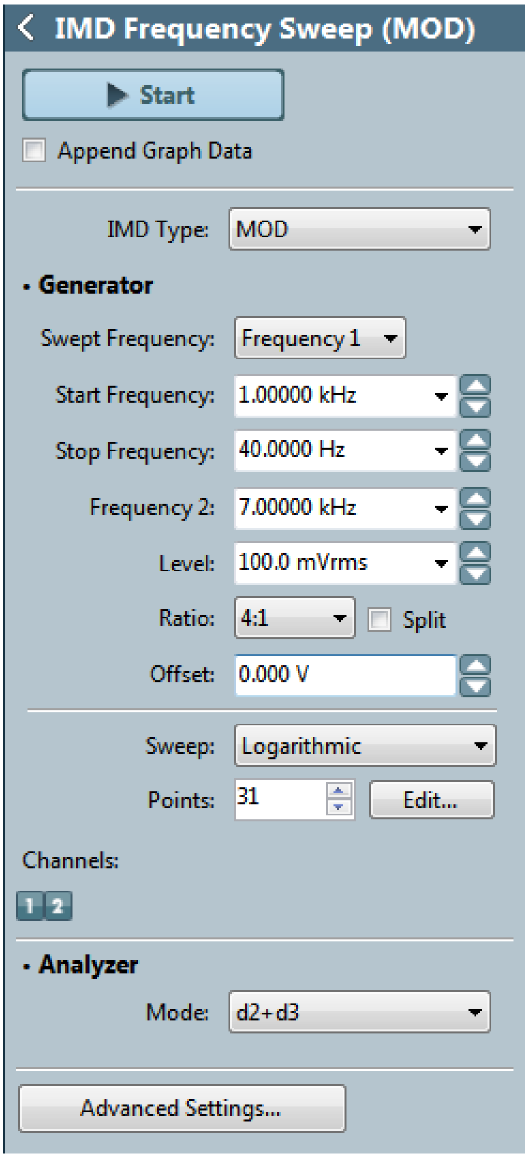
IMD Frequency Sweeps
(Sequence Mode)

Introduction

For the APx515, these measurements require a software option key. See page 166 for more information about software options.

IMD measurements use two tones of different frequencies summed into a stimulus signal. For IMD Frequency Sweep measurements, one of these tones is held at a fixed frequency while the other is swept through a range of frequencies. The result is displayed as an IMD Ratio result, with the swept frequency on the X-axis and the IMD Ratio on the Y-axis.

The IMD measurements in APx provide stimulus tones and analysis for four important methods of measuring intermodulation distortion: SMPTE, DFD, CCIF and MOD. Additionally, DIM IMD measurements are available with specific analyzer hardware.



Operation

If you have not yet set up your test, first go to Signal Path Setup (Chapter 6).

IMD Frequency Sweep measurements are not available in External Source configuration.

IMD Type

Select the IMD type here. Read more about IMD beginning on page 273 for descriptions of the IMD types and stimulus frequencies.

- SMPTE
- DFD
- CCIF
- MOD

Generator

Swept Frequency

SMPTE or MOD

For SMPTE or MOD, select Frequency 1 or Frequency 2 as the Swept Frequency.

DFD or CCIF

For DFD or CCIF, select Mean Frequency or Diff Frequency as the Swept Frequency.

Start Frequency

Select the Start frequency of the swept tone here. This frequency is constrained only by the DAC generator constraints.

Stop Frequency

Select the Stop frequency of the swept tone here. This frequency is constrained only by the DAC generator constraints.

(Unswept) Frequency

For SMPTE or MOD, the control located below Stop Frequency will be labeled Frequency 1 or Frequency 2.

The SMPTE and MOD stimulus frequency constraints are listed in the More on IMD discussion beginning on page 273.

For DFD or CCIF, the control located below Stop Frequency will be labeled Mean Frequency or Diff Frequency. In any case, enter the desired frequency for the unswept tone here.

The DFD and CCIF stimulus frequency constraints are listed in the More on IMD discussion beginning on page 273.

Level

Set the generator level here.

Ratio

For SMPTE, select a 1:1 or 4:1 ratio (level ratio of Freq 1 to Freq 2). For MOD, select 1:1, 4:1 or 10:1.

Split

The Split setting defeats the summing of the two IMD frequencies within the generator. Instead, the signals are “split” and routed independently to channel outputs. Frequency 1 or the Mean Frequency is routed to odd-numbered channels, and Frequency 2 or the Diff Frequency is routed to even-numbered channels.

Acoustic testing of microphones sometimes uses two stimulus loudspeakers, each with a single frequency component of an IMD stimulus. The stimulus signals are summed acoustically at the microphone element. This method provides a distortion result for the microphone while minimizing the distortion contributions of the loudspeakers.

Offset

Set any DC offset here.

Sweep

Select Logarithmic, Linear or Custom.

Points

For Logarithmic or Linear, set the number of points here.

Edit

Opens the Sweep Points editor. See page 457 for information about viewing and editing the Sweep Points table.

Step Size

For Linear, set the Step Size here.

Analyzer

Filters

No local low pass, high pass or weighting filters are available for this measurement. However, low pass, high pass and weighting filter settings made in Signal Path Setup > Input/Output will affect this measurement. Read more about APx Filters in Chapter 91.

Mode

The Mode selector is not available for the SMPTE Frequency Sweep.

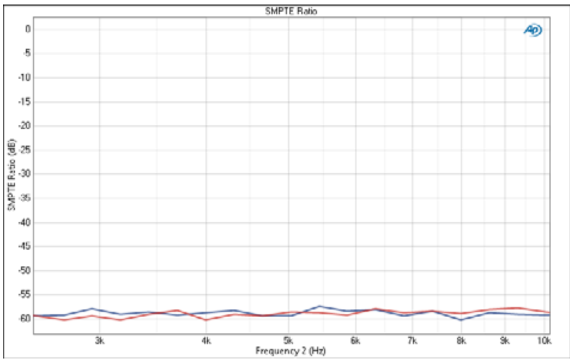
For CCIF and MOD, the Mode selector allows you to choose the distortion products or combinations of distortion products to be displayed. The default is d2 + d3 (2nd order plus 3rd order products).

Advanced Settings

If your test required special adjustments or settings, click Advanced Settings. See page 317.

See Chapter 98 for more information about units of measurement.

IMD Ratio



IMD Ratio plots the distortion ratio on the Y axis against the Generator frequency on the X axis, using the SMPTE, DFD, CCIF or MOD IMD method.

Units

Units available for IMD Ratio are

X-axis (analog)	***Y-axis***
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

46

IMD Level Sweeps
(Sequence Mode)

Introduction

For the APx515, these measurements require a software option key. See page 166 for more information about software options.

The SMPTE, DFD, CCIF and MOD Level Sweep measurements are combined into one measurement. Use the **IMD Type** menu to select **SMPTE, DFD, CCIF** or **MOD.**

IMD measurements use two tones of different frequencies summed into a stimulus signal. For IMD Level Sweep measurements, this stimulus is swept through a range of levels. The result is displayed as two IMD Ratio results, one with the swept generator level on the X-axis, the other with the measured level on the X-axis. Both plot the IMD Ratio on the Y-axis.

Operation

If you have not yet set up your test, first go to Signal Path Setup (Chapter 6).

IMD Level Sweep measurements are not available in External Source configuration.

Append Graph Data

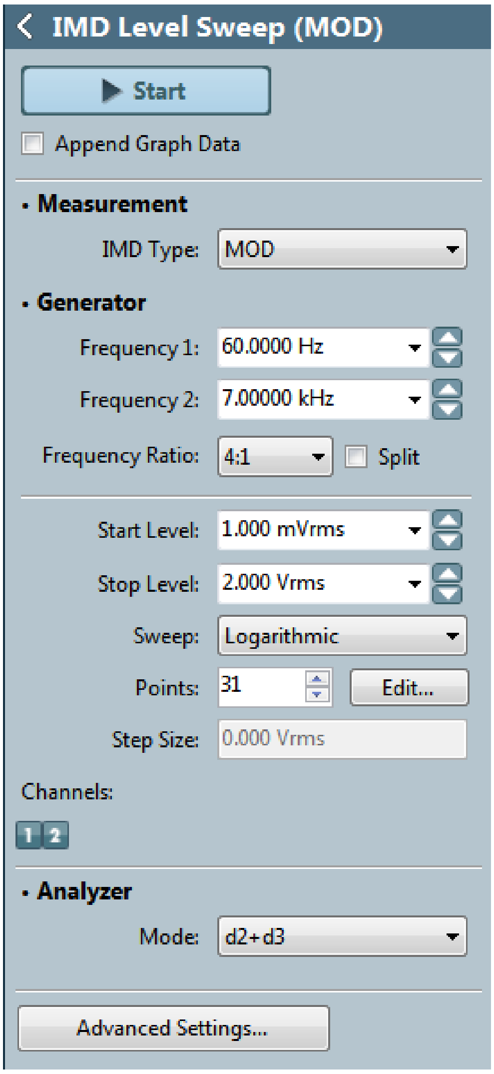
Normally, the current graph data for all results is deleted each time you start a new measurement. If the **Append** box is checked, the current data is kept in memory as a **Data Set** (see page 165), and the new measurement data is appended as a new data set. You can append many **Data Sets.**

Measurement

IMD Type

Select the IMD type here. Read more about IMD beginning on page 273.

- SMPTE



- DFD
- CCIF
- MOD

Generator

The configuration for this measurement must be the closed-loop configuration, using the APx sine generator as a stimulus. See Chapter 5 for information about the using the APx Generator and setting Waveform, Level and Frequency.

Frequency

Set the IMD stimulus here.

- For SMPTE or MOD, set Frequency 1 or Frequency 2. These measurements use dual-tone stimulus, with a lower, dominant frequency (Frequency 1) and a higher frequency of interest (Frequency 2). In a nonlinear device, the tones modulate each other. You can choose different frequency values here.
  Frequency 1 minimum setting is 40 Hz. Maximum setting varies with the Frequency 2 setting, ranging from 333.33 Hz to 1 kHz. The default is 60 Hz. The Frequency 2 setting range is 2 kHz to 20 kHz. The default is 7 kHz.
- For DFD or CCIF, set Mean Frequency or Diff Frequency.
  DFD and CCIF use a dual-tone stimulus, with two high frequency tones separated by a Difference Frequency, centered around a Mean Frequency. You can choose different frequency values here. Mean Frequency minimum settings vary with the Difference Frequency setting, ranging from 2.5 kHz to 12 kHz. Maximum setting is 20 kHz. The default is 12.5 kHz.
  The Difference Frequency setting range is 80 Hz to 2 kHz. The default is 80 Hz.

Frequency Ratio

For SMPTE, select a 1:1 or 4:1 ratio (level ratio of Freq 1 to Freq 2). For MOD, select 1:1, 4:1 or 10:1.

Split

The Split setting defeats the summing of the two IMD frequencies within the generator. Instead, the signals are “split” and routed independently to channel outputs. Frequency 1 or the Mean Frequency is routed to odd-numbered channels, and Frequency 2 or the Diff Frequency is routed to even-numbered channels.

Acoustic testing of microphones sometimes uses two stimulus loudspeakers, each with a single frequency component of an IMD stimulus. The stimulus signals are summed acoustically at the microphone element. This method provides a distortion result for the microphone while minimizing the distortion contributions of the loudspeakers.

Running the measurement

To make an IMD Level Sweep measurement, click Start. The generator will output the IMD stimulus to the DUT on the selected generator channels. The level of the signal will be swept from the Start Level to the Stop Level in the set number of points. You can select Logarithmic or Linear point spacing, or choose Edit... to customize the Sweep Points table.

Start Level

Select the Start level of the swept tone here.

Stop Level

Select the Stop level of the swept tone here.

Sweep

Select Logarithmic, Linear or Custom.

Points

For Logarithmic or Linear, set the number of points here.

Edit

Opens the Sweep Points editor. See page 462 for information about viewing and editing the Sweep Points table.

Step Size

For Linear, set the Step Size here.

Analyzer

No local low pass, high pass or weighting filters are available for this measurement. However, low pass, high pass and weighting filter settings made in Signal Path Setup > Input/Output will affect this measurement. Read more about APx Filters in Chapter 91.

Mode

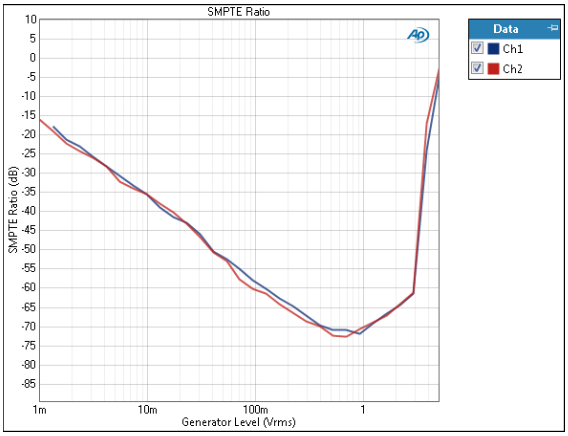
The Mode selector is not available for the SMPTE Frequency Sweep.

For CCIF and MOD, the Mode selector allows you to choose the distortion products or combinations of distortion products to be displayed. The default is d2 + d3 (2nd order plus 3rd order products).

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, click Advanced Settings. See Advanced Settings for Stepped Sweeps on page 458. See Chapter 98 for more information about units of measurement.

IMD Ratio vs. Generator Level



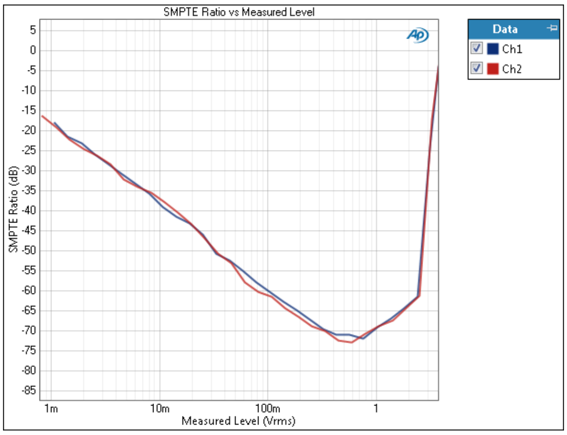
IMD Ratio plots the distortion on the Y axis against the Generator level on the X axis, using the SMPTE, DFD CCIF or MOD IMD method.

Units

Units available for the IMD Ratio result are

X-axis (analog)	*X-axis (digital)*	*Y-axis*
• Vrms	• FS	• x/y
• Vp	• %FS	• %
• Vpp	• dBFS	• ppm
• dBV	• dBrG	• dB
• dBu		
• dBrG		
• dBm		
• W (watts)		

IMD Ratio vs. Measured Level



IMD Ratio vs. Measured Level plots the distortion on the Y axis against the measured DUT output level on the X axis, using the SMPTE, DFD CCIF or MOD IMD method.

Units

Units available for the IMD Ratio vs. Measured Level result are

X-axis (analog)	*X-axis (digital)*	*Y-axis*
• Vrms	• FS	• x/y
• dBV	• %FS	• %
• dBu	• dBFS	• ppm
• dBrA	• dBrA	• dB
• dBrB	• dBrB	
• dBSPL1	• dBSPL1	
• dBSPL2	• dBSPL2	
• dBm)		
• W (watts)		

47

Impedance/Thiele-Small (Sequence Mode)

This measurement requires a software option key. See page 166 for more information about software options.

Impedance/Thiele-Small is only available when the analyzer input and output are set to an analog, non-acoustic configuration.

Overview

The Impedance/Thiele-Small measurement measures the complex impedance of a loudspeaker driver, providing impedance response curves and Thiele-Small parameters. The voltage across the driver is measured, and the current through the driver is calculated by measuring the voltage across a precision sense resistor in series with the driver. The underlying technology is the continuous sweep.

Impedance/Thiele-Small results available in APx500 are:

- Impedance Magnitude
- Impedance Phase
- Impedance Real
- Impedance Imaginary
- Thiele-Small parameters

These results are available from a single acquisition; however, depending upon the Thiele-Small mode selected, a second measurement may be necessary to obtain the full set of Thiele-Small parameters.

Thiele-Small parameters

The Thiele-Small (T-S) parameters are a set of electro-mechanical parameters that define the low-frequency performance of a loudspeaker driver.

T-S parameters are often referred to as “small signal” parameters, because the results are most useful when the driver remains in its linear range, not over-driven by a large signal. Driver specifications often include a table of T-S parameters to aid design engineers in driver selection and in enclosure design.

The T-S parameters are derived from measurements of a driver’s impedance curve, with additional information about the mechanical characteristics of the driver.

The additional information can be attained by modifying the mechanical loading of the driver and making a second measurement, or by entering the M_{MD} (moving mass) of the driver, measured outside of APx. These methods are discussed in detail below.

Symbol	*Unit*	*Description*
F_S	Hz	Resonant frequency of the driver
Q_{MS}	----	Mechanical *Q* at F_S
Q_{ES}	----	Electrical *Q* at F_S
Q_{TS}	----	Total *Q* at F_S
S_D	cm^2	Effective surface area of cone
R_E	Ω	Voice coil resistance
L_E	mH	Voice coil inductance
R_2	Ω	LR-2 model: voice coil parallel resistance
L_2	mH	LR-2 model: voice coil parallel inductance
E_{rm}	----	Wright model: motor resistance exponent
K_{rm}	----	Wright model: motor resistance coefficient
E_{xm}	----	Wright model: motor reactance exponent
K_{xm}	----	Wright model: motor reactance coefficient
R_{MS}	Ns/m	Mechanical resistance of suspension
C_{MS}	mm/N	Mechanical compliance of the suspension
M_{MS}	gram	Mechanical mass including air load
V_{AS}	liter	Acoustic volume with same compliance as suspension
Bl	Tm	Magnetic motor strength
η_0	%	Reference efficiency of the driver

R_2 and L_2 are only populated when the LR-2 model is selected. Erm, Krm, Exm and Kxm are only populated when the Wright model is selected.

Impedance Response

The impedance of a loudspeaker driver is given by dividing the voltage across the voice coil by the current passing through it. The reactance of the voice coil and the action of the mechanical system coupled to it do not present a pure resistance. When stimulated by an alternating current, the impedance is expressed as a complex number.

In the APx implementation, the impedance is calculated from a voltage measurement taken across the driver voice coil and the current in the driver circuit. The driver current is calculated by measuring the voltage across a sense resistor of known value, by I = E/R.

The stimulus signal is an audio sweep, resulting in an impedance response curve plotted against frequency. The underlying measurement technology is the APx continuous sweep.

Connections and Test Configurations

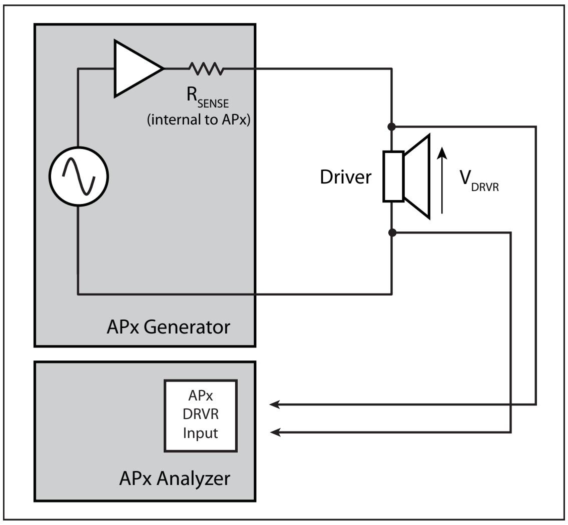
The Impedance/Thiele-Small measurement can use one of several test configurations to derive the driver current from the available voltages. Choose the test configuration suited to your situation.

In these tests, we are not measuring conventional audio parameters; we are measuring voltage or current in circuit components. Consequently, one or two of the analyzer input channels must have specific roles assigned, either to measure the voltage across the driver, the voltage across the sense resistor, or the total voltage across the driver and the sense resistor. Please pay close attention to the roles depicted in the illustrations below and identified on the **Channel** control in the software.

We recommend that you use balanced analyzer inputs for impedance measurements.

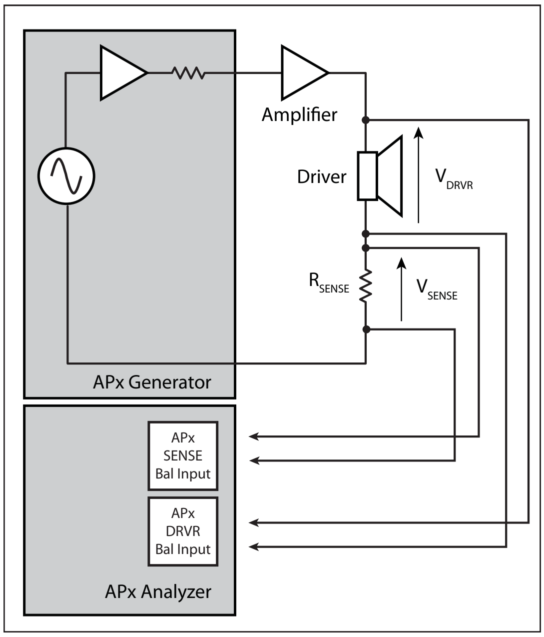
Choose the test configuration suited to your situation.

Internal



This method is convenient but less accurate than the External (2 Ch) configurations discussed below. It is appropriate for low-power demonstration purposes, or when no power amplifier or precision sense resistor is available. The Internal method uses the source resistor (a known value) in the final stage of the APx output circuitry as the current sense resistor. This method assumes that the APx directly drives the loudspeaker driver, and that no external power amplifier or sense resistor is used.

External (2 Ch) (bal)



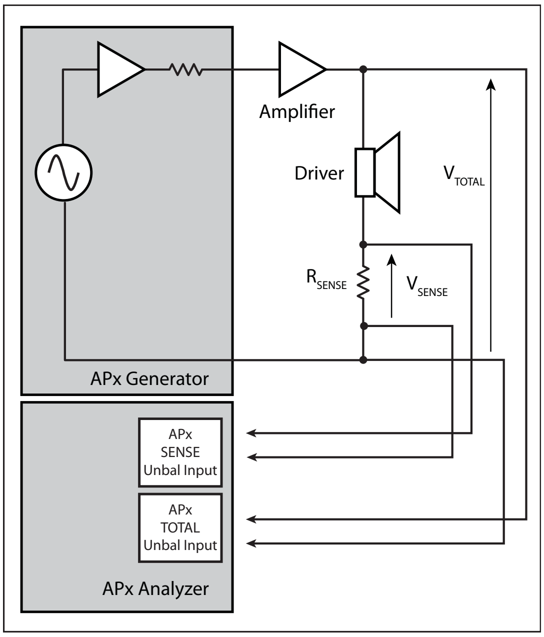
This is our recommended configuration. when making a 2-channel impedance measurement. This configuration is enabled when **Test Configuration** is set to **Exter-**

nal (2 Ch) and the analyzer inputs are set to balanced. Because V_{DRVR} and V_{SENSE} are measured simultaneously, no assumptions are made regarding the amplifier response. Impedance results are reported correctly across the frequency range.

The Audio Precision IMP1 Impedance Fixture can be used with this configuration.

Avoid the use of a bridged amplifier as the accuracy of results can be significantly degraded due to strong common mode signals at the analyzer input.

External (2 Ch) (unbal)



External (2 Ch) unbal

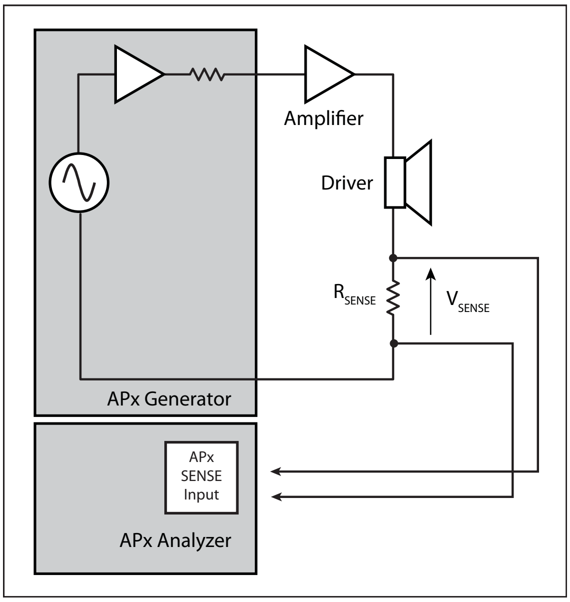
This configuration is enabled when Test Configuration is set to External (2 Ch) and the analyzer inputs are set to unbalanced.

This configuration is similar to External (2 Ch) (bal), but it allows the use of unbalanced analyzer inputs, which may be more convenient in some situations. V_{DRVR} is calculated by subtracting V_{SENSE} from V_{TOTAL}. When the analyzer is set to unbalanced inputs, this configuration is assumed and voltage measured at channel *n*+1 is subtracted from the voltage at channel *n*, provided the driver voltage.

The Audio Precision IMP1 Impedance Fixture can be used with this configuration.

A bridged amplifier cannot be used with unbalanced analyzer inputs.

External (1 Ch)



External (1 Ch)

This method uses an external power amplifier and sense resistor.

Set Correction Curve to select an amplifier correction curve in the project, or choose to Create New or Browse for File. If the correction curve accurately represents the amplifier in use, this configuration will make accurate impedance response measurements. See Amplifier Correction on page 292.

Enter the value of the sense resistor into the data entry field in APx. This is our recommended configuration when making a 1 channel impedance measurement.

If Correction Curve is set to None, the broadband gain of the amplifier must be known from a previous measurement, such as the APx Level and Gain measurement. Enter the gain and the value of the sense resistor into the data entry fields in APx.

When Correction Curve is set to None, this configuration cannot correct for the low-frequency rolloff present in an AC-coupled power amplifier. AC coupling will introduce phase shift that clearly affects the driver impedance curve. If your test requires a sweep that extends below 20 Hz, we recommend the use of a DC-coupled power amplifier, or that you use a correction curve as described above.

We recommend that you use balanced analyzer inputs for impedance measurements.

Avoid the use of a bridged amplifier as the accuracy of results can be significantly degraded due to strong common mode signals at the analyzer input.

Mounting the driver for testing

Free air

For the free-air measurement, the driver should be suspended or mounted to a rigid, open frame in free air, as far from reflecting surfaces as possible. The driver should be oriented so that the line of travel of the voice coil is perpendicular to the force of gravity.

Loading the driver for a second measurement

Unless you know the M_{MD} of your driver (see Known M_{MD}, below), the full set of Thiele-Small parameters requires a second measurement, where the driver has been loaded with either added mass or the air compliance of a sealed enclosure.

Added Mass

The driver should be mounted as for a free-air measurement, with mass added to the driver in a balanced pattern that does not interfere with the motion of the cone. This is typically done by placing a uniform toroid of modeling clay (such as Plasticine or Handi-TAK) around the driver dust cap.

Adding mass to the driver will shift its resonant frequency (the peak in the impedance curve) downward. You must add enough mass to shift the resonant frequency by at least 5%. APx accepts a range of 1 g to 1000 g.

Known Volume

The driver should be mounted on a sealed enclosure, in the same orientation to the Earth as indicated in the free air instructions.

If the driver is facing in to the enclosure, calculate the total volume by adding the volume of the enclosure to the volume of the truncated conic section formed by the driver cone.

If the driver is mounted inside a sealed enclosure facing out, you must subtract the volume of the driver cone, frame and magnet assembly from the volume of the enclosure.

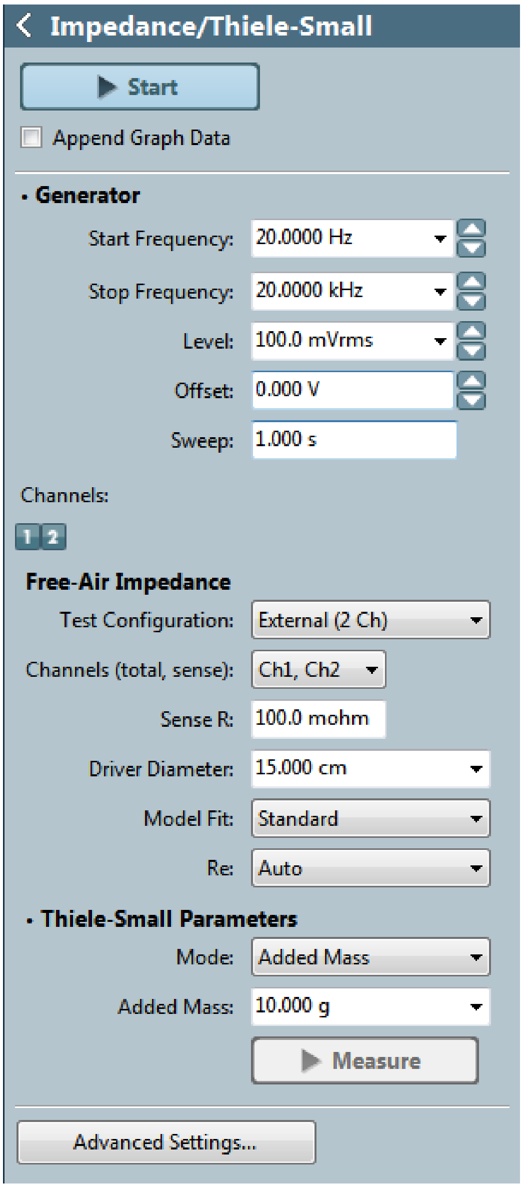
Mounting the driver on a sealed enclosure will shift its resonant frequency (the peak in the impedance curve) upward by reducing the compliance of the system; the smaller the enclosure, the more the resonant frequency is shifted. You must reduce the compliance enough to shift the resonant frequency by at least 5%. APx accepts a range of 0.01 liter to 1000 liters.

Known M_{MD}

Rather than making two measurements for T-S as described above, you can enter the M_{MD} (the moving mass of the driver without air load) and derive the full set of parameters in one measurement pass. The M_{MD} may be available from a manufacturer's specification, or can be obtained by disassembling an identical driver and weighing the moving parts. Enter the M_{MD} into the **Known M_{MD}** field.

The **Known M_{MD}** technique has the advantage of faster and more precise measurements.

Common Controls



Operation

If you have not yet set up your test, first go to Signal Path Setup (Chapter 6).

Note: be certain to disable all input filters for this measurement. Filtering the acquisi-

tion will result in incorrect impedance curves. For low-frequency drivers, be sure that the input high-pass filter is set to DC (DC coupled).

Before you run the measurement

Before you run the measurement, you must choose a Test Configuration and enter a number of values as described in detail below. If you are using the Known M_{MD} mode, you must choose that mode in the Thiele-Small Parameters area, and enter the exact value of the M_{MD} in the Mmd entry field before running the measurement.

Append Graph Data

Normally, the graph data in memory is deleted each time you start a new measurement. If the Append Graph Data box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox and a Notes field for each data set.

For the Impedance/Thiele-Small measurement, only the Free-Air Impedance data (the first measurement) can be appended as new Data Sets (Measured 1, Measured 2, etc.). The data acquired in the second Thiele-Small Parameters acquisition (for Added Mass or Known Volume) are applied to the final measured Data Set, and are available for viewing as additional traces on that graph. The second measurement can be run repeatedly, updating the second set of traces but not appending new Data Sets.

Generator

The configuration for this measurement must be the closed-loop configuration, using the APx sine generator as a stimulus. See Chapter 5 for information about the using the APx Generator and setting Waveform, Level and Frequency.

If you have not yet set up your test, first go to Signal Path Setup (Chapter 6).

The Impedance/Thiele-Small measurement uses a continuous sweep (log chirp) stimulus signal, swept between the frequencies entered in Start Frequency and Stop Frequency, at the value set in Level.

Note: Loudspeakers and drivers under test can be damaged by signals that exceed their level or frequency limits. Be sure to set generator and amplifier level and sweep frequencies to values within the limits of the loudspeaker.

Click Start. This plots the impedance response curves and populates the Thiele-Small parameter values calculated from the impedance measurement. If you have elected to use the Known M_{MD} mode, this will be the full set of T-S parameters. Otherwise, a second measurement pass (see Thiele-Small Parameters, below) is required.

Start Frequency and Stop Frequency

These controls set the frequency range of the sweep. For loudspeaker driver testing, the sweep range is typically set to the nominal range of the driver. See page 220 for more about continuous sweep ranges, which are dependent upon the instrument.

Level

Set the generator level here.

Offset

Set any DC offset to the generator signal here.

Sweep

A continuous sweep stimulus consists of a sine wave swept logarithmically across the sweep range in the period of time set in Sweep.

The default for Sweep is 1 s. Minimum is 50.0 ms; maximum is 5 s.

Longer Sweep settings provide greater resolution and signal-to-noise ratio, but increase measurement time.

Analyzer: Free-Air Impedance

Test Configuration

This measurement supports three test configurations. The selection made here affects the availability of other settings below. The test configurations are:

- Internal
- External (2 Ch)
- External (1 Ch)

See page 286 for a full description of these configurations.

Internal

See a diagram of the Internal test configuration on page 286.

This method is convenient but less accurate than the “External (2 Ch)” configuration discussed below. It is appropriate for low-power demonstration purposes, or when no power amplifier or precision sense resistor is available.

Channel

The Internal Test Configuration requires 1 input channel, which is assigned the role of “drvr”. This channel measures the voltage across the loudspeaker driver. No external power amplifier or external sense resistor is used.

External (2 Ch)

This is our recommended configuration. It requires an external power amplifier and an external sense resistor. It requires 2 analyzer input channels, but because both the voltage across the driver and the current through the driver are measured simultaneously, it produces the best results.

Channel

Balanced inputs

See a diagram of the External (2 Ch) (bal) test configuration on page 286.

When using balanced inputs, the External (2 Ch) Test Configuration requires one input channel to be assigned the role of “drvr”. This channel measures the voltage across the loudspeaker driver.

It requires a second input channel to be assigned the role of “sense”. This channel measures the voltage across the sense resistor.

Channel roles are assigned in adjacent pairs, with “drvr” always the lower numbered channel.

Unbalanced inputs

See a diagram of the External (2 Ch) (unbal) test configuration on page 287.

When using unbalanced inputs, the External (2 Ch) Test Configuration requires one input channel to be assigned the role of “total”. This channel measures the total voltage across the loudspeaker driver and the sense resistor. This is the same as the amplifier output voltage.

It requires a second input channel to be assigned the role of “sense”. This channel measures the voltage across the sense resistor.

Channel roles are assigned in adjacent pairs, with “total” always the lower-numbered channel. The voltage across the driver is calculated by subtracting the sense voltage from the total voltage.

Sense R:

Enter the value of the external sense resistor in this field.

External (1 Ch)

See a diagram of the External (1 Ch) test configuration on page 287.

This configuration allows you to make an impedance measurement using an external power amplifier and an external sense resistor, but requires only 1 analyzer input channel. You must use an amplifier correction curve (see below) or provide a broadband amplifier gain value. We recommend using an amplifier correction curve, as it is more accurate. Characteristics of the amplifier used and the accuracy of the correction curve will affect your results.

Correction Curve:

- None
  If you are not using a correction curve, choose None and enter a value for the amplifier gain in the field below. The response of your amplifier will affect the accuracy of the impedance curves. See External (1 Ch) Test Configuration on page 287 for more information.
- If there are Amplifier Correction Curves in the project, they will be listed here and available for selection.
- Create New
  To create an amplifier correction curve, choose Create New. See Amplifier Correction on page 292.
- Browse for file
  To use a previously saved amplifier correction curve, choose Browse for file.See Amplifier Correction on page 292.

Amplifier Gain

This field is only available when Test Configuration is set to External (1 Ch) configuration and Correction Curve is set to None. Determine the gain of your amplifier using a measurement such as the APx Level and Gain, and enter the value here. See External (1 Ch) Test Configuration on page 287 for more information.

Channel

The External (1 Ch) Test Configuration requires one input channel to be assigned the role of “sense”. This channel measures the voltage across the sense resistor.

Sense R:

Enter the value of the external sense resistor in this field.

Driver Diameter

Enter the driver diameter in this field.

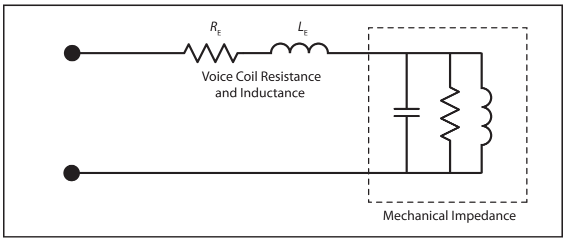
Model Fit

After the preliminary set of Thiele-Small parameters are obtained from the free air pass, a model of the impedance curve is created. This model is plotted with the result data to allow you to see how well the model curve and the data fit. The model curve is displayed as a trace called “Fit.” Choose

- Standard
- LR-2 or
- Wright

Standard model

The standard model provides a good fit at lower frequencies but may not represent the physical driver well at higher frequencies.



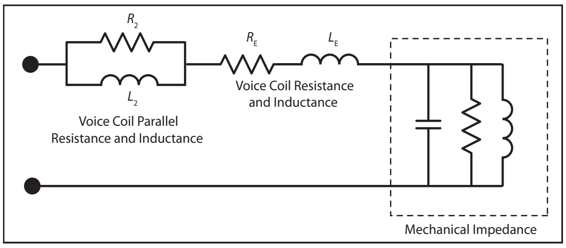
Standard Model

The moving mass of the driver is modeled as a resonant system, shown as the dotted box on the right. All moving coil drivers have a voice coil with resistance and inductance. The Standard Model assumes that the simple series resistance plus inductance of the voice coil accurately models the electrical impedance of the driver.

In a practical system, eddy current losses in the magnet and pole piece cause the real part of the impedance to climb with frequency, which the Standard Model does not predict.

LR-2 model

The LR-2 and Wright models add an additional network of resistance and inductance to the standard model and typically provide a better fit at high frequencies.

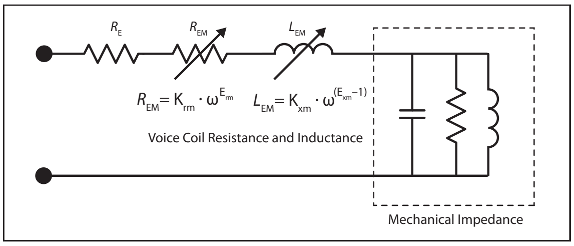


LR-2 Model

The LR-2 Model assumes that the network shown above in series with the mechanical impedance accurately models the electrical impedance of the coil, including eddy current losses. This model provides a more accurate fit to the measured impedance curve than the Standard Model, and can be implemented with physical components or digital filters.

For more information, see Mark Dodd, Wolfgang Klippel, and Jack Oclee-Brown, “Voice Coil Impedance as a Function of Frequency and Displacement”, AES 117th Convention, Oct 28-31 2004. This paper is available from the Audio Engineering Society eLibrary at aes.org.

Wright model



Wright Model

The Wright Model assumes that the network shown above in series with the mechanical impedance accurately models the electrical impedance of the coil, including eddy current losses. This model provides a very accurate fit to the measured impedance curve, but cannot be implemented with physical components or digital filters because it uses unrealizable parameters, such as fractional resistance and inductance. With certain software tools, the Wright model is useful in crossover and enclosure design.

For more information, see J. R. Wright, “An Empirical Model for Loudspeaker Motor Impedance”, JAES, Vol. 38, No. 10, October 1990. This paper is available from the Audio Engineering Society eLibrary at aes.org.

R_E

R_E is a Thiele-Small parameter representing the DC resistance (DCR) of the voice coil. Choose

- Auto
  Auto obtains R_E in finding the best model fit to the impedance curve.
- Fixed (DCR)
  Fixed (DCR) allows you to enter the DC resistance of the voice coil, measured outside of APx using a precision ohmmeter.

DCR

Enter the resistance value in the DCR field.

Thiele-Small Parameters

Mode

The Mode selected in Thiele-Small parameters determines whether or not a second measurement pass is required, and what further information must be entered for the second pass. If you are using the Known M_{MD} mode, you must choose that mode and enter the exact value of the M_{MD} in the **Mmd** entry field before you click **Start**. See Known M_{MD} on page 288.

For **Added Mass** or **Known Volume**, you must first run the free-air impedance measurement, as described above in **Running the Measurement**.

When that measurement has been made, choose the driver loading mode for Thiele-Small: **Added Mass** or **Known Volume**.

For **Known M_{MD}**, only the first free-air measurement is necessary.

Added Mass

If you do not know the M_{MD} of the driver, you can choose the **Added Mass** mode to fully populate the table of T-S parameters. You must first run the free-air impedance measurement initiated by the **Start** button, as described above in **Running the Measurement**. For **Added Mass**, you must add mass to the driver as described in **Mounting the driver for testing: Free air** on page 288. This will shift the resonance frequency of the driver downward. Enter the exact value for the added mass in the **Added Mass** field and click **Measure**.

Known Volume

If you do not know the M_{MD} of the driver, you can choose the **Known Volume** mode to fully populate the table of T-S parameters. You must first run the free-air impedance measurement initiated by the **Start** button, as described above in **Running the Measurement**. For **Known Volume**, you must mount the driver to a sealed enclosure of known volume, as described in **Mounting the driver for testing: Free air** on page 288. This will shift the resonance frequency of the driver upward. Enter the exact value of enclosure volume in the field provided and click **Measure**.

Known M_{MD}

If you know the M_{MD} (moving mass) of the driver under test, set **Mode** in Thiele-Small parameters to **Known M_{MD}**, enter the exact value of the M_{MD} in the **Mmd** entry field, and click **Start** as described above in **Running the Measurement**. In this case, only the one free-air measurement is necessary to obtain the full set of Thiele-Small parameters. See **Known M_{MD}** in **Mounting the driver for testing: Free air** on page 288.

Using the **Known M_{MD}** mode is not always possible. If you do not know the M_{MD} (moving mass) from a specification, you must dismantle the driver and weigh its moving components.

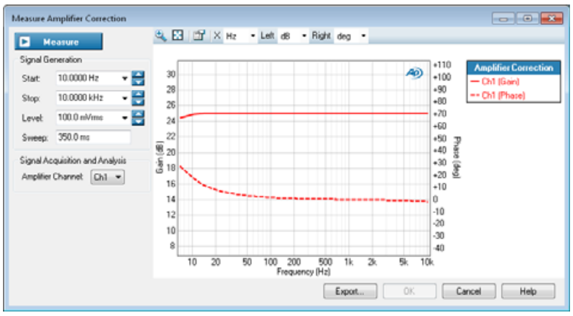
Measure

For **Added Mass** or **Known Volume** modes, after the impedance response measurement initiated by the **Start** button is complete, you must make a second measurement to obtain the full set of T-S parameters. Click **Measure** to run the second measurement pass.

Amplifier Correction

For accurate 1-channel impedance response measurements, the amplifier's gain response and phase response must be known.

The **Measure Amplifier Correction** dialog provides a convenient way to make this measurement and attach the correction curve to the project.



Signal Generation

Select a sweep range, level and duration.

Connections and Signal Acquisition and Analysis

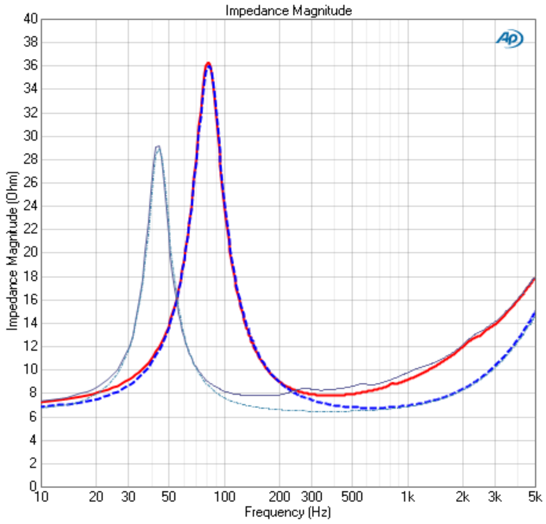
In the **Measure Amplifier Correction** dialog, select the channel to be used for measurement. Connect the amplifier input to the APx generator output for this channel, and the amplifier output to APx analyzer input for this channel.

Click **Measure**. APx will measure and display the amplifier's gain response and phase response.

Click **Export** to save the amplifier correction curve outside the project as an *.xls or *.csv file.

Click **OK** to leave the dialog. You will be prompted to name the amplifier correction curve, which will then be attached to the project.

Impedance Magnitude



This result provides a graphical display of the impedance response of the driver, plotting the complex impedance magnitude versus frequency. The model fit (Standard model) curve is also plotted on the same graph, shown here as a dashed line.

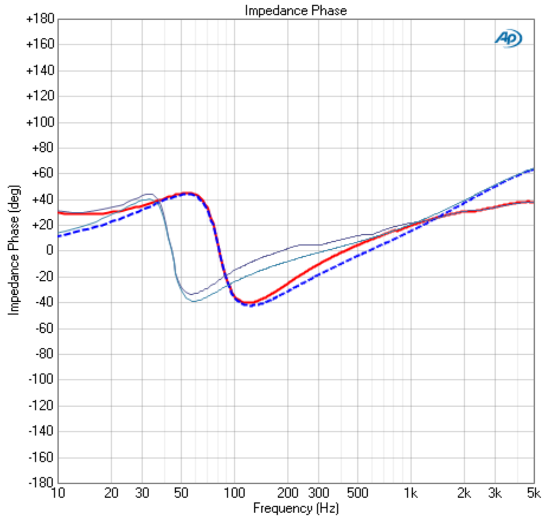
If a second measurement is made using the **Added Mass** or **Known Volume** driver loads, the loaded trace and its model fit are plotted on the same graph. The example here (and in the other results) was made by adding mass to the driver, producing the traces with the lower magnitude and lower resonance frequency.

Units

Units available for Impedance/Thiele-Small Impedance Magnitude are:

X-axis	*Y-axis*
• Hz	• ohms
• dHz	
• F/R	
• %Hz	

Impedance Phase



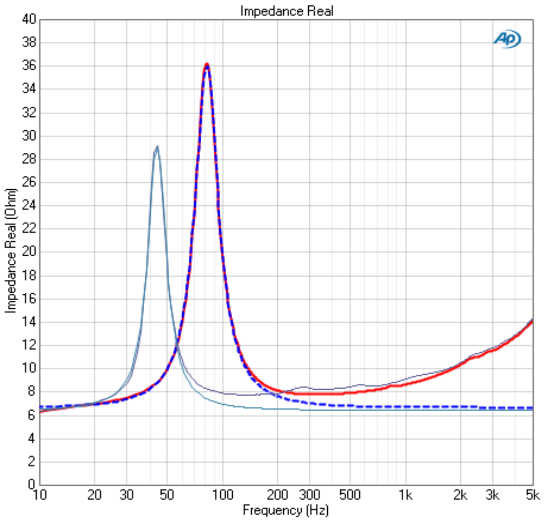
This result provides a graphical display of the impedance phase response of the driver, plotting the impedance phase versus frequency.The dashed line is the model fit trace, here using the Standard model. Traces for **Added Mass** are also shown.

Units

Units available for Impedance/Thiele-Small Impedance Phase are:

X-axis	*Y-axis*
• Hz	• deg
• dHz	• rad
• F/R	
• %Hz	

Impedance Real



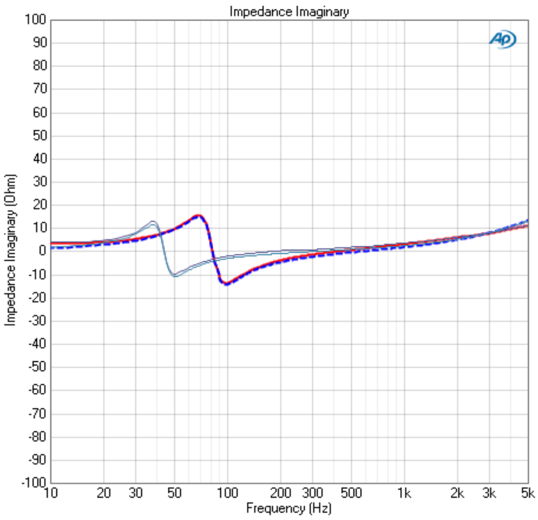
Driver impedance is a complex number, with both real and imaginary components. This result provides a graphical display of the real component of the impedance response of the driver, plotting the real impedance value versus frequency. The dashed line is the model fit trace, here using the Standard model. Traces for **Added Mass** are also shown.

Units

Units available for Impedance/Thiele-Small Impedance Real are:

X-axis	*Y-axis*
• Hz	• ohms
• dHz	
• F/R	
• %Hz	

Impedance Imaginary



Driver impedance is a complex number, with both real and imaginary components. This result provides a graphical display of the imaginary component of the impedance response of the driver, plotting the imaginary impedance value versus frequency. The dashed line is the model fit trace, here using the Standard model. Traces for **Added Mass** are also shown.

Units

Units available for Impedance/Thiele-Small Impedance Magnitude are:

X-axis	*Y-axis*
• Hz	• ohms
• dHz	
• F/R	
• %Hz	

Analysis: Thiele-Small Parameters

This result populates a table with Thiele-Small parameter values calculated from the impedance response measurement. If you have elected to use the **Known M_{MD}** mode, or if you have made the second measurement in **Added Mass** or **Known Volume** mode, this will be the full set of T-S parameters.

If the resonant frequency of the driver has not moved down (for **Added Mass**) or up (for **Known Volume**) at least 5%, an error message will be shown, suggesting that you add more mass, or reduce the enclosure volume.

Otherwise, a single free-air impedance response measurement will only populate a subset of the Thiele-Small parameters.

Parameter	Value	Lower Limit	Upper Limit
F_S	50.50 Hz	----	----
Q_{MS}	2.07	----	----
Q_{ES}	0.56	----	----
Q_{TS}	0.44	----	----
S_D	452.39 cm^2	----	----
R_E	5.51 Ω	----	----
L_E	0.38 mH	----	----
R_2	---- Ω	----	----
L_2	---- mH	----	----
K_{rm}	----	----	----
E_{rm}	----	----	----
K_{xm}	----	----	----
E_{xm}	----	----	----
R_{MS}	14.85 Ns/m	----	----
C_{MS}	---- mm/N	----	----
M_{MS}	---- g	----	----
V_{AS}	---- l	----	----
Bl	---- Tm	----	----
η_o	---- %	----	----

Thiele-Small parameters (partial)

The parameters C_{MS}, M_{MS}, V_{AS}, *Bl*, and η_0 are not yet populated. Also, R_2 and L_2 are only populated when the LR-2 model is selected, and E_{rm}, K_{rm}, E_{xm} and K_{xm} are only populated when the Wright model is selected.

Parameter	Value	Lower Limit	Upper Limit
F_S	50.50 Hz	----	----
Q_{MS}	2.07	----	----
Q_{ES}	0.56	----	----
Q_{TS}	0.44	----	----
S_D	452.39 cm^2	----	----
R_E	5.51 Ω	----	----
L_E	0.38 mH	----	----
R_2	---- Ω	----	----
L_2	---- mH	----	----
K_{rm}	----	----	----
E_{rm}	----	----	----
K_{xm}	----	----	----
E_{xm}	----	----	----
R_{MS}	14.85 Ns/m	----	----
C_{MS}	0.79 mm/N	----	----
M_{MS}	12.63 g	----	----
V_{AS}	228.30 l	----	----
Bl	6.27 Tm	----	----
η_o	4.99 %	----	----

Thiele-Small parameters (complete)

R_2 and L_2 are only populated when the LR-2 model is selected, and E_{rm}, K_{rm}, E_{xm} and K_{xm} are only populated when the Wright model is selected.

Setting Limits for Thiele-Small parameters

Parameter	Value	Lower Limit	Upper Limit
F_S	55.37 Hz	53.0	56.0
Q_{MS}	2.27	2.00	2.25
Q_{ES}	0.63	----	----
Q_{TS}	0.50	----	----

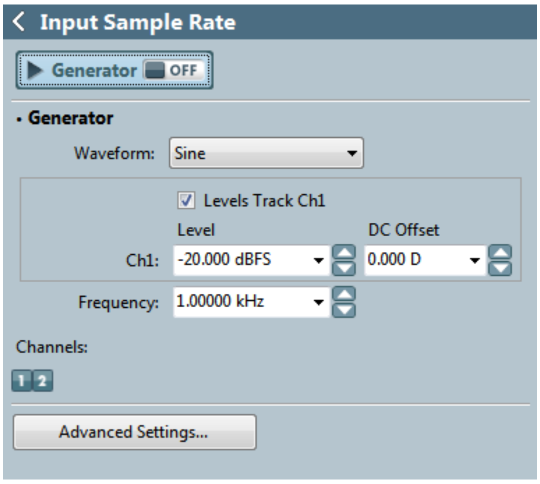
Thiele-Small parameters with limits applied (detail)

You can set arbitrary limits for any or all of the Thiele-Small parameters by entering limit values into the results grid. Results that fall outside of the limits will be highlighted in the grid, and in a sequence the measurement will be flagged as **Failed**.

48

Input Sample Rate
(Sequence Mode)

The Input Sample Rate measurement provides a single-value meter result, the measured sample rate of the incoming digital audio signal. Signal Path Setup Input/Output must be set to a digital input.



Generator

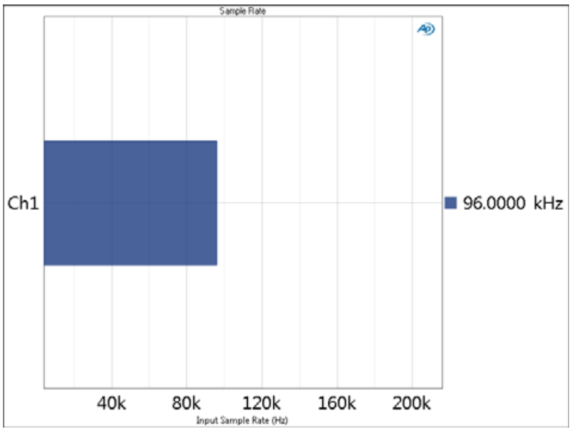
See Chapter 5 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

Advanced Settings

Advanced Settings makes available individual channel output level and offset controls, and input ranging and settling controls. See Advanced Settings for single value measurements on page 317.

See Chapter 98 for more information about units of measurement.

Sample Rate



Input Sample Rate provides a single-value meter result, displaying of the measured sample rate at the digital receiver.

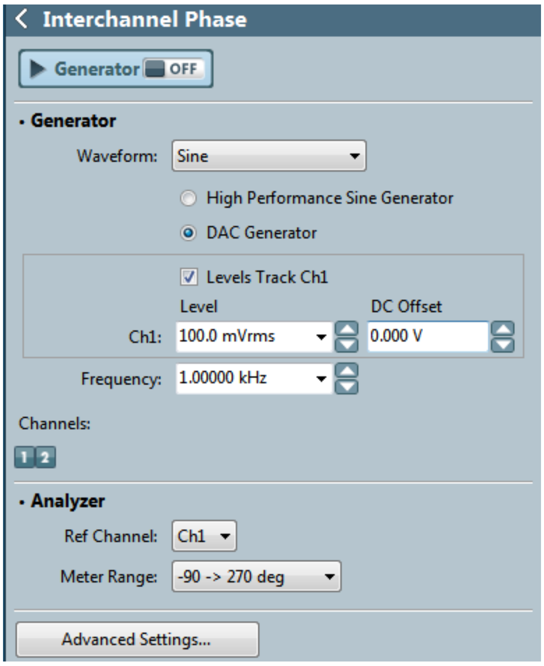
Units

Units available for Input Sample Rate are

- Hz
- dHz
- F/R
- %Hz

49

Interchannel Phase
(Sequence Mode)



The Interchannel Phase measurement provides a single-value result that shows the relative phase of the DUT channels at a single frequency. One channel is chosen as the phase reference channel, and the remaining channels are measured against it. For more information about phase, see page 300.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 6 for information about Signal Path Setup.

Generator

The typical configuration for this measurement is the closed-loop configuration, using the APx generator as a stimulus. Alternatively, you can make open-loop measurements, using either the External Source or File Input configuration.

See Chapter 5 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 49 for a comparison of features.

Running the measurement

To measure Interchannel Phase, click the Generator button to On. The generator will output the test signal to the DUT on the selected generator channels. Read the Interchannel Phase for each channel from the meter bar display. Channel phase is relative to the selected Reference Channel.

Analyzer

No local low pass, high pass or weighting filters are available for this measurement. However, low pass, high pass and weighting filter settings made in Signal Path Setup > Input/Output will affect this measurement.

Ref Channel

The phase of each channel is measured against the selected Reference Channel. There is no phase result for the Reference Channel itself, and no data is displayed for that channel.

Meter Range

You can set the phase meter to examine the phase results in one of three meter ranges:

0 > 360 deg

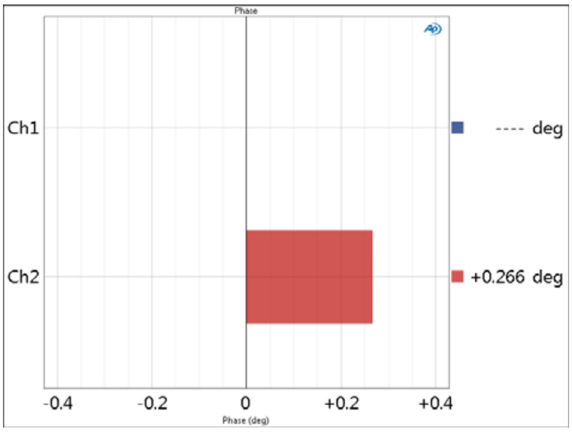
–90 > 270 deg

–180 > 180 deg

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, click **Advanced Settings**. See Advanced Settings for single value measurements on page 317.

Phase



Phase provides a single-value meter result, displaying the phase between the selected reference channel and the other input channels.

Units

Units available for Interchannel Phase are

- deg
- rad

See Chapter 98 for more information about units of measurement.

Measuring Interchannel Phase when configured for External Source

Operation with an external source is slightly different from the operation described above, which uses the internal generator. First, be sure you are configured for External Source. Go to Signal Path Setup and select **None (External)** for the output connector in Output Configuration.

When measuring from an external source, instead of using the generator you will play an audio signal from the DUT. For Interchannel Phase, the audio signal must be periodic and have two or more channels. See External Source Test Signals on page 150.

Reading the results is the same: the phase difference for each channel is shown on the meter bar display. Channel phase is relative to the selected Reference Channel. The meters will read interchannel phase as long as the external source plays the signal.

See Chapter 5 for more information about External Source measurements.

Running the measurement in File Input configuration

See Analyzing Audio Files in Chapter 19 for more information about File Input configuration.

If you have chosen **File** for an input configuration in Signal Path Setup, first click the **File List...** button to add input file(s) to a list for analysis.

If your DUT is to be recording the WAV file, select a generator waveform and turn the generator **ON**, as described above. Configure your DUT to save a WAV file(s) whose file name(s) and location match the name specified in the File List.

If you are measuring an existing file (not recording from the generator but working in External Source), be sure that the file name(s) and location match the name(s) specified in the File List.

Finally, click the **Analyze** button. The first settled reading (typically within the first 0.10 second of the file) is shown on the meter bar display. To take a reading later in the file, go to the **Settling** tab in **Advanced Settings** and increase the **Delay Time** to the desired point in the file.

More about Phase

The acoustic vibrations that are sound and the electrical vibrations that represent sound as audio signals are waves.

At a given moment in time, two waves of the same frequency can be at the same point in their wave cycle (in phase) or at some different point in the cycle (out of phase). Waves are referenced to circular motion, and a complete cycle of a wave is 360 degrees of angle. 0 degrees, 360 degrees and any integer multiple of 360 all represent the same point on a circle, and all describe signals that are in phase.

Differences in phase may be observed between any two signals. In audio testing, phase is typically measured between channels in multichannel devices, and between the input and output of a device under test.

50

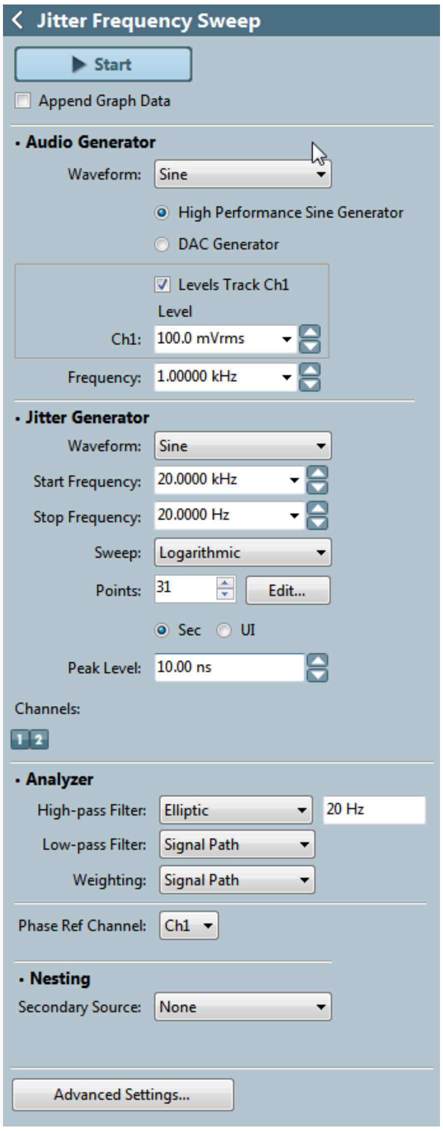
Jitter Frequency Sweep (Sequence Mode)

Jitter measurements require an analyzer fitted with the Advanced Master Clock (AMC), standard in the APx555 and an option for APx52x and APx58x. The analyzer must also be fitted with a jitter-enabled I/O module, such as ADIO or DSIO. Output configuration must be set to a jitter-enabled output.

Results for Jitter Frequency Sweep include:

• RMS Level	• Deviation	• THD Level
• Gain*	• Phase	• Distortion Product Ratio
• Relative Level	• THD+N Ratio	• Distortion Product Level
• Peak Level	• THD+N Level	• SINAD
• Average Jitter Level*	• THD Ratio	

* These results will only be populated if the analyzer input is set to measure the jitter signal. See Measuring Audio or Jitter, below.



Overview

The Jitter Frequency Sweep stimulates the DUT with a digital signal. The embedded audio waveform is selected in Audio Generator. The timebase for the digital signal is modulated by the jitter generator, which is swept across a frequency range.

The results are either

- measurements of the audio output of the DUT (analog or digital), which is typically degraded when the timebase is jittered; or
- for DUTs with a digital output, direct measurements of the jitter in the DUT output signal, rather than the embedded audio signal. Select **Audio** or **Jitter** in Signal Path Setup > Input/Output > Input Configuration.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 6 for information about Signal Path Setup.

Additionally, you must choose where to apply the jitter, and whether to measure the audio or the jitter on the incoming signal, as described next.

Applying jitter

Go to the Clocks panel in Signal Path Setup, and select whether you would like to apply the jitter sweep to a digital output, or to the Reference Out and Sync Out connectors.

Select Apply To:

- Digital Output
  If you select Digital Output, you must also go to Signal Path Setup > Input/Output and select a digital output that supports jitter generation.
- Reference/Sync
  If you select Reference/Sync, you must connect the Reference Out or the Sync Out to the appropriate sync input on your DUT.

For a jitter sweep, you do not need to Enable jitter on the Clocks panel. Jitter is automatically enabled for the duration of the sweep.

Measuring Audio or Jitter

For a DUT with a digital output, go to Signal Path Setup > Input/Output to choose a digital input for the APx analyzer.

Select Measure:

- Audio
  Audio de-embeds the audio signal from the digital interface signal and routes it to the analyzer. This is normal APx operation.
- Jitter (UI) or Jitter (sec)
  Jitter routes the digital interface signal to jitter demodulator and a dedicated ADC (analog to digital converter), and then to the analyzer.

For a default sweep of 20 kHz to 20 Hz in 31 logarithmic steps, simply click Start. If you want to specify other parameters for your sweep, review Generator and Analyzer settings, discussed below. Click Start when you have approved the settings.

Append Graph Data

Normally, the current graph data for all results is deleted each time you start a new measurement. If the Append box is checked, the current data is kept in memory as a Data Set (see page 165), and the new measurement data is appended as a new data set. You can append many Data Sets.

Audio Generator

The Audio Generator generates the audio signal that is embedded in the digital interface signal, as in most APx measurements.

Waveform, Frequency and Level

The configuration for this measurement must be the closed-loop configuration, using the APx generator as a stimulus.

See Chapter 5 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings.

Jitter Generator

The Jitter Generator generates the signal that modulates the jitter clock, which is used as the timebase for the digital interface signal used in this measurement.

Waveform

Select Sine or Square.

Start Frequency

Set the sweep start frequency here. The available range is 2 Hz to 200 kHz.

Stop Frequency

Set the sweep stop frequency here. The available range is 2 Hz to 200 kHz.

Sweep

Choose

- Logarithmic (logarithmic point spacing)
- Linear (linear point spacing)
- Custom
  Edit sweep spacing and number of Points to create a Custom sweep.

Points

Set the number of sweep points here. The default is 31 points.

Points > Edit

Open the Sweep Points dialog to edit, import or export the Sweep Points table.

Step Size

For linear sweeps, enter the Step Size here.

Peak Level

Set the jitter level here. Note that the level is expressed in peak units of time, seconds or UI, selectable below the Peak Level field.

Seconds / UI

Choose the jitter units here.

Channels

Toggle specific output channels on or off by clicking on the **Channel** number button.

EQ

The generated jitter signal can be equalized. Go to Signal Path Setup > **Clocks** > **Jitter Generator** to apply an EQ curve. See the Jitter Generator topic in **Clocks**. beginning on page 60.

Analyzer

Note that for this measurement, the analyzer can measure either the audio signal (de-embedded from the interface signal) or the jitter (demodulated from the interface signal), as selected in Signal Path Setup > **Input/Output**.

Filters

Note that these filters will affect the audio signal measurements or the jitter signal measurements, as selected in Signal Path Setup > Input/Output.

Local high-pass, low-pass and weighting filters are available for this measurement. The low-pass and high-pass filters, if selected, are applied to the entire measured signal and affect all the results. The weighting filter, if selected, affects only the distortion and distortion plus noise results. See Measurement Filters on page 547 for detailed information about the filters available locally for some measurements.

Low pass, high pass and weighting Signal Path Filters settings (see page 545) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass or high-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active. A local weighting filter setting does not override any Signal Path filter settings; the local weighting filter is applied in addition to any active Signal Path filters.

Phase Ref Channel

Interchannel Phase measurements must be referenced to one channel. Set the phase reference channel here.

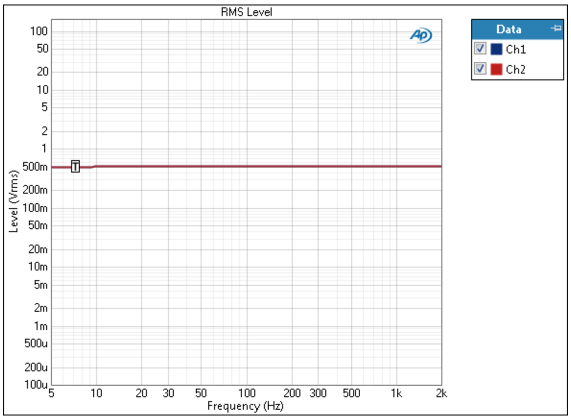
Nesting

Jitter Frequency sweeps can be nested. Read about Nested Sweeps beginning on page 161.

Advanced Settings

If your test required special adjustments or settings, click Advanced Settings.

RMS Level results



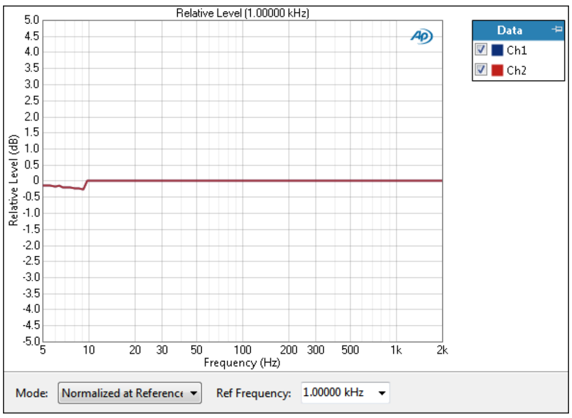
Jitter Frequency Sweep: RMS Level result uses a fixed audio stimulus signal while the jitter generator is moved across a range of frequencies in a specified number of points.

Units

Units available for Jitter Frequency Sweep: RMS Level results are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*	*Jitter signals*
• Hz	• Vrms	• dBFS	• UI
• dHz	• dBV	• FS	• dBUI
• F/R	• dBu	• %FS	• s
• %Hz	• dBrA	• dBrA	
	• dBrB	• dBrB	
	• dBSPL1	• dBSPL1	
	• dBSPL2	• dBSPL2	
	• dBm		
	• W (watts)		

Jitter Frequency Sweep: Relative Level



The Jitter Frequency Sweep: Relative Level result uses a fixed audio stimulus signal while the jitter generator is moved across a range of frequencies in a specified number of points. Additionally, in this result the response curve is plotted in relation to the measured level at a selected jitter frequency.

Mode

Select one of two result display modes.

Normalized at Reference

Enter a reference frequency for normalization in the **Ref Frequency** field. The measured level at the jitter frequency you choose is set as 0 dB on the graph, and the DUT response is plotted in relation to this reference. You can change the **Ref Frequency** at any time (except after appending) and the graph will immediately be redrawn to reflect the new setting.

Auto-Centered in Limits

This control is only usable if upper and lower limits are set on this graph. In some applications, the general conformance of a trace to the shape of a set of limits is of importance, with absolute level less important or irrelevant. **Auto-Centered in Limits** centers the data trace relative to the upper and lower limits. To exclude possible exceptional data points at frequency extremes, the frequency range that is considered for the centering operation is constrained by the values in the **Min** and **Max** fields.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Ref Frequency

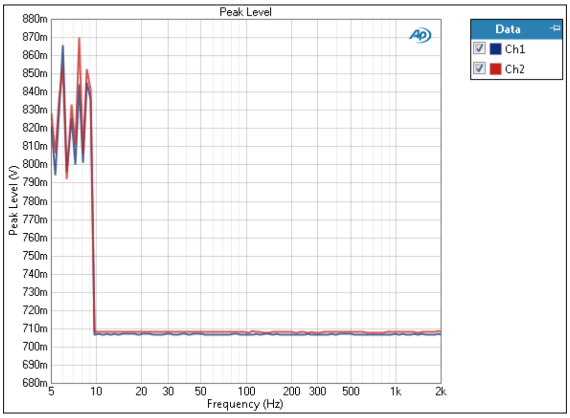
Choose a Reference Frequency here.

Units

Units available for Jitter Frequency Sweep: Relative Level are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Peak Level



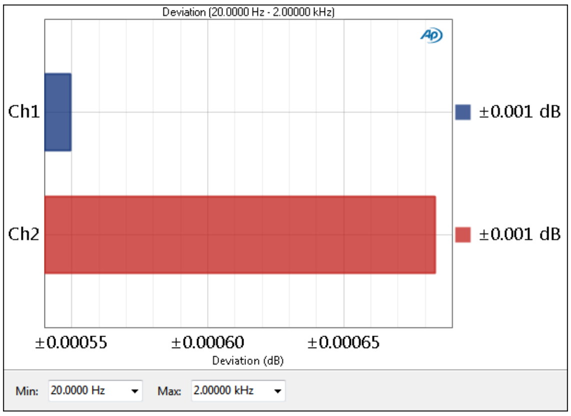
Jitter Frequency Sweep: Peak Level result uses a fixed audio stimulus signal while the jitter generator is moved across a range of frequencies in a specified number of points. This is a frequency response sweep, with peak-scaled results.

Units

Units available for Jitter Frequency Sweep Peak Level results are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*	*Jitter*
• Hz	• V	• D	• UI
• dHz		• hex	• dBUI
• F/R			• s
• %Hz			

Deviation



The Jitter Frequency Sweep: Deviation result uses a fixed audio stimulus signal while the jitter generator is moved across a range of frequencies in a specified number of points. This result is a single value measurement computed from the stepped sweep that shows the level deviation (the total range of level variation) of each channel as a meter bar. You can specify a minimum and maximum frequency to define the range of the deviation measurement.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Min

Set the minimum frequency of the range of interest here.

Max

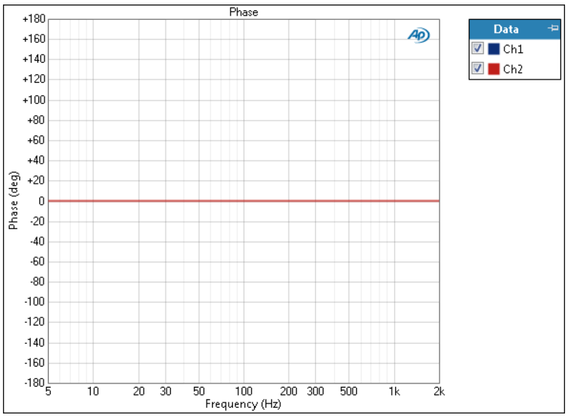
Set the maximum frequency of the range of interest here.

Units

Units available for Jitter Frequency Sweep: Deviation are

- dB

Phase



The Phase result uses a fixed audio stimulus signal while the jitter generator is moved across a range of frequencies in a specified number of points. One channel is chosen as the phase reference channel, and the remaining channels are measured against it.

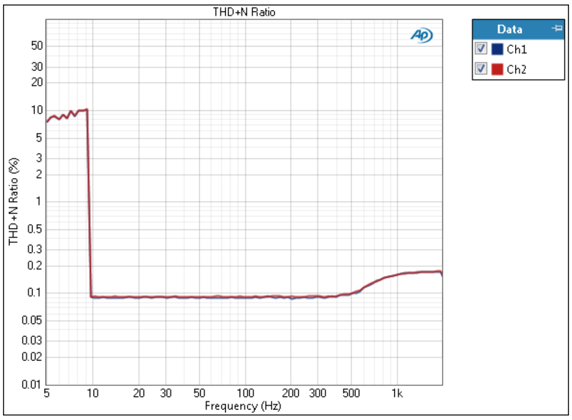
Select a phase reference channel in **Phase: Ref Channel**. The remaining channels are plotted against the selected channel.

Units

Units available for Jitter Frequency Sweep: Phase are:

X-axis	*Y-axis*
• Hz	• deg
• dHz	• rad
• F/R	
• %Hz	

THD+N Ratio



The THD+N Ratio result uses a fixed audio stimulus signal while the jitter generator is moved across a range of frequencies in a specified number of points. In this result the ratio of the level of the THD+N (total harmonic distortion plus noise) to the total signal in the DUT output is plotted against frequency.

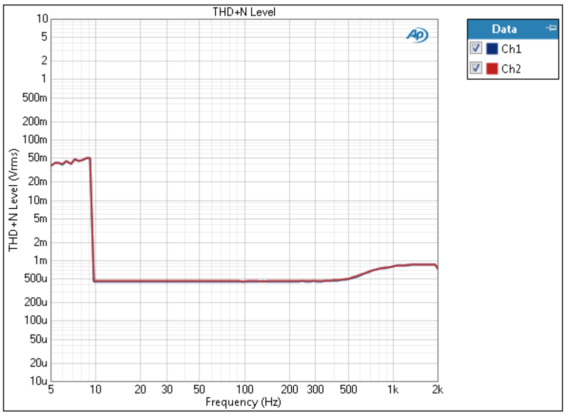
See page 475 for more information about THD+N.

Units

Units available for Jitter Frequency Sweep: THD+N Ratio are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

THD+N Level



The THD+N Level result uses a fixed audio stimulus signal while the jitter generator is moved across a range of frequencies in a specified number of points. In this result the level of the THD+N (total harmonic distortion plus noise) in the DUT output is plotted against frequency.

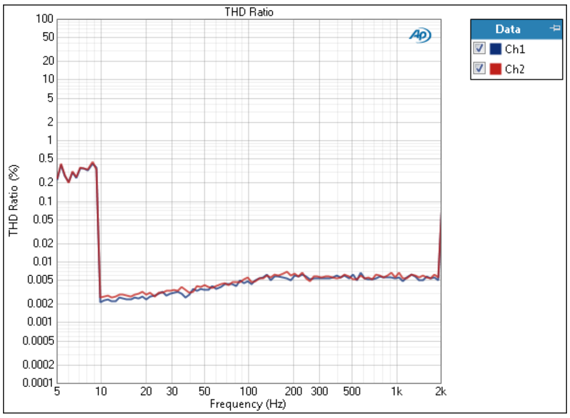
See page 475 for more information about THD+N.

Units

Units available for Jitter Frequency Sweep: THD+N Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*	*Jitter signals*
• Hz	• Vrms	• dBFS	• UI
• dHz	• dBV	• FS	• dBUI
• F/R	• dBu	• %FS	• s
• %Hz	• dBrA	• dBrA	
	• dBrB	• dBrB	
	• dBSPL1	• dBSPL1	
	• dBSPL2	• dBSPL2	
	• dBm		
	• W (watts)		

THD Ratio



The THD Ratio result uses a fixed audio stimulus signal while the jitter generator is moved across a range of frequencies in a specified number of points. In this result the ratio of the level of the THD (total harmonic distortion) to the total signal in the DUT output is plotted against frequency.

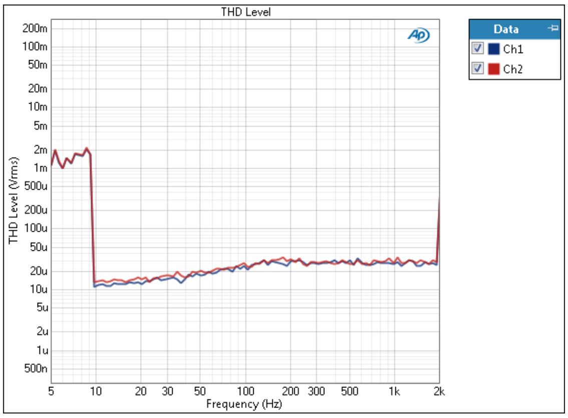
See page 475 for more information about THD.

Units

Units available for Jitter Frequency Sweep: THD Ratio are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

THD Level



The THD Level result uses a fixed audio stimulus signal while the jitter generator is moved across a range of frequencies in a specified number of points. In this result the level of the THD (total harmonic distortion) in the DUT output is plotted against frequency.

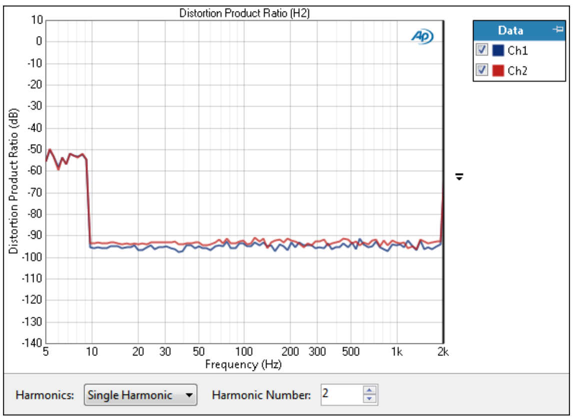
See page 475 for more information about THD.

Units

Units available for Jitter Frequency Sweep: THD Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*	*Jitter signals*
• Hz	• Vrms	• dBFS	• UI
• dHz	• dBV	• FS	• dBUI
• F/R	• dBu	• %FS	• s
• %Hz	• dBrA	• dBrA	
	• dBrB	• dBrB	
	• dBSPL1	• dBSPL1	
	• dBSPL2	• dBSPL2	
	• dBm		
	• W (watts)		

Distortion Product Ratio



The Jitter Frequency Sweep Distortion Product Ratio result uses a fixed audio stimulus signal while the jitter generator is moved across a range of frequencies in a specified number of points. In this result the ratio of the level of the selected harmonic distortion product to the fundamental in the DUT output is plotted against frequency.

Select the harmonic distortion product(s) you wish to view using the **Harmonics** controls, described below.

Additional Controls for Distortion Product Ratio

Harmonics

For a graph of the ratio of one specific harmonic product to the signal level, select **Single Harmonic** and set the **Harmonic Number** to your choice, from 2nd harmonic to 20th harmonic.

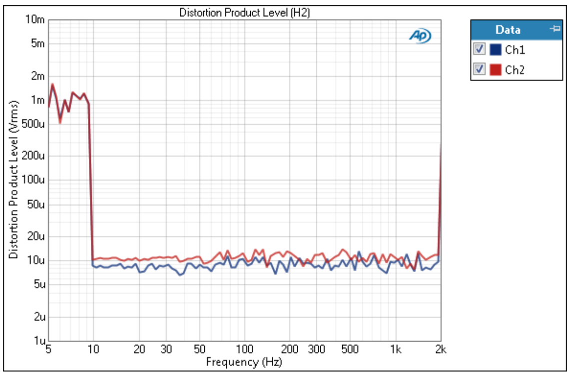
For a graph of the sum of the levels of any combination of harmonic products (from **H2** through **H20**), divided by the signal level, select **Sum of Harmonics**. Then click **Select Harmonics...** to open a result definition dialog.

Units

Units available for Jitter Frequency Sweep Distortion Product Ratio are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Distortion Product Level



The Jitter Frequency Sweep Distortion Product Level result uses a fixed audio stimulus signal while the jitter generator is moved across a range of frequencies in a specified number of points. In this result the level of the selected harmonic distortion product in the DUT output is plotted against frequency.

Select the harmonic distortion product(s) you wish to view using the **Harmonics** controls, described below.

Additional Controls for Distortion Product Level

Harmonics

For a graph of the level of one specific harmonic product, select **Single Harmonic** and set the **Harmonic Number** to your choice, from 2nd harmonic to 20th harmonic.

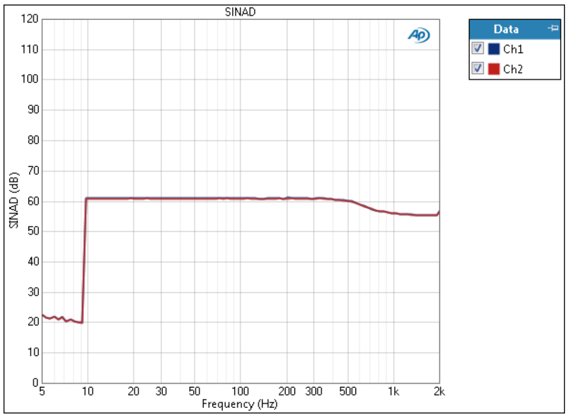
For a graph of the level of the sum of several or all harmonic products (from **H2** through **H20**), select **Sum of Harmonics**, then click **Select Harmonics...** to open a result definition dialog.

Units

Units available for Jitter Frequency Sweep Distortion Product Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*	*Jitter signals*
• Hz	• Vrms	• dBFS	• UI
• dHz	• dBV	• FS	• dBUI
• F/R	• dBu	• %FS	• s
• %Hz	• dBrA	• dBrA	
	• dBrB	• dBrB	
	• dBSPL1	• dBSPL1	
	• dBSPL2	• dBSPL2	
	• dBm		
	• W (watts)		

SINAD



The Stepped Level Sweep: SINAD result uses a fixed audio stimulus signal while the jitter generator is moved across a range of frequencies in a specified number of points. X axis is generator level; Y axis is DUT output SINAD ratio.

Units

Units available for SINAD are:

X-axis (analog)	*X-axis (digital)*	*Y-axis*
• Vrms	• dBFS	• x/y
• Vp	• FS	• dB
• Vpp	• %FS	
• dBV	• dBrG	
• dBu		
• dBrG		
• dBm		
• W (watts)		

51

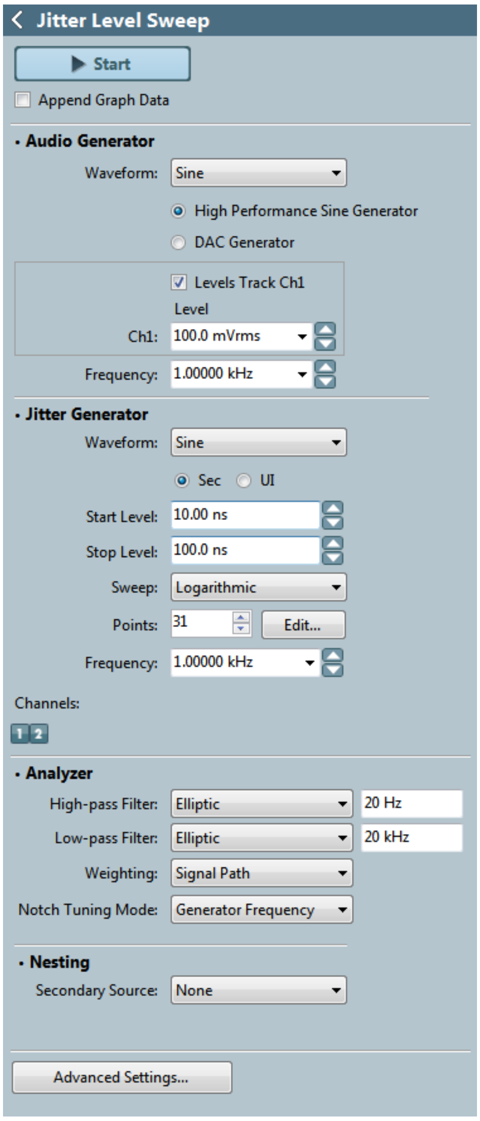
Jitter Level Sweep
(Sequence Mode)

Jitter measurements require an analyzer fitted with the Advanced Master Clock (AMC), standard in an APx555 and an option for APx525, 526, 582, 585, and 586. The analyzer must also be fitted with a jitter-enabled I/O module, such as ADIO or DSIO. Output configuration must be set to a jitter-enabled output.

Results for Jitter Level Sweep include:

• Level	• Average Jitter Level*	• THD Level
• Gain*	• THD+N Ratio	• SINAD
• Linearity*	• THD+N Level	• Distortion Product Level
• Peak Level	• THD Ratio	• Distortion Product Ratio

* These results will only be populated if the analyzer input is set to measure the jitter signal. See **Measuring Audio or Jitter**, below.



Overview

The Jitter Level Sweep stimulates the DUT with a digital signal. The embedded audio waveform is selected in Audio Generator. The timebase for the digital signal is modulated by the jitter generator, which is swept across a range of levels.

The results are either

- measurements of the audio output of the DUT (analog or digital), which is typically degraded when the timebase is jittered; or
- for DUTs with a digital output, direct measurements of the jitter in the DUT output signal, rather than the embedded audio signal. Select **Audio** or **Jitter** in Signal Path Setup > Input/Output > Input Configuration.

Operation

If you have not yet configured your signal path, first go to Signal Path Setup (Chapter 6) to select and configure inputs and outputs, references and other settings.

This measurement must be configured as closed-loop configuration, using the APx generator as a stimulus.

Additionally, you must choose where to apply the jitter, and whether to measure the audio or the jitter on the incoming signal, as described next.

Applying jitter

Go to the Clocks panel in Signal Path Setup, and select whether you would like to apply the jitter sweep to a digital output, or to the Reference Out and Sync Out connectors.

Select Apply To:

- Digital Output
  If you select Digital Output, you must also go to Signal Path Setup > Input/Output and select a digital output that supports jitter generation.
- Reference/Sync
  If you select Reference/Sync, you must connect the Reference Out or the Sync Out to the appropriate sync input on your DUT.

For a jitter sweep, you do not need to Enable jitter on the Clocks panel. Jitter is automatically enabled for the duration of the sweep.

Measuring Audio or Jitter

For a DUT with a digital output, go to Signal Path Setup > Input/Output to choose a digital input for the APx analyzer.

Select Measure:

- Audio
  Audio de-embeds the audio signal from the digital interface signal and routes it to the analyzer. This is normal APx operation.
- Jitter (UI) or Jitter (sec)
  Jitter routes the digital interface signal to jitter demodulator and a dedicated ADC (analog to digital converter) and to the analyzer.

For a default sweep of 20 kHz to 20 Hz in 31 logarithmic steps, simply click Start. If you want to specify other parameters for your sweep, review Generator and Analyzer settings, discussed below. Click Start when you have approved the settings.

Append Graph Data

Measured data are grouped in a Data Set, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement.

If the Append Graph Data box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 573.

Audio Generator

The Audio Generator generates the audio signal that is embedded in the digital interface signal, as in most APx measurements.

Waveform, Frequency and Level

See Chapter 5 for information about the using the APx Generator and setting Waveform, Level and Frequency.

Jitter Generator

The Jitter Generator generates the signal that modulates the jitter clock, which is used as the timebase for the digital interface signal used in this measurement.

Waveform

Select Sine, Square or Noise.

Start Level

Set the sweep start level here. The available range is 0 s to 1.592 µs, peak. Default is 10.00 ns.

Stop Level

Set the sweep stop level here. The available range is 0 s to 1.592 µs, peak. Default is 100.00 ns.

Seconds / UI

Choose the jitter units here.

Sweep

Choose

- Logarithmic (logarithmic point spacing)
- Linear (linear point spacing)
- Custom
  Edit sweep spacing and number of Points to create a Custom sweep.

Points

Set the number of sweep points here. The default is 31 points.

Points > Edit

Open the Sweep Points dialog to edit, import or export the Sweep Points table.

Step Size

For linear sweeps, enter the Step Size here.

Frequency

Set the jitter frequency here.

Seconds / UI

Choose the jitter units here.

Channels

Toggle specific output channels on or off by clicking on the **Channel** number button.

Analyzer

Note that for this measurement, the analyzer can measure either the audio signal (de-embedded from the interface signal) or the jitter (demodulated from the interface signal), as selected in Signal Path Setup > Input/Output.

Filters

Note that these filters will affect the audio signal measurements or the jitter signal measurements, as selected in Signal Path Setup > Input/Output.

Local high-pass, low-pass and weighting filters are available for this measurement. The low-pass and high-pass filters, if selected, are applied to the entire measured signal and affect all the results. The weighting filter, if selected, affects only the distortion and distortion plus noise results. See Measurement Filters on page 547 for detailed information about the filters available locally for some measurements.

Low pass, high pass and weighting Signal Path Filters settings (see page 545) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass or high-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active. A local weighting filter setting does not override any Signal Path filter settings; the local weighting filter is applied in addition to any active Signal Path filters.

Notch Tuning Mode

For the harmonic distortion results, a tunable notch filter is used to remove the fundamental from the measurement. The tuning can be

- **Generator Frequency**
  The current APx audio generator frequency.
- **Jitter Generator Frequency**
  The APx jitter generator frequency for the current step.
- **Measured Frequency**
  The measured frequency for the current step. When the analyzer channels are receiving different frequencies, the notch filter for each channel is centered on the frequency in that channel.
- **Fixed Frequency**
  A fixed frequency selected by the user. When **Fixed Frequency** is selected, a **Filter Frequency:** entry field becomes available beneath the **Notch Tuning Mode** control.

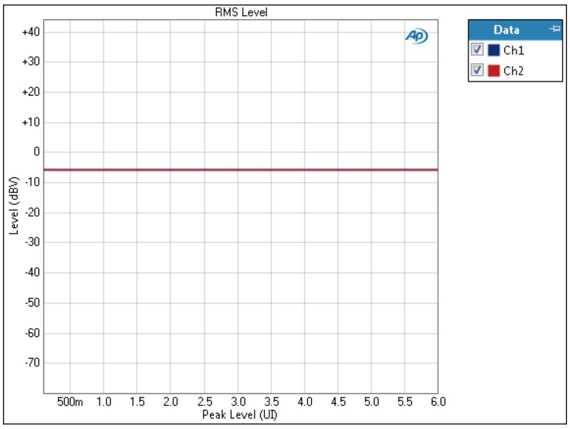
Nesting

Jitter Frequency sweeps can be nested. Read about Nested Sweeps beginning on page 161.

Advanced Settings

If your test required special adjustments or settings, click Advanced Settings.

Jitter Level Sweep: RMS Level



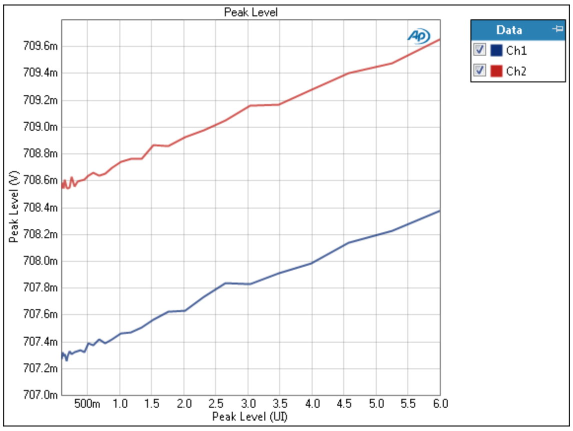
The **Jitter Level Sweep: RMS Level** result uses a fixed audio stimulus signal while the jitter generator is moved across a range of levels in a specified number of points. X axis is generator level; Y axis is DUT output level.

Units

Units available for Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis (analog)*	*Y-axis (digital)*
• Vrms	• dBFS	• Vrms	• dBFS
• Vp	• FS	• dBV	• FS
• Vpp	• %FS	• dBu	• %FS
• dBV	• dBrG	• dBrA	• dBrA
• dBu		• dBrB	• dBrB
• dBrG		• dBSPL1	• dBSPL1
• dBm		• dBSPL2	• dBSPL2
• W (watts)		• dBm	
		• W (watts)	

Peak Level



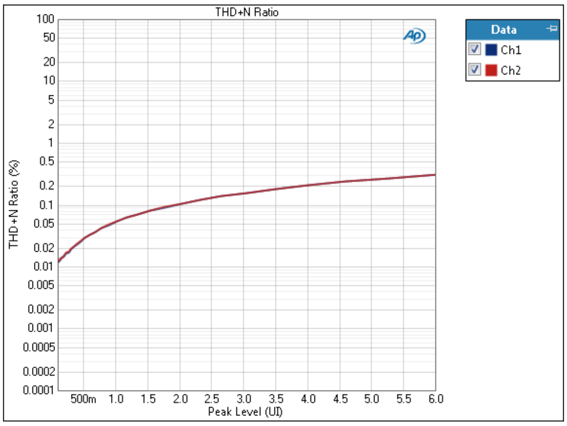
The **Jitter Level Sweep: Peak Level** result uses a fixed audio stimulus signal while the jitter generator is moved across a range of levels in a specified number of points. X axis is generator level; Y axis is the band-pass filtered DUT output level, scaled in peak units.

Units

Units available for Peak Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis (analog)*	*Y-axis (digital)*
• Vrms	• dBFS	• V	• D
• Vp	• FS		
• Vpp	• %FS		
• dBV	• dBrG		
• dBu		*Y-axis (jitter)*	
• dBrG		• UI	
• dBm		• dBUI	
• W (watts)		• s	

Jitter Level Sweep: THD+N Ratio



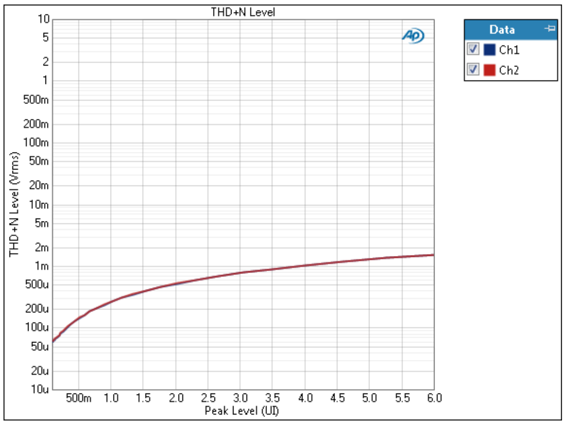
The Jitter Level Sweep: THD+N Ratio result uses a fixed audio stimulus signal while the jitter generator is moved across a range of levels in a specified number of points. X axis is generator level; Y axis is DUT output THD+N ratio.

Units

Units available for THD+N Ratio are:

X-axis (analog)	*X-axis (digital)*	*Y-axis*
• Vrms	• dBFS	• x/y
• Vp	• FS	• %
• Vpp	• %FS	• ppm
• dBV	• dBrG	• dB
• dBu		
• dBrG		
• dBm		
• W (watts)		

Jitter Level Sweep: THD+N Level



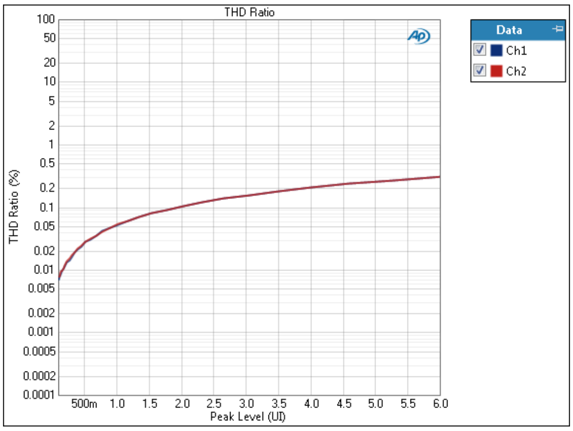
The Jitter Level Sweep: THD+N Level result uses a fixed audio stimulus signal while the jitter generator is moved across a range of levels in a specified number of points. X axis is generator level; Y axis is DUT output THD+N level.

Units

Y-axis units available for THD+N Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis (analog)*	*Y-axis (digital)*
• Vrms	• dBFS	• Vrms	• dBFS
• Vp	• FS	• dBV	• FS
• Vpp	• %FS	• dBu	• %FS
• dBV	• dBrG	• dBrA	• dBrA
• dBu		• dBrB	• dBrB
• dBrG		• dBSPL1	• dBSPL1
• dBm		• dBSPL2	• dBSPL2
• W (watts)		• dBm	
		• W (watts)	

Jitter Level Sweep: THD Ratio



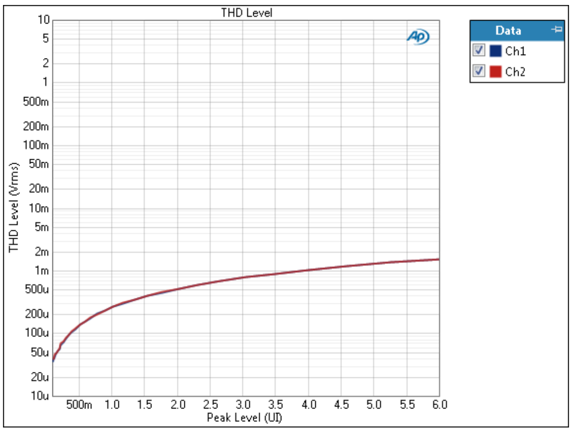
The Jitter Level Sweep: THD Ratio result uses a fixed audio stimulus signal while the jitter generator is moved across a range of levels in a specified number of points. X axis is generator level; Y axis is DUT output THD ratio.

Units

Units available for THD Ratio are:

X-axis (analog)	*X-axis (digital)*	*Y-axis*
• Vrms	• dBFS	• x/y
• Vp	• FS	• %
• Vpp	• %FS	• ppm
• dBV	• dBrG	• dB
• dBu		
• dBrG		
• dBm		
• W (watts)		

Jitter Level Sweep: THD Level



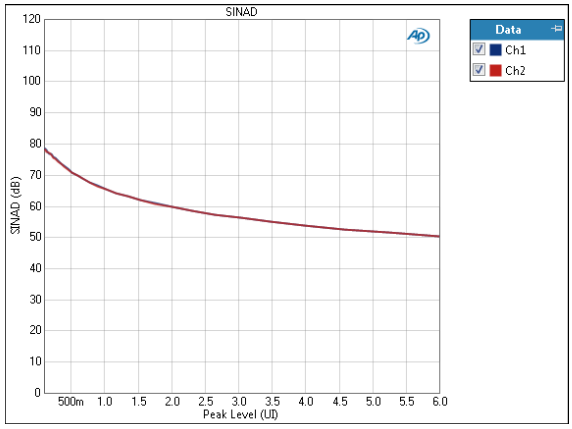
The Jitter Level Sweep: THD Level result uses a fixed audio stimulus signal while the jitter generator is moved across a range of levels in a specified number of points. X axis is generator level; Y axis is DUT output THD level.

Units

Y-axis units available for THD Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis (analog)*	*Y-axis (digital)*
• Vrms	• dBFS	• Vrms	• dBFS
• Vp	• FS	• dBV	• FS
• Vpp	• %FS	• dBu	• %FS
• dBV	• dBrG	• dBrA	• dBrA
• dBu		• dBrB	• dBrB
• dBrG		• dBSPL1	• dBSPL1
• dBm		• dBSPL2	• dBSPL2
• W (watts)		• dBm	
		• W (watts)	

Jitter Level Sweep: SINAD



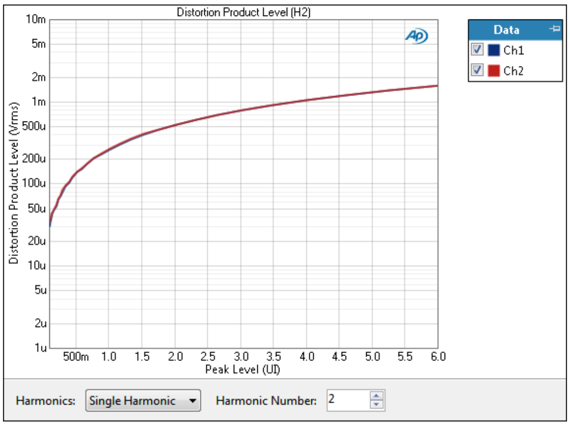
The Jitter Level Sweep: SINAD result uses a fixed audio stimulus signal while the jitter generator is moved across a range of levels in a specified number of points. X axis is generator level; Y axis is DUT output SINAD ratio.

Units

Units available for SINAD are:

X-axis (analog)	*X-axis (digital)*	*Y-axis*
• Vrms	• dBFS	• x/y
• Vp	• FS	• dB
• Vpp	• %FS	
• dBV	• dBrG	
• dBu		
• dBrG		
• dBm		
• W (watts)		

Jitter Level Sweep: Distortion Product Level



The Jitter Level Sweep Distortion Product Level result uses a fixed audio stimulus signal while the jitter generator is moved across a range of levels in a specified number of points. In this result the level of the selected harmonic distortion product in the DUT output is plotted against generator level.

Select the harmonic distortion product(s) you wish to view using the **Harmonics** controls, described below.

Harmonics

For a graph of the level of one specific harmonic product, select **Single Harmonic** and set the **Harmonic Number** to your choice, from 2nd harmonic to 20th harmonic.

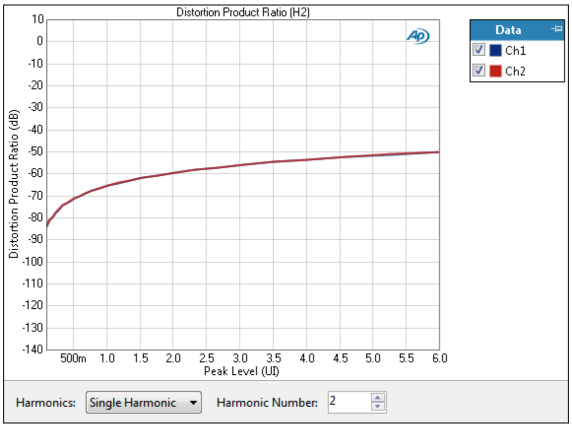
For a graph of the level of the sum of several or all harmonic products (from **H2** through **H20**), select **Sum of Harmonics**, then click **Select Harmonics...** to open a result definition dialog.

Units

Units available for Jitter Level Sweep Distortion Product Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis (analog)*	*Y-axis (digital)*
• Vrms	• dBFS	• Vrms	• dBFS
• Vp	• FS	• dBV	• FS
• Vpp	• %FS	• dBu	• %FS
• dBV	• dBrG	• dBrA	• dBrA
• dBu		• dBrB	• dBrB
• dBrG		• dBSPL1	• dBSPL1
• dBm		• dBSPL2	• dBSPL2
• W (watts)		• dBm	
		• W (watts)	

Jitter Level Sweep: Distortion Product Ratio



The Jitter Level Sweep Distortion Product Ratio result uses a fixed audio stimulus signal while the jitter generator is moved across a range of levels in a specified number of points. In this result the ratio of the level of the selected harmonic distortion product to the fundamental in the DUT output is plotted against generator level.

Select the harmonic distortion product(s) you wish to view using the **Harmonics** controls, described below.

Harmonics

For a graph of the ratio of one specific harmonic product to the signal level, select **Single Harmonic** and set the **Harmonic Number** to your choice, from 2nd harmonic to 20th harmonic.

For a graph of the sum of the levels of any combination of harmonic products (from **H2** through **H20**), divided by the signal level, select **Sum of Harmonics**. Then click **Select Harmonics...** to open a result definition dialog.

Units

Units available for Jitter Level Sweep Distortion Product Ratio are:

X-axis (analog)	*X-axis (digital)*	*Y-axis*
• Vrms	• dBFS	• x/y
• Vp	• FS	• %
• Vpp	• %FS	• ppm
• dBV	• dBrG	• dB
• dBu		
• dBrG		
• dBm		
• W (watts)		

52

Level and Gain
(Sequence Mode)

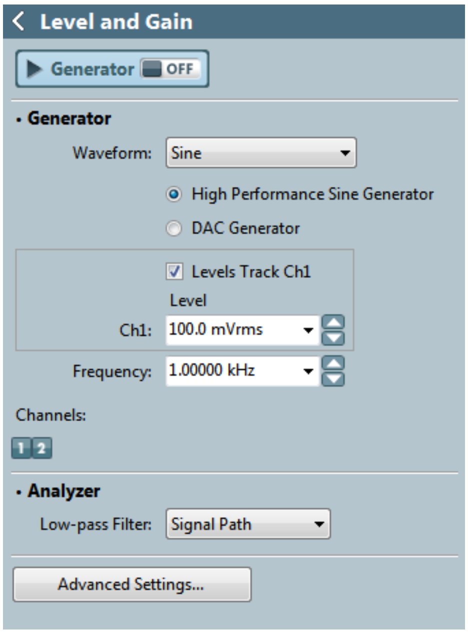
The Level and Gain measurements provide single-value results that show the output level or gain in each DUT channel, as measured at each analyzer input.

Level and Gain results available in APx500 are:

- RMS Level
- Peak Level
- Gain
- Average Jitter Level

Gain is not available in external source configuration, or if a square wave, a generator waveform or File Input has been selected for the measurement.

Average Jitter Level results are only available when Jitter is selected in Signal Path Setup > Input/Output > Measure.



Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 6 for information about Signal Path Setup.

Generator

The typical configuration for this measurement is the closed-loop configuration, using the APx generator as a stimulus. Alternatively, you can make open-loop measurements, using either the External Source or File Input configuration.

See Chapter 5 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 49 for a comparison of features.

Analyzer

This measurement can be configured to measure jitter in Signal Path Setup > Input > Measure. Read detailed information about jitter generation and measurement beginning on page 60.

Low-pass Filter

A local low-pass filter is available for this measurement. The low-pass filter, if selected, is applied to the entire measured signal and affects all the results. See Measurement Filters on page 547 for detailed information about the filters available locally for some measurements.

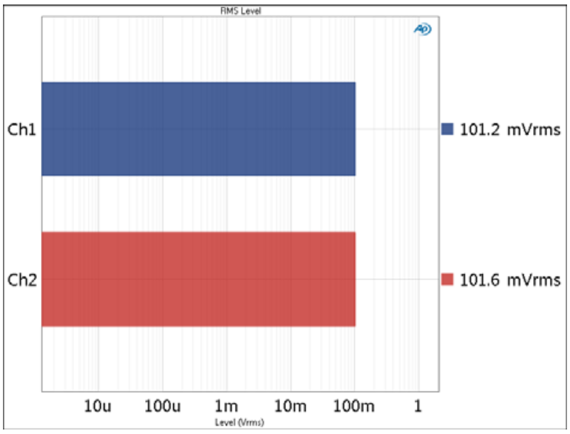
Low pass, high pass and weighting Signal Path Filters settings (see page 545) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active.

Advanced Settings

Advanced Settings makes available individual channel output level and offset controls, and input ranging and settling controls. See Advanced Settings for single value measurements on page 317.

See Chapter 98 for more information about units of measurement.

RMS Level



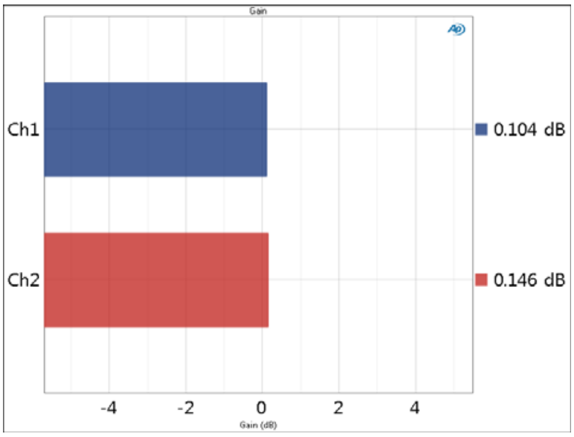
RMS Level provides a single-value meter result, displaying the rms output level from each DUT channel, as measured at each analyzer input.

Units

Units available for RMS Level are

Analog Signals	***Digital Signals***	***Jitter signals***
• Vrms	• dBFS	• UI
• dBV	• FS	• dBUI
• dBu	• %FS	• s
• dBrA	• dBrA	
• dBrB	• dBrB	
• dBSPL1	• dBSPL1	
• dBSPL2	• dBSPL2	
• dBm		
• W (watts)		

Gain



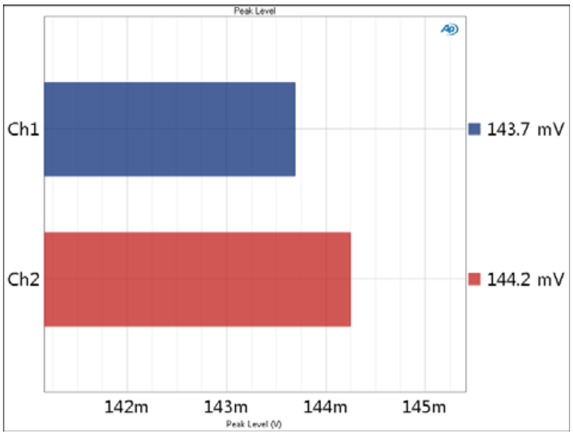
The Gain measurement result provides a single-value measurement of the DUT gain for channel, as measured at each analyzer input.

Units

Units available for Gain are

Same-domain	***Cross-domain***
• x/y	• FS/Vrms
• %	• dB(FS/Vrms)
• ppm	or
• dB	• Vrms/FS
	• dB(Vrms/FS)

Peak Level



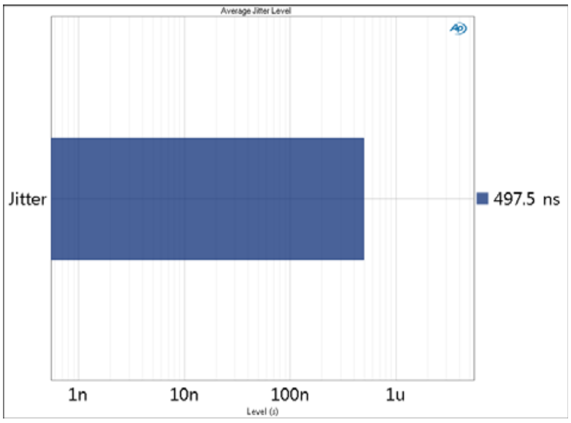
Peak Level provides a single-value meter result, displaying of the peak output level from each DUT channel, as measured at each analyzer input.

Units

Units available for Peak Level are

Analog Signals	***Digital Signals***	***Jitter signals***
• Vrms	• dBFS	• UI
• dBV	• FS	• dBUI
• dBu	• %FS	• s
• dBrA	• dBrA	
• dBrB	• dBrB	
• dBSPL1	• dBSPL1	
• dBSPL2	• dBSPL2	
• dBm		
• W (watts)		

Average Jitter Level



Average Jitter Level provides a single-value meter result, displaying the average jitter level from the digital receiver.

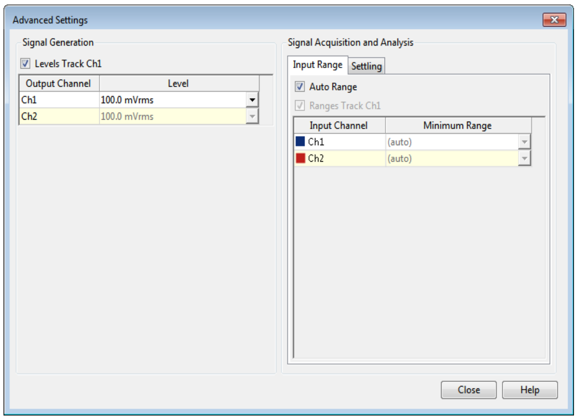
Units

Units available for Average Jitter Level are

Analog Signals	***Digital Signals***	***Jitter signals***
• Vrms	• dBFS	• UI
• dBV	• FS	• dBUI
• dBu	• %FS	• s
• dBrA	• dBrA	
• dBrB	• dBrB	
• dBSPL1	• dBSPL1	
• dBSPL2	• dBSPL2	
• dBm		
• W (watts)		

Advanced Settings for Single Value Measurements

The default settings here are appropriate for most measurements. You may want to make minor adjustments for special situations.



Signal Generation

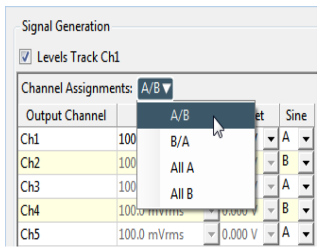
If **Track first channel level** is checked (the default), the generator output level values for channel 1 are copied to the other channels, and output levels for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels.

To set individual generator output channel levels, uncheck the **Track first channel level** checkbox and enter values in the output channel Level fields.

To set individual generator output channel levels, uncheck the “Track first channel level” checkbox and enter values in the output channel Level fields.

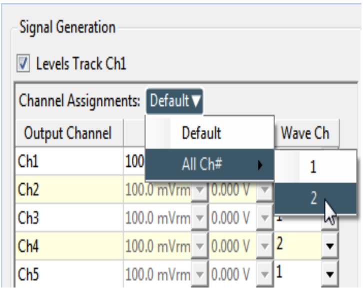
Set Channel Assignments for special waveforms

For Split Sine special waveform generation, you can set channel mapping. By default, channel A audio is mapped to odd numbered output channels, channel B audio to even numbered output channels. These assignments can be remapped by changing individual settings in the Sine column, or by selection one of several presets from the Set Channel Assignments menu.



Set Channel Assignments for generator waveforms

For stereo or multichannel generator waveform files, you can set channel mapping. By default, channel 1 audio is mapped to channel 1 output, channel 2 to channel 2, and so on. If the number of channels in the waveform file is less than the number of output channels, the waveform channels resume numbering at 1 and wrap to the next available output channel. These assignments can be remapped by changing individual settings in the **Wave Ch** column, or by selection one of several presets from the **Set Channel Assignments** menu.



File Playback

For generator waveform file playback, you can view the length of the file, and you **can** adjust the playback start position. Select **Seconds** or Samples for the **Length** and **Start Position** units. Enter a new **Start Position** if desired.

Signal Acquisition and Analysis

Input Range tab

Digital inputs do not require ranging, and the Input Range tab is unavailable when inputs are set to a digital format.

If **Auto Range** is **checked** (the default), each analog channel determines its input range automatically, based on the level of the input signal. If the input signal level changes beyond a ranging threshold, **Auto Range** will cause the input ranging circuits to move up or down for proper ranging.

See page 551 for more information about ranging and autoranging.

If **Auto Range** is **unchecked**, you can set a fixed range for each analog input channel. If **Track first channel range** is **checked** (the default), the fixed input range setting for channel 1 is copied to the other channels, and range settings for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels.

To set individual analog input channel ranges, **uncheck** the **Track first channel range** checkbox and enter values in the input channel Range fields.

Settling tab

One or two Settling tabs are available for Single Value measurement results. These tabs may have different parameters and default values depending upon the measurement result to which they're attached. See page 552 for more information about settling.

53

Level Ratio
(Sequence Mode)

The Level Ratio measurement provides a single-value meter result of the ratio of the level in each channel to the level in the selected Reference channel

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 6 for information about Signal Path Setup.

Generator

The typical configuration for this measurement is the closed-loop configuration, using the APx generator as a stimulus. Alternatively, you can make open-loop measurements, using either the External Source or File Input configuration.

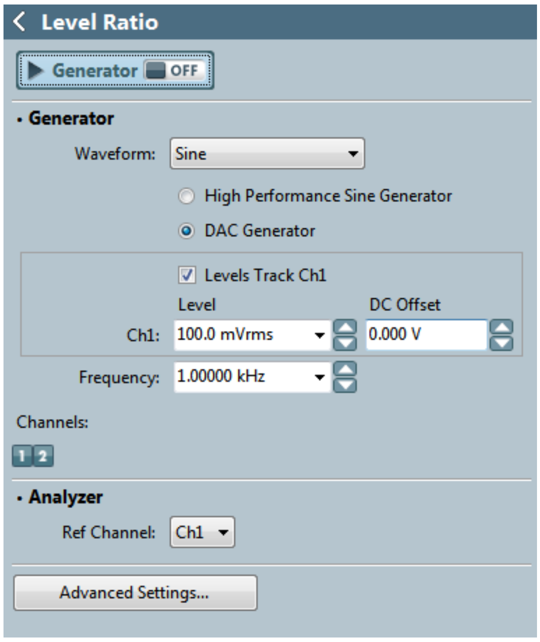
See Chapter 5 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 49 for a comparison of features.

Analyzer

No local low pass, high pass or weighting filters are available for this measurement. However, low pass, high pass and weighting filter settings made in Signal Path Setup



> Input/Output will affect this measurement.

Ref Channel

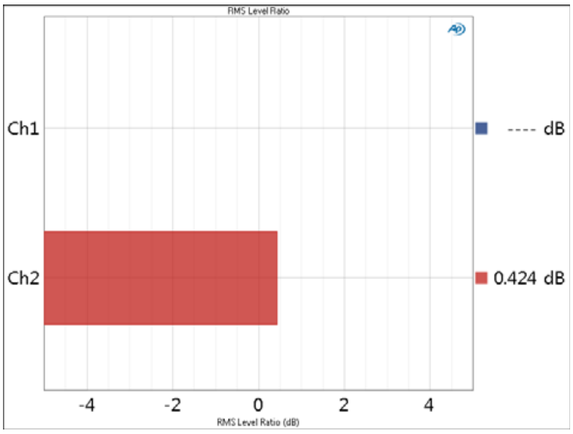
The level ratio of each channel is measured against the selected Reference Channel. There is no level ratio for the Reference Channel itself, and no data is displayed for that channel.

Advanced Settings

Advanced Settings makes available individual channel output level and offset controls, and input ranging and settling controls. See Advanced Settings for single value measurements on page 317.

See Chapter 98 for more information about units of measurement.

Level Ratio



Level Ratio provides a single-value meter result, displaying the ratio of the level in the non-reference input channel(s) to the level in the Reference Channel.

Units

Units available for Level Ratio are

- x/y
- %
- ppm
- dB

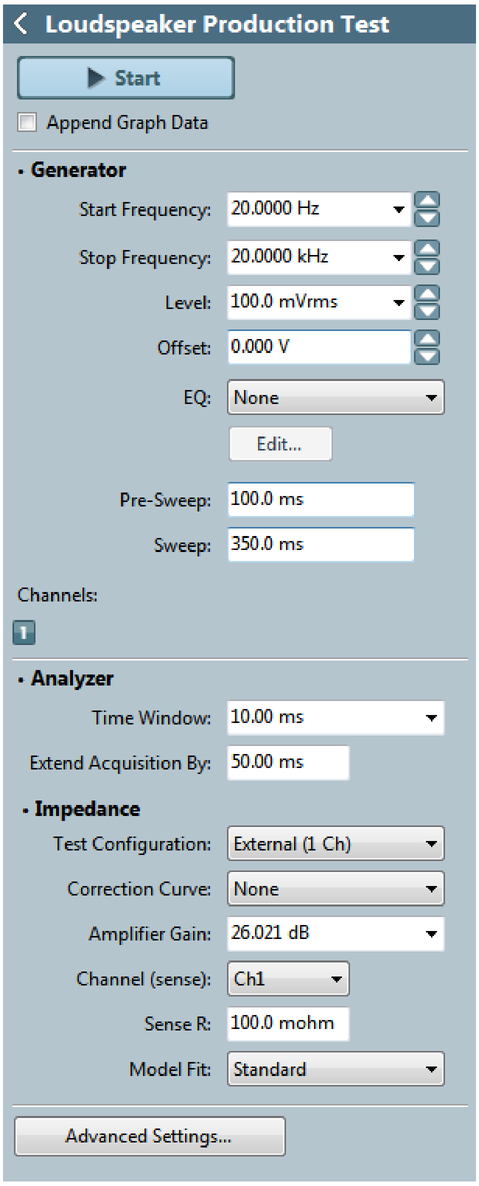
54

Loudspeaker Production Test
(Sequence Mode)

Introduction

This measurement requires a software option key. See page 166 for more information about software options.

Loudspeaker Production Test is only available when the analyzer input and output are set to an analog, non-acoustic configuration.



Loudspeaker Production Test combines two types of measurements in one test to facilitate fast loudspeaker testing in a production environment. An external power amplifier, a measurement microphone with power supply, and an impedance fixture with a sense resistor are required. An acoustic isolation chamber is optional.

The two types of measurements are

- an acoustic measurement, providing frequency response, phase, distortion, rub and buzz and impulse response results;
- an electromechanical impedance measurement, providing impedance response curves and a subset of Thiele-Small parameter results.

The underlying technology is the continuous sweep, providing both acoustic and impedance results in one fast measurement.

Loudspeaker Production Test results available in APx500 are:

- Level (Freq. Response)
- Relative Level
- Phase
- Distortion Product Ratio
- Distortion Product Level
- Rub and Buzz
- Impulse Response (for polarity
- Impedance Magnitude
- Impedance Phase
- Thiele-Small parameters

All of these results are available from a single acquisition.

The acoustic measurement

Set a measurement microphone in front of the loudspeaker under test. Connect this microphone to its power supply and to an undesignated APx analyzer input. An undesignated input is one that does not have a role assigned to it for the impedance measurement. See Connections and test configurations on the next page.

Determining the value for the Time Window

The acoustic measurement is based on the APx Acoustic Response measurement, which makes quasi-anechoic measurements by limiting the duration of the impulse response to the time before the first reflection. This requires setting the measurement Time Window to the proper value. In Acoustic Response, this is an interactive process where the user manually sets a cursor on a graph display, or measures the distance for the direct and first reflection paths in the acoustic space. You can read about determining an optimal Time Window value in Chapter 21.

For production testing, the supervising production engineer must determine the optimal value for the Time Window (using Acoustic Response or some other method) at the time the production station is being set up, and recording the value for entry into the Time Window field in Loudspeaker Production Test. This value can be used until there is a significant change in the acoustic testing space.

One or two of the analyzer input channels must be dedicated to the impedance measurement, measuring voltages and calculating currents. One (or more) of the remaining channels measures the audio acquired by a measurement microphone for an acoustic result.

The impedance measurement

The electromechanical impedance measurement is based on the APx Impedance/Thiele-Small measurement, where the current through the driver and the voltage across the driver must be determined. For fast and reliable production testing, Loudspeaker Production Test only makes only a single-pass, free-air impedance measurement, providing a subset of the Thiele-Small parameters, sufficient to identify manufacturing defects.

An audio power amplifier and a sense resistor are required, and one or two additional analyzer inputs, depending upon the chosen test configuration. The optional Audio Precision IMP1 impedance fixture provides sense resistors, amplifier and analyzer connectors and switching.

You will need to precisely measure the value of the sense resistor, and if you are using the External (1 Ch) test configuration, you will need to create an amplifier correction curve. If you do not create an amplifier correction curve, you must precisely measure the broadband gain of the amplifier before making the measurement, and enter the value into the Amplifier Gain field.

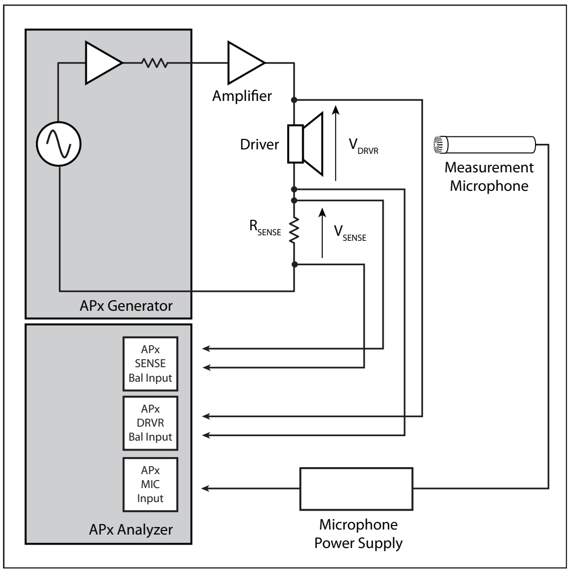
Connections and Test Configurations

Loudspeaker Production Test measurements can use a choice of test configurations to derive the driver current from the available voltages. Select the test configuration suited to your situation.

In these tests, we are not measuring conventional audio parameters; we are measuring voltage or current in circuit components. Consequently, one or two of the analyzer input channels have specific roles assigned, either to measure the voltage across the driver, the voltage across the sense resistor, or the total voltage across the driver and the sense resistor. Please pay close attention to the roles depicted in the illustrations linked here, and identified on the Channel control in the software.

The Audio Precision IMP1 Impedance Fixture can be used with either configuration.

External (2 Ch) (bal)



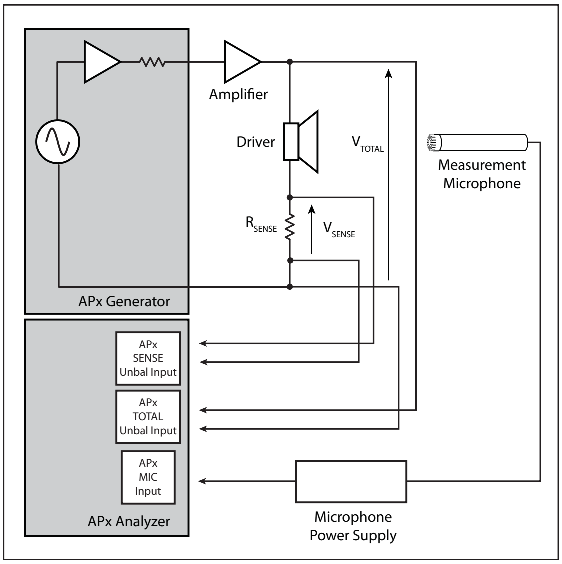
This is our recommended configuration when making a 2-channel impedance measurement. This configuration is enabled when Test Configuration is set to External (2 Ch) and the analyzer inputs are set to balanced.

The Audio Precision IMP1 Impedance Fixture can be used with this configuration.

We recommend that you use balanced analyzer inputs for impedance measurements.

Avoid the use of a bridged amplifier as the accuracy of results can be significantly degraded due to strong common mode signals at the analyzer input.

External (2 Ch) (unbal)



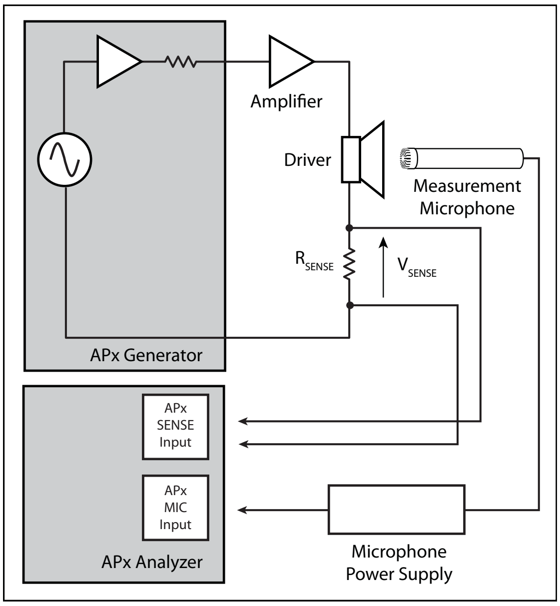
This configuration is enabled when **Test Configuration** is set to **External (2 Ch)** and the analyzer inputs are set to unbalanced.

This configuration is similar to **External (2 Ch)** (bal), but it allows the use of unbalanced analyzer inputs, which may be more convenient in some situations. V_{DRVR} is calculated by subtracting V_{SENS}E from V_{TOTAL}. When the analyzer is set to unbalanced inputs, this configuration is assumed and voltage measured at channel *n*+1 is subtracted from the voltage at channel *n*, provided the driver voltage.

The Audio Precision IMP1 Impedance Fixture can be used with this configuration.

A bridged amplifier cannot be used with unbalanced analyzer inputs.

External (1 Ch)



This method uses an external power amplifier and sense resistor. This is the only configuration that supports Loudspeaker Production Test using an analyzer with only two inputs.

Set **Correction Curve** to select an amplifier correction curve in the project, or choose to **Create New** or **Browse for File.** If the correction curve accurately represents the amplifier in use, this configuration will make accurate impedance response measurements. Enter the value of the sense resistor into the data entry field in APx. This is our recommended configuration when making a 1 channel impedance measurement.

If **Correction Curve** is set to **None**, the broadband gain of the amplifier must be known from a previous measurement, such as the APx Level and Gain measurement. Enter the gain and the value of the sense resistor into the data entry fields in APx.

When **Correction Curve** is set to **None**, this configuration cannot correct for the low-frequency rolloff present in an AC-coupled power amplifier. AC coupling will introduce phase shift that clearly affects the driver impedance curve and resonance frequency. If your test requires a sweep that extends below 20 Hz, we recommend the use of a DC-coupled power amplifier, or that you use a correction curve as described above.

The Audio Precision IMP1 Impedance Fixture can be used with this configuration.

We recommend that you use balanced analyzer inputs for impedance measurements.

Avoid the use of a bridged amplifier as the accuracy of results can be significantly *degraded due to strong common mode signals at the analyzer input.*

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 6 for information about Signal Path Setup.

After making the settings below, click Start.

Generator

Append Graph Data

Measured data are grouped in a Data Set, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the Append Graph Data box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 573.

Generator

The Loudspeaker Test measurement uses a continuous sweep (log-swept sine chirp) stimulus signal, swept between the frequencies entered in Start Frequency and Stop Frequency, at the value set in Level.

Running the Measurement

To use Loudspeaker Production Test, click Start. The generator will output the test signal to the DUT on the selected generator channels. Read the results in the selected result view.

Start Frequency and Stop Frequency

These controls set the frequency range of the sweep. For testing a loudspeaker, set the range to the range of your loudspeaker or driver. See page 220 for more about continuous sweep ranges, which are dependent upon the instrument.

Level

Set the generator level here.

Note: Loudspeakers under test can be damaged by signals that exceed their level or frequency limits. Be sure to set generator and amplifier level and sweep frequencies to values within the limits of the loudspeaker.

EQ

You can equalize the generator signal to make a pre-emphasized stimulus. See Generator Equalization (page 170) for a general discussion of this feature.

Choosing generator equalization

Select None, Relative or Absolute from the EQ drop-down menu.

- None applies no equalization to the generator.
- Relative applies the gain or loss specified in the EQ table to the current generator level for each channel.
- Absolute sets the generator level for all channels to the level specified in the EQ table. Note: when Absolute is selected, current channel level settings are lost.

Edit

If you have chosen to equalize the generator signal, you can edit the EQ table here. See Edit EQ Table (page 170).

Pre-Sweep and Sweep times

A continuous sweep stimulus consists of a sine wave swept logarithmically across the sweep range in a short period of time.

To enable the DUT to stabilize before the beginning of the defined sweep measurement, there is an adjustable Pre-Sweep time. The actual sweep stimulus signal begins below the defined Start Frequency and is swept upward, reaching the Start Frequency at the end of the Pre-Sweep time. The sum of the times in Pre-Sweep and Sweep is the total duration of the sweep.

- The default for Pre-Sweep is 100.0 ms. Minimum is 0 s; maximum is 1.0 s.
- The default for Sweep is 350.0 ms. Minimum is 50.0 ms; maximum is 5 s.

Longer Pre-Sweep settings allow more time for DUT stabilization, but increase measurement time. Longer Sweep settings provide greater resolution and signal-to-noise ratio, but increase measurement time.

Analyzer

Time Window

Enter a value for the Time Window here. See page 322 for more information about Time Window values.

Extend Acquisition By

By default, a continuous sweep measurement makes an acquisition 50 ms longer than the stimulus, to include possible time-delayed artifacts created in the DUT.

In some cases, you may want to extend the acquisition further. Enter a new value in the Extend Acquisition By: field. Minimum extension is 0 s; maximum is 3 s.

Impedance

Test Configuration

This measurement supports two test configurations. The selection made here affects the availability or options of the Amplifier Gain, Channel and Sense R settings below. The test configurations are:

- External (2 Ch) or
- External (1 Ch)

See page 322 for a full description of these configurations.

External (2 Ch)

For the Loudspeaker Production Test measurement, this configuration is not supported by 2-channel analyzers.

This configuration uses an external power amplifier and an external sense resistor. It requires 2 analyzer input channels for the impedance measurement, but because both the voltage across the driver and the current through the driver are measured simultaneously, it produces the best results.

Since Loudspeaker Production Test also requires at least one channel for the acoustic measurement microphone, the Loudspeaker Production Test measurement only supports the External (2 Ch) configuration for analyzers with more than two input channels.

Channel

Balanced inputs

See a diagram of this configuration on page 322.

We recommend the use of balanced inputs for impedance testing. When using balanced inputs, the External (2 Ch) Test Configuration requires one input channel to be assigned the role of “drvr”. This channel measures the voltage across the loudspeaker driver.

It requires a second input channel to be assigned the role of “sense”. This channel measures the voltage across the sense resistor.

Channel roles are assigned in adjacent pairs, with “drvr” always the lower numbered channel.

Unbalanced inputs

See a diagram of this configuration on page 323.

When using unbalanced inputs, the External (2 Ch) Test Configuration requires one input channel to be assigned the role of “total”. This channel measures the total voltage across the loudspeaker driver and the sense resistor. This is the same as the amplifier output voltage.

It requires a second input channel to be assigned the role of “sense”. This channel measures the voltage across the sense resistor.

Channel roles are assigned in adjacent pairs, with “total” always the lower-numbered channel. The voltage across the driver is calculated by subtracting the sense voltage from the total voltage.

External (1 Ch)

See a diagram of this configuration on page 323.

This configuration only requires 1 channel for the impedance measurement, allowing support for 2-channel amplifiers when using Loudspeaker Production Test. A second channel is used for the acoustic measurement. Like the above configurations, it requires an external power amplifier and an external sense resistor.

For the External (1 Ch) configuration, you must use an amplifier correction curve (see below) or provide a broadband amplifier gain value. We recommend using an amplifier correction curve, as it is more accurate. Characteristics of the amplifier used and the accuracy of the correction curve will affect your results.

Correction Curve:

- None
  If you are not using a correction curve, choose None and enter a value for the broadband amplifier gain in the field below. The response of your amplifier will affect the accuracy of the impedance curves. See External (1 Ch) Test Configuration on page 323 for more information.
- If there are Amplifier Correction Curves in the project, they will be listed here and available for selection.
- Create New
  To create an amplifier correction curve, choose Create New. See Amplifier Correction on page 292.
- Browse for file
  To use a previously saved amplifier correction curve, choose Browse for file.

Amplifier Gain

This field is only available when Test Configuration is set to the External (1 Ch) configuration and Correction Curve is set to None. Determine the broadband gain of your amplifier using a measurement such as the APx Level and Gain, and enter the value here.

Channel

The **External (1 Ch)** Test Configuration requires one input channel to be assigned the role of "**sense**". This channel measures the voltage across the sense resistor.

Sense R:

Enter the value of the sense resistor here.

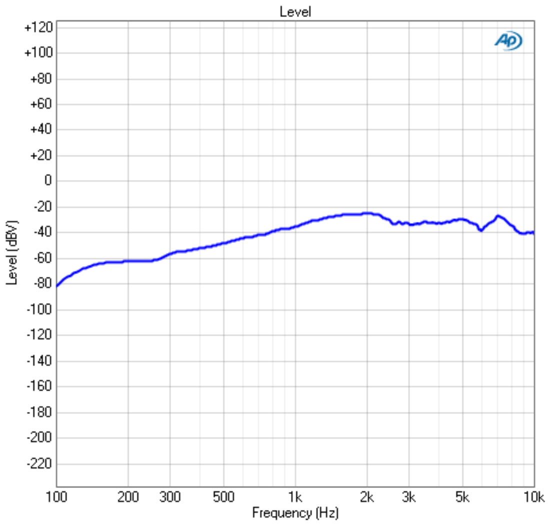
Model Fit

After the preliminary set of Thiele-Small parameters are obtained from the free air pass, a model of the impedance curve is created. This model is plotted with the result data to allow you to see how well the model curve and the data fit. The model curve is displayed as a trace called "Fit."Choose

- Standard
- LR-2 or
- Wright

See page 291 for detailed information and diagrams of the driver models.

Level (Frequency Response)



Loudspeaker Production Test: Level is a continuous sweep result that displays level plotted against frequency. This is the frequency response of the loudspeaker.

A note about the gray area on the graph

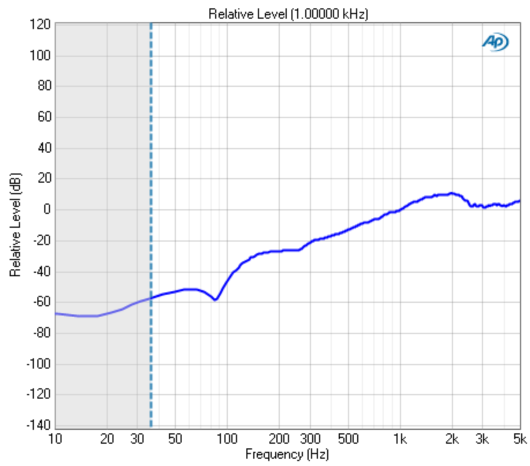
An inevitable consequence of the finite length of an impulse response is that in the frequency domain, the data below some low frequency will be unreliable. In this graph, the area of unreliable data is indicated by gray shading. As the **Time Window** is increased, the frequency of the bound defining this area is decreased.

Units

Units available for Loudspeaker Production Test Level (Frequency Response) are:

X-axis	***Y-axis***
• Hz	• Vrms
• dHz	• dBV
• F/R	• dBu
• %Hz	• dBrA
	• dBrB
	• dBSPL1
	• dBSPL2
	• dBm
	• W (watts)

Relative Level



The Relative Level result is a continuous sweep measurement that provides a graphical display of the frequency response of the loudspeaker. In this result the DUT output level is plotted against frequency, relative to the level at a selected frequency set in the **Ref Frequency** field.

This enables you to specify the frequency that will be set as 0 dB and view the response in relation to that frequency. A midband frequency such as 1 kHz is often set as the reference frequency, and is the default here.

Additional Controls for Relative Level

These settings are made in the Result Settings bar, beneath the graph display.

Mode

Normalized at Reference

Enter a reference frequency for normalization in the Ref Frequency field. The measured level at the frequency you choose is set as 0 dB on the graph, and the DUT response is plotted in relation to this reference. You can change the Ref Frequency at any time (except after appending) and the graph will immediately be redrawn to reflect the new setting.

Auto-Centered in Limits

This control is only usable if upper and lower limits are set on this graph. In some applications, the general conformance of a trace to the shape of a set of limits is of importance, with absolute level less important or irrelevant. Auto-Centered in Limits centers the data trace relative to the upper and lower limits. To exclude possible exceptional data points at frequency extremes, the frequency range that is considered for the centering operation is constrained by the values in the Min and Max fields.

Ref Frequency

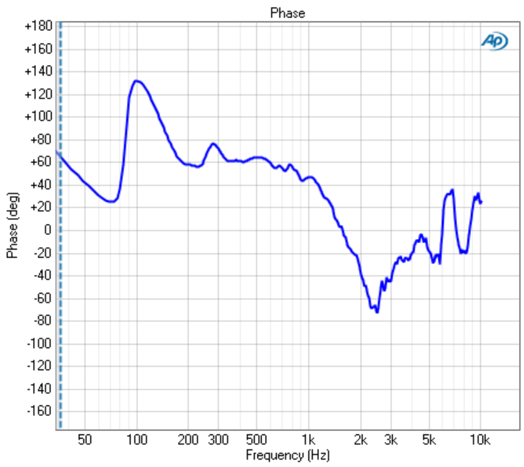
Choose a Reference Frequency here.

Units

Units available for Loudspeaker Production Test: Relative Level are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Phase



The Loudspeaker Production Test Phase result is a continuous sweep measurement that displays the DUT output phase, plotted against frequency.

See page 300 for more information about phase measurements.

Additional Controls for Phase

These settings are made in the Result Settings bar, beneath the graph display.

Mode

Loudspeaker Production Test Phase has four result modes:

Relative to Ch1

In this mode, the absolute phase of channel one is subtracted from the absolute phase of channels numbered greater than 1. The result is plotted against frequency for each channel numbered 2 and above, showing the phase difference (from channel 1) for each channel. Since channel 1 is used as the reference, it is not plotted in this result. This mode shows “unwrapped” phase differences.

Input-to-output

In the input-to-output mode, the absolute phase of each channel, from device input to device output, is plotted against frequency, “unwrapped.” Input-to-output mode includes device delay.

Input-to-output, wrapped

This mode shows the same result as input-to-output phase, but “wrapped” within the range of –180° to +180°.

Input-to-output, excess

This mode shows the input-to-output phase (unwrapped), but removes the linear component (the device delay).

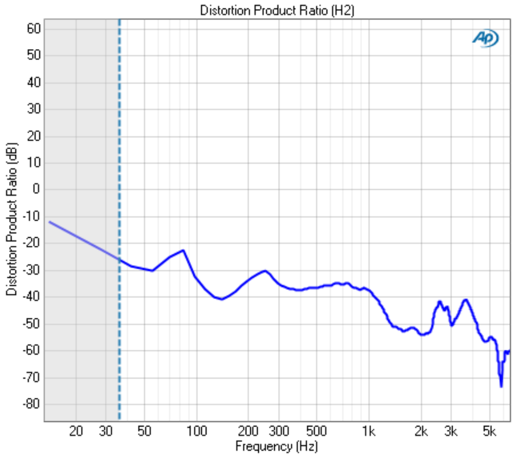
See page 215 for more information about wrapped and unwrapped phase.

Units

Units available for Loudspeaker Production Test Phase are:

X-axis	*Y-axis*
• Hz	• deg
• dHz	• rad
• F/R	
• %Hz	

Distortion Product Ratio



The Loudspeaker Production Test Distortion Product Ratio result is a continuous sweep measurement that provides a graphical display of the selected harmonic distortion products present in each audio input channel. In this result the ratio of the level of the selected harmonic distortion product to the fundamental in the DUT output is plotted against frequency.

Additional Controls for Distortion Product Ratio

Harmonics

For a graph of the ratio of one specific harmonic product to the fundamental level, select **Single Harmonic** and set the **Harmonic Number** to your choice, from 2nd harmonic to 20th harmonic.

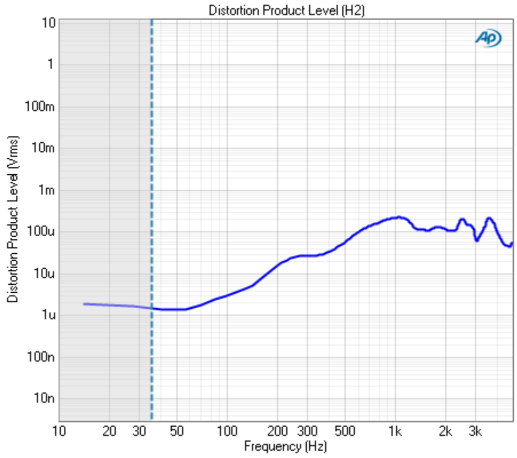
For a graph of the sum of the levels of any combination of harmonic products (from **H2** through **H20**), divided by the fundamental level, select **Sum of Harmonics.** Then click **Select Harmonics...** to open a result definition dialog.

Units

Units available for Loudspeaker Production Test Distortion Product Ratio are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Distortion Product Level



The Loudspeaker Production Test Distortion Product Level result is a continuous sweep measurement that provides a graphical display of the selected harmonic distortion products. In this result the level of the selected harmonic distortion product in the DUT output is plotted against frequency.

Additional Controls for Distortion Product Level

Harmonics

For a graph of the level of one specific harmonic product, select **Single Harmonic** and set the **Harmonic Number** to your choice, from 2nd harmonic to 20th harmonic.

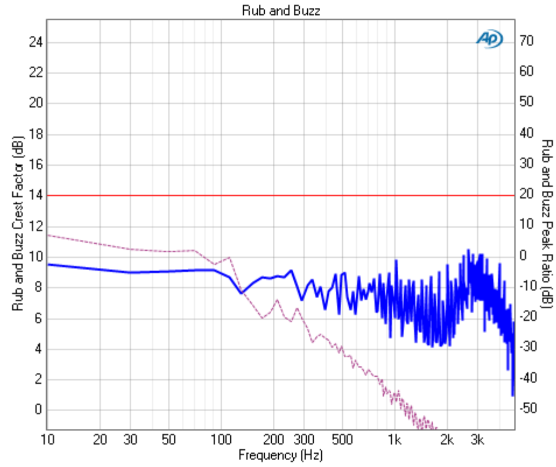
For a graph of the level of the sum of several or all harmonic products (from **H2** through **H20**), select **Sum of Harmonics**, then click **Select Harmonics...** to open a result definition dialog.

Units

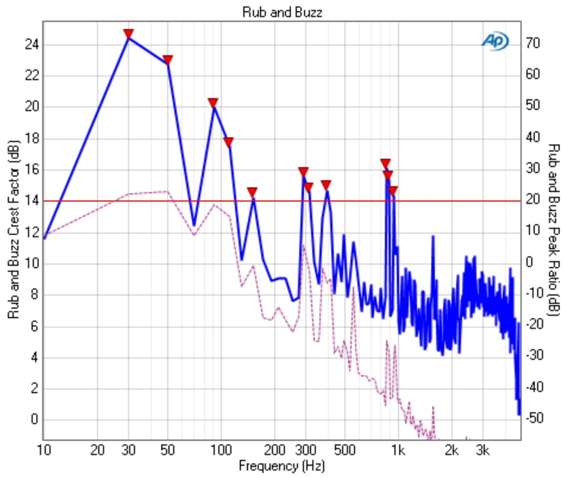
Units available for Loudspeaker Production Test Distortion Product Level are:

X-axis	*Y-axis (analog)*
• Hz	• Vrms
• dHz	• dBV
• F/R	• dBu
• %Hz	• dBrA
	• dBrB
	• dBSPL1
	• dBSPL2
	• dBm
	• W (watts)

Rub and Buzz



Rub and Buzz good result



Rub and Buzz poor result

The Loudspeaker Production Test: Rub and Buzz result uses a continuous sweep measurement to display the Rub and Buzz Crest Factor and Peak Ratio. See More about Loudspeaker Testing on page 185 for a more detailed discussion of using rub and buzz in loudspeaker testing.

Rub and Buzz presents two results on one graph. In the APx implementation, detection of rub and buzz requires a simultaneous interpretation of both of these results; a rub and buzz crest factor result is plotted in reference to the left Y-axis, and a rub and buzz peak ratio result is plotted in reference to the right Y-axis. Limit behavior is different on a graph with 2 Y-axes. See Drawing limits for XY Graphs (two Y-axes) on page 579 and Editing limits for XY Graphs (two Y-axes) on page 581.

You can split the dual axis graph into two individual graphs. Right-click on the Rub and Buzz result and choose Split from the context menu. Two new results will be added, presenting the data for the Left and Right axes individually.

Additional Control for Rub and Buzz

High-pass Factor

The data plotted on these graphs is high-pass filtered to reveal the rub and buzz signature results. The filter tracks the chirp stimulus frequency, moving upward with the sweep. The corner frequency of the filter is the product of the instantaneous chirp frequency multiplied by the **High-pass Factor**, which can be set in the range from 10 to 30. This results in a graph where the X-axis represents the stimulus frequency, and the plotted data represent the much higher frequency sounds generated by the interaction of the stimulus frequency with loose particles or rubbing components within the loudspeaker system. For interpretation of the results, see More About Loudspeaker Testing on page 185.

Units

Units available for Rub and Buzz are:

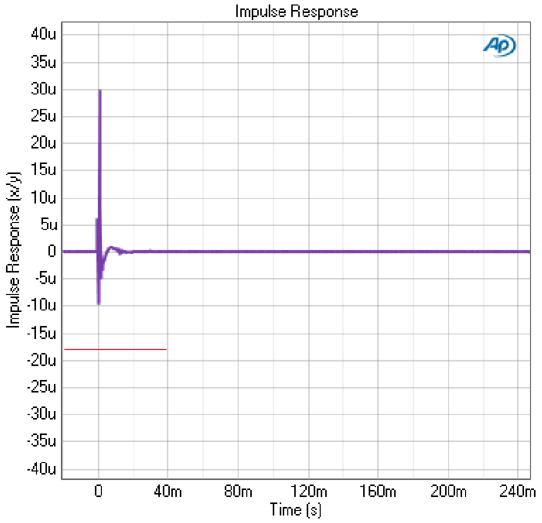
X-axis	*Y-axes*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Impulse Response

The Loudspeaker Production Test: Impulse Response view is a continuous sweep result that displays the Impulse Response of the system under test. This result is particularly useful in production test to verify correct system polarity.

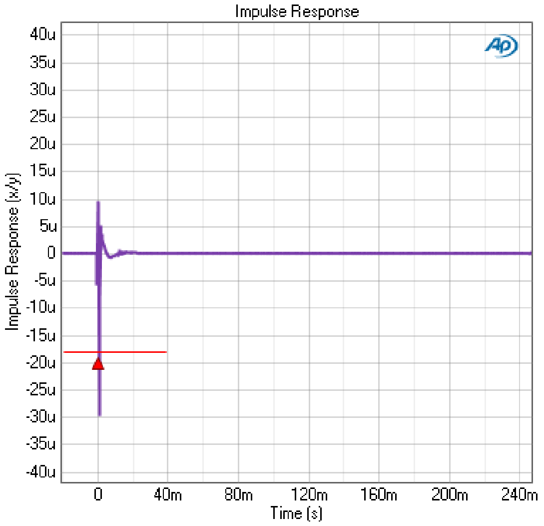
See page 218 for more information about impulse response.

Impulse Response correct polarity



This is a Loudspeaker Production Test Impulse Response result from a system that has correct polarity. The primary impulse is positive. Note the limit set at –18u, to flag reversed polarity.

Impulse Response reversed polarity



This is a Loudspeaker Production Test Impulse Response result from a system that has reversed polarity. The primary impulse is negative. Notice that the limit set at –18u has been crossed and a limit failure marker has been set.

Result Settings bar

Interpolation

For time-domain displays, the APx500 graphs are normally plotted using sinc function interpolation. Interpolation adds points to the displayed trace that do not exist in the measured data, to make data trends more easily visualized. The default in APx is to have Interpolation **On**. However, digital domain signals are sometimes best understood when viewing actual samples, with interpolation switched **Off**.

Interpolation and limit failure markers.

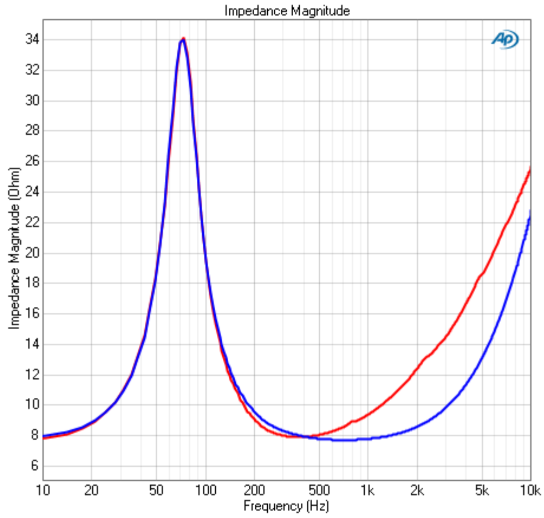
Limit failure markers (small orange triangles) are placed at the first data point that exceeds a limit. When interpolation is on and the graph is zoomed in to high magnification, the limit failure markers may appear to be “floating” in the graph, not at the intersection of the limit line and the interpolated data trace. Although this appears to be an error, it is actually correct behavior. Turn interpolation **Off**, and you will see that the marker is at the first (real) data point that exceeds the limit. There is no data at the intersection, although interpolation suggests there is. See an Illustration of interpolation and limit marker interaction on page 581.

Units

Units available for Impedance/Thiele-Small Impulse Response are:

X-axis	***Y-axis***
• s (seconds)	• x/y

Impedance Magnitude



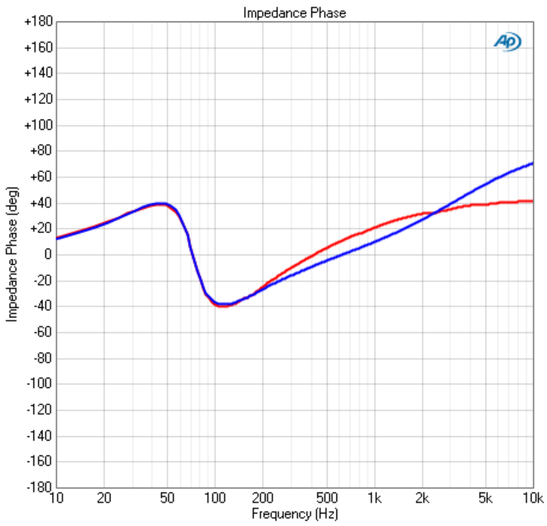
This result provides a graphical display of the impedance response of the driver, plotting the complex impedance magnitude versus frequency.

Units

Units available for Impedance/Thiele-Small Impedance Magnitude are:

X-axis	*Y-axis*
• Hz	• ohms
• dHz	
• F/R	
• %Hz	

Impedance Phase



This result provides a graphical display of the impedance phase response of the driver, plotting the impedance phase versus frequency.

Units

Units available for Impedance/Thiele-Small Impedance Phase are:

X-axis	*Y-axis*
• Hz	• deg
• dHz	• rad
• F/R	
• %Hz	

Thiele-Small Parameters

Parameter	Value	Lower Limit	Upper Limit
F_S	50.50 Hz	----	----
Q_{MS}	2.07	----	----
Q_{ES}	0.56	----	----
Q_{TS}	0.44	----	----
S_D	452.39 cm^2	----	----
R_E	5.51 Ω	----	----
L_E	0.38 mH	----	----
R_2	---- Ω	----	----
L_2	---- mH	----	----
K_{rm}	----	----	----
E_{rm}	----	----	----
K_{xm}	----	----	----
E_{xm}	----	----	----
R_{MS}	14.85 Ns/m	----	----

Thiele-Small parameters

Thiele-Small populates a table with values calculated from the impedance response measurement data. When compared to a known good set of Thiele-Small parameters for a golden unit, this subset can be useful in identifying manufacturing faults in drivers and loudspeakers. R_2 and L_2 are only populated when the LR-2 model is selected, and E_{rm}, K_{rm}, E_{xm} and K_{xm} are only populated when the Wright model is selected.

Setting Limits for Thiele-Small parameters

Parameter	Value	Lower Limit	Upper Limit
F_S	55.37 Hz	53.0	56.0
Q_{MS}	2.27	2.00	2.25
Q_{ES}	0.63	----	----
Q_{TS}	0.50	----	----

Thiele-Small parameters with limits applied (detail)

You can set arbitrary limits for any or all of the Thiele-Small parameters by entering limit values into the results grid. Results that fall outside of the limits will be highlighted in the grid, and in a sequence the measurement will be flagged as **Failed**.

55

Maximum Output
(Sequence Mode)

Introduction

For the APx515, this measurement requires a software option key. See page 166 for more information about software options.

The Maximum Output measurements provide single-value results that show the Maximum Output for each DUT channel. Maximum Output uses a sine wave stimulus signal that is adjusted to the level that produces the target distortion in a selected channel (regulated). This method is often employed to perform measurements at an amplifier's maximum output level.

Go to page 433 for more information about regulation.

Maximum Output measurements are not available in External Source or File Input configurations.

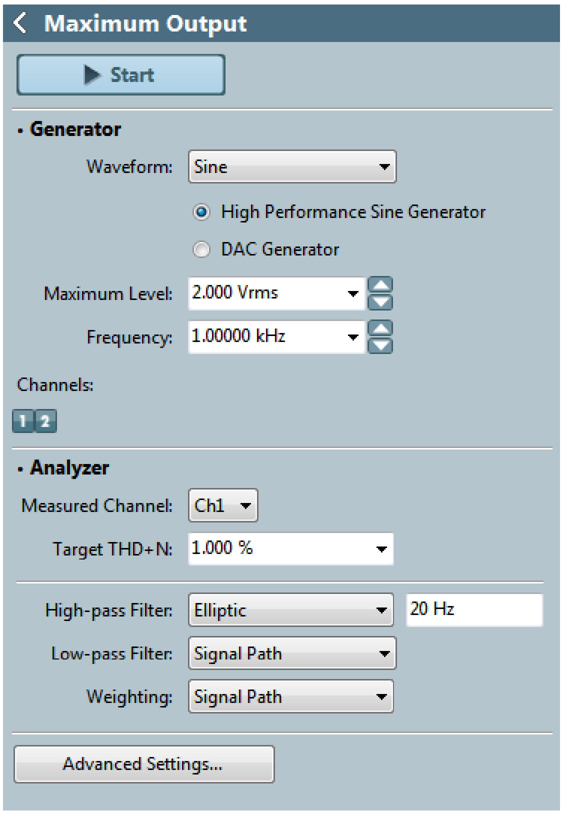
Maximum Output results available in APx500 are:

- RMS Level
- THD+N Level
- THD+N Ratio.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 6 for information about Signal Path Setup.

Click **Start**. The generator will output a sine wave to the DUT on the selected generator channels at the frequency set in the Signal Generation panel. The generator level is automatically adjusted while the analyzer observes the Measured Channel to achieve the level of distortion you specified. If another channel exhibits more distortion than the channel you initially selected, you can set that channel as the Measured Channel and run the measurement again.



Generator

This measurement must be performed in the closed-loop configuration, using the APx generator as a stimulus.

See Chapter 5 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 49 for a comparison of features.

Maximum Level

This setting limits the APx generator output to the selected value to protect your DUT.

Analyzer

Measured Channel

Choose a channel to measure for the **Target THD+N**.

Target THD+N

Choose a target distortion and enter the value in the Target **THD+N field.**

Filters

Local high-pass, low-pass and weighting filters are available for this measurement. The low-pass and high-pass filters, if selected, are applied to the entire measured signal and affect all the results. The weighting filter, if selected, affects only the distortion and distortion plus noise results. See Measurement Filters on page 547 for detailed information about the filters available locally for some measurements.

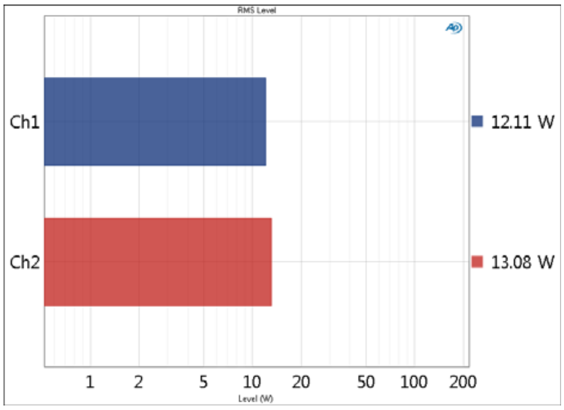
Low pass, high pass and weighting Signal Path Filters settings (see page 545) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass or high-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active. A local weighting filter setting does not override any Signal Path filter settings; the local weighting filter is applied in addition to any active Signal Path filters.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, click **Advanced Settings**. See Advanced Settings for Maximum Output on page 335.

See Chapter 98 for more information about units of measurement.

Maximum Output: Level



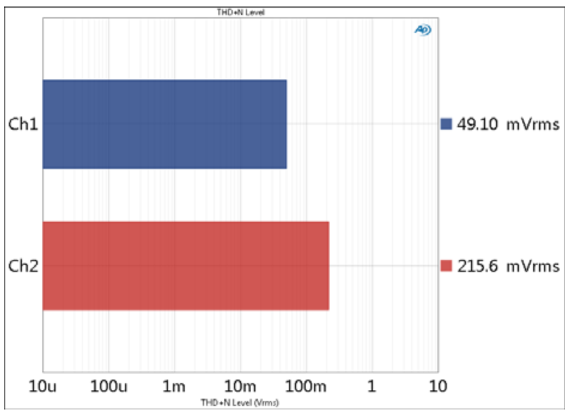
Read the regulated RMS Level for each channel from the meter bar display.

Units

Units available for Maximum Output: RMS Level are

Analog Signals	*Digital Signals*
• Vrms	• dBFS
• dBV	• FS
• dBu	• %FS
• dBrA	• dBrA
• dBrB	• dBrB
• dBSPL1	• dBSPL1
• dBSPL2	• dBSPL2
• dBm	
• W (watts)	

Maximum Output measurements: THD+N Level



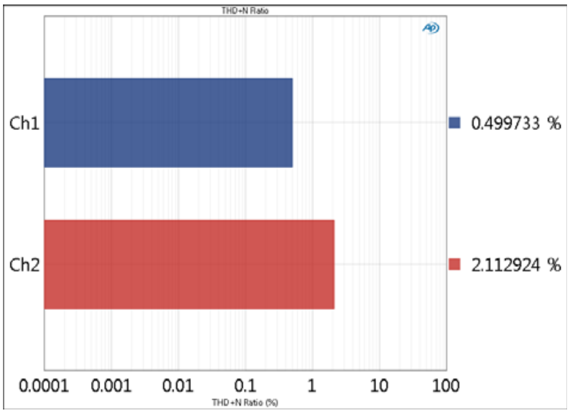
Read the THD+N Level for each channel from the meter bar display.

Units

Units available for Maximum Output: THD+N Level are

Analog Signals	***Digital Signals***
• Vrms	• dBFS
• dBV	• FS
• dBu	• %FS
• dBrA	• dBrA
• dBrB	• dBrB
• dBSPL1	• dBSPL1
• dBSPL2	• dBSPL2
• dBm	
• W (watts)	

Maximum Output measurements: THD+N Ratio



Read the THD+N Ratio for each channel from the meter bar display.

Units

Units available for Maximum Output: THD+N Ratio are

- x/y
- %
- ppm
- dB

Advanced Settings for Maximum Output

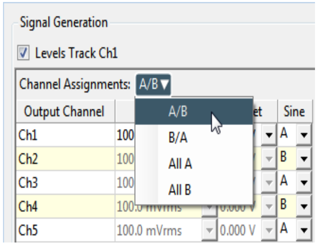
The default settings here are appropriate for most measurements. You may want to make minor adjustments for special situations.

Signal Generation

No advanced settings for signal generation are available for Maximum Output (CEA-2006) or Regulated Frequency Sweep measurements.

Set Channel Assignments (Maximum Output measurement)

For **Split Sine** special waveform generation, you can set channel mapping. By default, channel A audio is mapped to odd numbered output channels, channel B audio to even numbered output channels. These assignments can be remapped by changing individual settings in the **Sine** column, or by selection one of several presets from the **Set Channel Assignments** menu.



Signal Acquisition and Analysis

Input Range tab

Digital inputs do not require ranging, and the Input Range tab is unavailable when inputs are set to a digital format.

If **Auto Range** is checked (the default), each analog channel determines its input range automatically, based on the level of the input signal. If the input signal level changes beyond a ranging threshold, Auto Range will cause the input ranging circuits to move up or down for proper ranging.

See Chapter 92 for more information about ranging.

If **Auto Range** is unchecked, you can set a fixed range for each analog input channel. If **Track first channel range** is checked (the default), the fixed input range setting for channel 1 is copied to the other channels, and range settings for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels.

To set individual analog input channel ranges, **uncheck** the “Track first channel range” checkbox and enter values in the input channel Range fields.

Settling tab

Two Settling tabs are available for regulated measurements, Settling (Level) and Settling (THD+N).

See Chapter 92 for more information about Settling for regulated measurements.

56

Maximum Output (CEA-2006)
(Sequence Mode)

Introduction

For the APx515, this measurement requires a software option key. See page 166 for more information about software options.

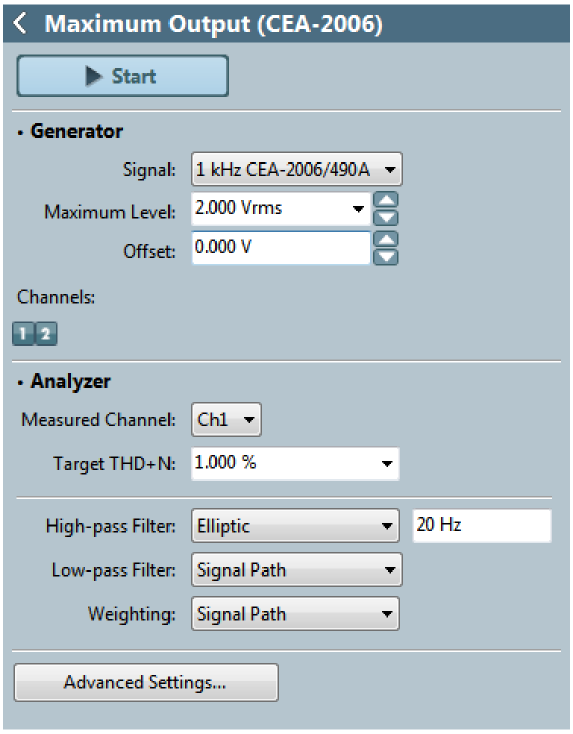
The Maximum Output (CEA-2006) measurements provide single-value results that show the DUT output for each channel during the burst ON interval, after regulation is performed. Maximum Output (CEA-2006) measurements use a tone-burst stimulus signal that is adjusted to the level that produces a target distortion in a selected channel (regulated). This method is specified in the standard CEA-2006 “Testing & Measurement Methods for Mobile Audio Amplifiers,” section 5.2, Dynamic Power testing.

Go to page 433 for more information about regulation, and to page 339 for more information about burst measurements.

Maximum Output (CEA-2006) results are not available in External Source or File Input configurations.

Maximum Output (CEA-2006) results available in APx500 are:

- RMS Level
- THD+N Level
- THD+N Ratio.



Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 6 for information about Signal Path Setup.

Click Start. The generator will output a sine burst to the DUT on the selected generator channels at the frequency set in the Signal Generation panel. The generator is automatically adjusted while the analyzer observes the Measured Channel during the burst ON interval to achieve the level of distortion you specified. If another channel exhibits more distortion than the channel you initially selected, you can set that channel as the Measured Channel and run the measurement again.

Generator

This measurement must be performed in the closed-loop configuration, using the APx generator as a stimulus.

See Chapter 5 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings.

Signal

Two sine burst signals are available:

- 1 kHz CEA-2006/490A
- 50 Hz CEA-2006

Maximum Level

This setting limits the APx generator output to the selected value to protect your DUT.

Analyzer

Measured Channel

Choose a channel to measure for the **Target THD+N**.

Target THD+N

Choose a target distortion and enter the value in the Target **THD+N field.**

Filters

Local high-pass, low-pass and weighting filters are available for this measurement. The low-pass and high-pass filters, if selected, are applied to the entire measured signal and affect all the results. The weighting filter, if selected, affects only the distortion and distortion plus noise results. See Measurement Filters on page 547 for detailed information about the filters available locally for some measurements.

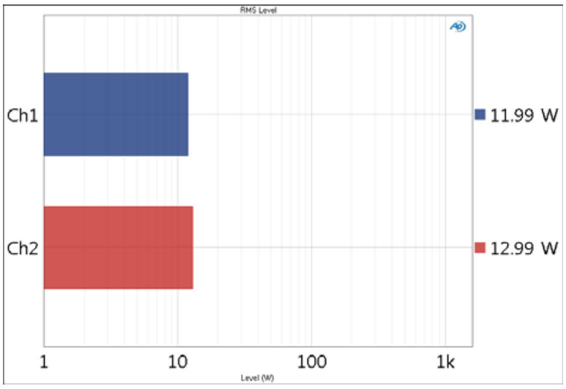
Low pass, high pass and weighting Signal Path Filters settings (see page 545) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass or high-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active. A local weighting filter setting does not override any Signal Path filter settings; the local weighting filter is applied in addition to any active Signal Path filters.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, click **Advanced Settings**. See Advanced Settings for Burst Measurements on page 339.

See Chapter 98 for more information about units of measurement.

Maximum Output (CEA-2006): Level



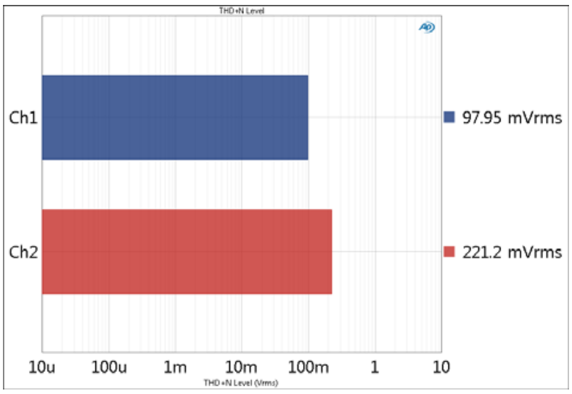
Read the RMS Level for each channel from the meter bar display.

Units

Units available for Maximum Output (CEA-2006): RMS Level are

Analog Signals	***Digital Signals***
• Vrms	• dBFS
• dBV	• FS
• dBu	• %FS
• dBrA	• dBrA
• dBrB	• dBrB
• dBSPL1	• dBSPL1
• dBSPL2	• dBSPL2
• dBm	
• W (watts)	

Maximum Output (CEA-2006): THD+N Level



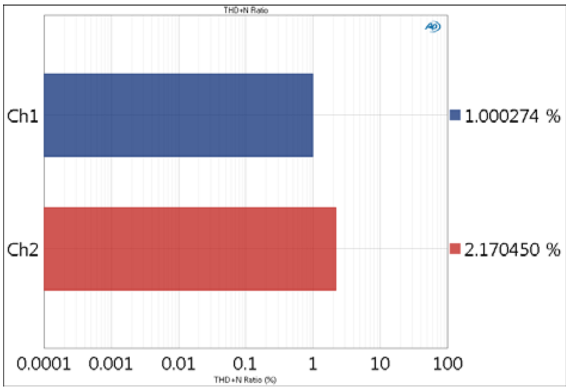
Read the THD+N Level for each channel from the meter bar display.

Units

Units available for Maximum Output (CEA-2006): THD+N Level are

Analog Signals	*Digital Signals*
• Vrms	• dBFS
• dBV	• FS
• dBu	• %FS
• dBrA	• dBrA
• dBrB	• dBrB
• dBSPL1	• dBSPL1
• dBSPL2	• dBSPL2
• dBm	
• W (watts)	

Maximum Output (CEA-2006): THD+N Ratio



Read the THD+N Ratio for each channel from the meter bar display.

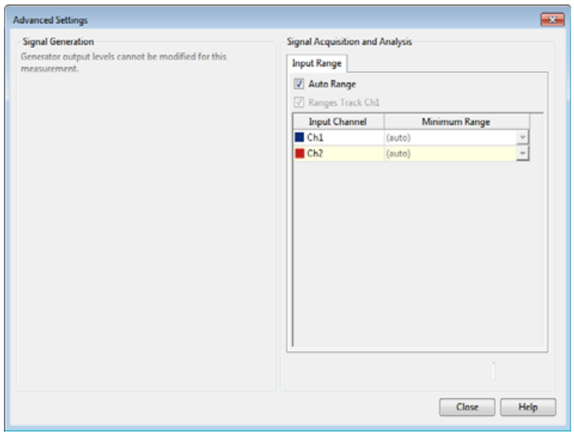
Units

Units available for Maximum Output (CEA-2006): THD+N Ratio are

- x/y
- %
- ppm
- dB

Advanced Settings for Burst Measurements

The default settings here are appropriate for most measurements. You may want to make minor adjustments for special situations.



Signal Generation

There are no advanced settings for signal generation for regulated burst measurements.

Signal Acquisition and Analysis

Input Range tab

Digital inputs do not require ranging, and the Input Range tab is unavailable when inputs are set to a digital format.

Maximum Output (CEA-2006) uses a modified auto-ranging process to find the correct input range.

The burst measurement acquisition begins in the minimum range and moves up from there if necessary. You can specify a higher range by entering the value in the **Minimum Range** field.

The regulation algorithm moves the range upward if necessary while searching for the target distortion.

If your Maximum Output (CEA-2006) measurement is a step in an automated sequence that you would like to run as fast as possible, you can optimize the speed of the ranging process by setting the minimum range to the correct range for the measurement.

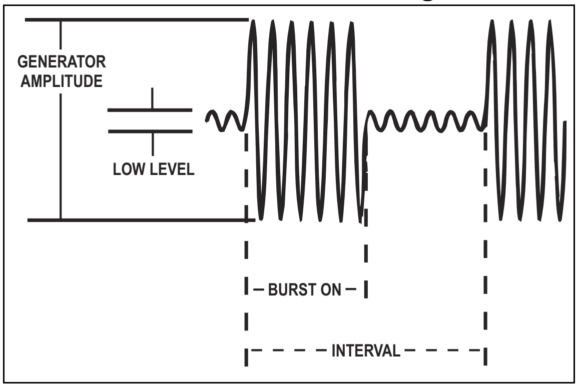
See Chapter 92 for more information about ranging.

No settling is available for Maximum Output (CEA-2006) results.

More about burst measurements

“Tone bursts” have been used for many years as stimulus signals for audio testing.

A burst signal is usually a sine wave that is switched between two levels (that is, modulated by a square wave), with the higher level typically 100% of the set generator amplitude and the lower level typically 10% (–20 dB). The “duty cycle” of the burst is usually less than 50% (the “on” time is less than the “off” time). Well-formed tone burst signals will have an integer number of sine cycles in the burst, and the beginning and end of the burst will be zero-crossing.



Tone Burst Diagram

Other common burst signals include “shaped bursts” (sine waves modulated by a wave shape other than a square wave).

Burst signals for CEA-2006 in APx500

The burst signals in APx500 are designed to satisfy the requirements of the standard CEA-2006 “Testing & Measurement Methods for Mobile Audio Amplifiers,” section 5.2, Dynamic Power testing.

Two signals are specified in the standard: for full-range amplifiers, the signal shall be a repetition of a burst of 20 cycles of a 1 kHz sine wave at 100%, followed by 480 cycles of 1 kHz at 10% (–20 dB). For limited-range amplifiers (subwoofer amplifiers), the signal shall be a repetition of a burst of 10 cycles of a 50 Hz sine wave at 100%, followed by 20 cycles of 50 Hz at 10% (–20 dB).

APx500 provides both these stimulus signals in the Max Output (Burst) measurement result.

Analyzing burst signals in APx500

The first few cycles of the burst are ignored to minimize spurious results caused by initial transients. For the 1 kHz burst, the first 9 cycles are ignored and the last 11 cycles are analyzed. For the 50 Hz burst, the first 2 cycles are ignored and the last 8 cycles are analyzed. Any delay in the DUT is automatically taken into account.

57

Measurement Recorder (Sequence Mode)

The Measurement Recorder is a general-purpose diagnostic tool that provides a record of level, gain, interchannel phase, THD+N ratio or level, DC level, frequency, or SINAD versus elapsed time. The record can be very long (up to one week), providing a means of monitoring the output of a device under test over an extended period of time.

The Measurement Recorder does not require a specific test signal. It can be used with any audio signal within the input range of the analyzer.

Measurement Recorder results available in APx500 are:

- RMS Level
- Gain
- Phase
- THD+N Ratio
- THD+N Level
- THD Ratio
- THD Level
- SINAD
- Bandpass Level
- Frequency
- DC Level
- Peak Level
- Average Jitter Level

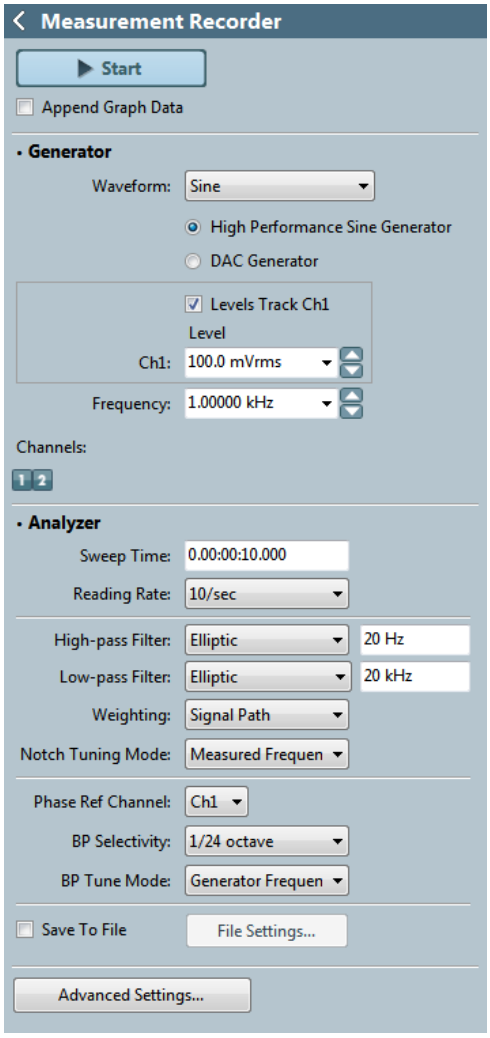
Gain is not available in external source configuration, or if a square wave, a generator waveform or File Input has been selected for the measurement.

Operation

If you have not yet set up your test, first go to Signal Path Setup (Chapter 6) to select and configure inputs and outputs, references and other settings.

Append Graph Data

Measured data are grouped in a Data Set, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox and a Notes field for each data set. See page 573 for more information about managing Data Sets.



Running the measurement

To use Measurement Recorder: Level, click **Start**. The generator will output the test signal to the DUT on the selected generator channels. The measurement recording will begin and the graph will display as the record is acquired. The record will continue until the end time set in Sweep Time.

Note: For digital inputs, sample rate must be stable before the Measurement Recorder begins. When playing a digital external source (such as a DVD), we advise that you start the playback device first, then click the measurement Start button.

Generator

The typical configuration for this measurement is the closed-loop configuration, using the APx generator as a stimulus. See Signal Generation (Chapter 5) for detailed information about the using the APx Generator and setting Waveform, Level and Frequency.

Analyzer

Sweep Time

Sweep Time sets the length of the measurement record. Minimum time is 0.1 s, maximum time is 7 days.

Enter the Sweep Time following this pattern:

d:hh:mm:ss.s

where d = days, hh = hours, mm = minutes, ss = seconds. Days and fractional seconds are optional.

Reading Rate

Reading Rate sets a maximum number of times the input signal will be read per second. In most cases the number of readings will be very close to this value. However, the rate is dependent upon PC performance and channel count, and may be lower. See More About Reading Rate on page 349.

Filtering

Local high-pass, low-pass and weighting filters are available for this measurement. The low-pass and high-pass filters, if selected, are applied to the entire measured signal and affect all the results. The weighting filter, if selected, affects only the distortion and distortion plus noise results. See Measurement Filters on page 547 for detailed information about the filters available locally for some measurements.

Low pass, high pass and weighting Signal Path Filters settings (see page 545) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass or high-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active. A local weighting filter setting does not override any Signal Path filter settings; the local weighting filter is applied in addition to any active Signal Path filters.

Notch Tuning Mode

For the harmonic distortion results, a tunable notch filter is used to remove the fundamental from the measurement. The filter can be tuned to:

- Generator Frequency
  The current APx audio generator frequency. When the generator channels are outputting different frequencies (Split Frequency generation), the notch filter center is set to Frequency A. This mode is not available when using a generator waveform file.
- Jitter Generator Frequency
  The current APx jitter generator frequency, when jitter generation is available and enabled. Read more about jitter beginning on page 60.
- Measured Frequency
  The current measured frequency. When the analyzer channels are receiving different frequencies, the notch filter for each channel is centered on the frequency in that channel.
- Fixed Frequency
  A fixed frequency selected by the user. When Fixed Frequency is selected, a Filter Frequency entry field becomes available beneath the Notch Tuning Mode control.

Phase Reference Channel

This selection is only relevant to the Measurement Recorder: Phase measurement. See page 344.

BP Selectivity

This control only affects the Bandpass Level result.

For the Bandpass Level result, you can choose from a number of bandpass filter widths using the BP Selectivity control. The list is ordered from narrowest at the top, to widest at the bottom. The Window Width and x Octave filters are tuned according to the BP Tune Mode setting.

- Window width
  This is the window width of the underlying FFT, typically only a few hertz wide. This selection has very steep skirts and a flat top.
- 1/24 octave
  This selection has –3 dB points that are 1/24 octave apart.
- 1/12 octave
  This selection has –3 dB points that are 1/12 octave apart.
- 1/9 octave
  This selection has –3 dB points that are 1/9 octave apart.

- 1/6 octave
  This selection has –3 dB points that are 1/6 octave apart.
- 1/3 octave
  This selection has –3 dB points that are 1/3 octave apart.
- 1 octave
  This selection has –3 dB points that are 1 octave apart.

All of the “x octave” filters are equivalent to second-order analog bandpass filters, with 6 dB/octave skirts. Q is chosen to achieve the specified bandwidth at the –3 dB points.

BP Min Freq: / BP Max Freq:

When **Rectangular Band** is the **BP Selectivity** choice, these fields become available for lower and upper frequency cutoff point entry.

BP Tune Mode

This control only affects the Bandpass Level result.

The bandpass **Window width** and **x octave** filters are centered at some user-defined frequency for each step of the measurement. This can be:

- Generator Frequency
  The current APx audio generator frequency. When the generator channels are outputting different frequencies (Split Frequency generation), the bandpass filter center is set to Frequency A.
- Jitter Generator Frequency
  The current APx jitter generator frequency, when jitter generation is available and enabled. Read more about jitter beginning on page 60.
- Measured Frequency
  The current measured frequency. When the analyzer channels are receiving different frequencies, the notch filter for each channel is centered on the frequency in that channel.
- Fixed Frequency
  A fixed frequency selected by the user. When **Fixed Frequency** is selected, a **Filter Frequency** entry field becomes available beneath the **BP Tune Mode** control.

Save to File

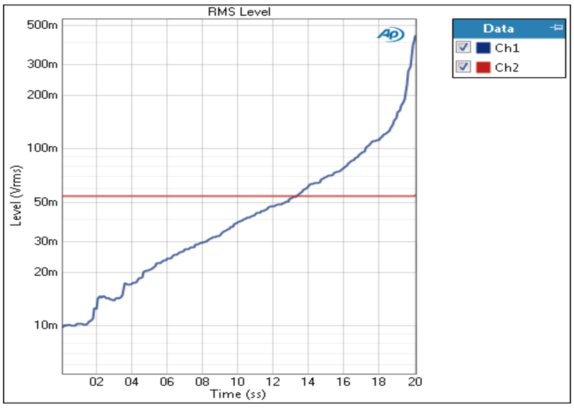
Click this option to save the acquired audio to a file on disk. See **Recording Audio to a File** on page 347.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, see Advanced Settings for the Measurement Recorder on page 350. See Chapter 98 for more information about units of measurement.

RMS Level

The Measurement Recorder: RMS Level result will provide a record of RMS level versus elapsed time for any audio signal. The illustration shows a 20-second record of a 10 kHz sine wave on two channels, with a 10 kHz EQ control adjusted from about –20 dB to about +12 dB during the course of the recording.



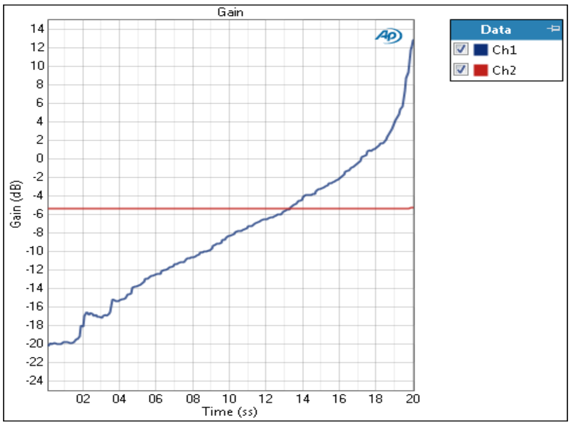
Units

Units available for RMS Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*	*Jitter*
• s	• Vrms	• dBFS	• UI
	• dBV	• FS	• dBUI
	• dBu	• %FS	• s
	• dBrA	• dBrA	
	• dBrB	• dBrB	
	• dBSPL1	• dBSPL1	
	• dBSPL2	• dBSPL2	
	• dBm	• D	
	• W (watts)		
	• V		

Gain

The Measurement Recorder: Gain measurement result will provide a record of gain versus elapsed time for any audio signal. The illustration shows a 20-second record of a 10 kHz sine wave on two channels, with a 10 kHz EQ control adjusted from about –20 dB to about +12 dB during the course of the recording.



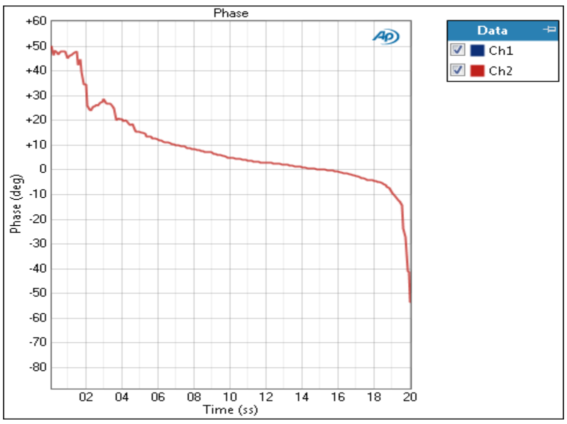
Units

Units available for: Gain are:

X-axis	*Y-axis (same-domain)*	*Y-axis (cross-domain)*
• s	• x/y • % • ppm • dB	• FS/Vrms • dB(FS/Vrms) or • Vrms/FS • dB(Vrms/FS)

Phase

The Measurement Recorder: Phase measurement result will provide a record of relative phase of the DUT channels at a single frequency. One channel is chosen as the phase reference channel, and the remaining channels are measured against it. The illustration shows a 20-second record of a 10 kHz sine wave on two channels, with a 10 kHz EQ control adjusted from about –20 dB to about +12 dB during the course of the recording. For more information about phase, see page 300.

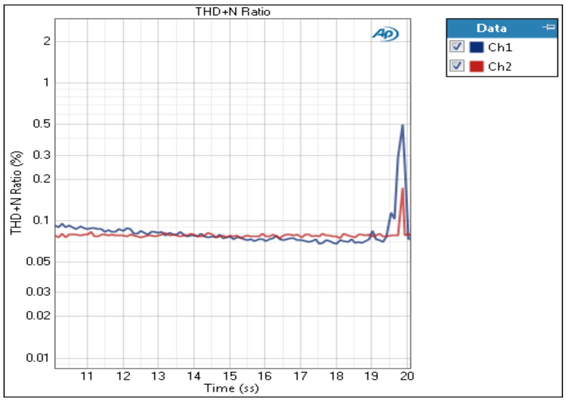


Units

Units available for Phase are:

X-axis	*Y-axis*
• s	• deg • rad

THD+N Ratio



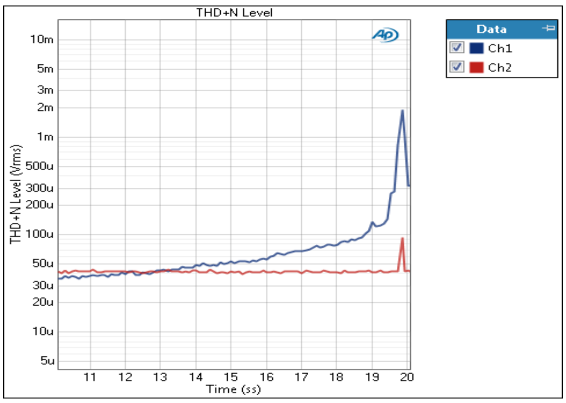
The Measurement Recorder: THD+N Ratio measurement result provides a record of the THD+N ratio versus elapsed time for any audio signal. The illustration shows a 20-second record of a 10 kHz sine wave on two channels, with a 10 kHz EQ control adjusted from about –20 dB to about +12 dB during the course of the recording.

Units

Units available for THD+N Ratio are:

X-axis	*Y-axis*
• s	• x/y • % • ppm • dB

THD+N Level



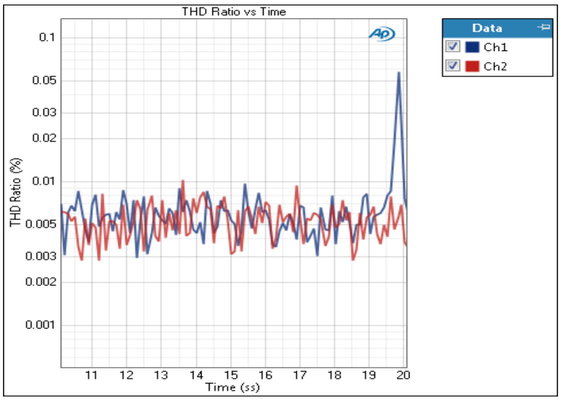
The Measurement Recorder: THD+N Level measurement result provides a record of THD+N level versus elapsed time for any audio signal. The illustration shows a 20-second record of a 10 kHz sine wave on two channels, with a 10 kHz EQ control adjusted from about –20 dB to about +12 dB during the course of the recording.

Units

Units available for THD+N Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*	*Jitter*
• s	• Vrms	• dBFS	• UI
	• dBV	• FS	• dBUI
	• dBu	• %FS	• s
	• dBrA	• dBrA	
	• dBrB	• dBrB	
	• dBSPL1	• dBSPL1	
	• dBSPL2	• dBSPL2	
	• dBm	• D	
	• W (watts)		
	• V		

THD Ratio



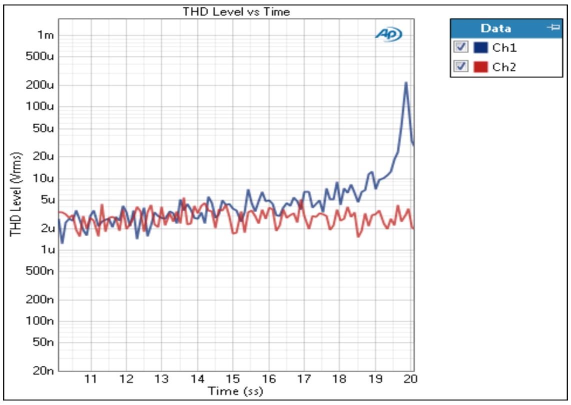
The Measurement Recorder: THD Ratio measurement result provides a record of the THD ratio versus elapsed time for any audio signal. The illustration shows a 20-second record of a 10 kHz sine wave on two channels, with a 10 kHz EQ control adjusted from about –20 dB to about +12 dB during the course of the recording.

Units

Units available for THD Ratio are:

X-axis	*Y-axis*
• s	• x/y
	• %
	• ppm
	• dB

THD Level



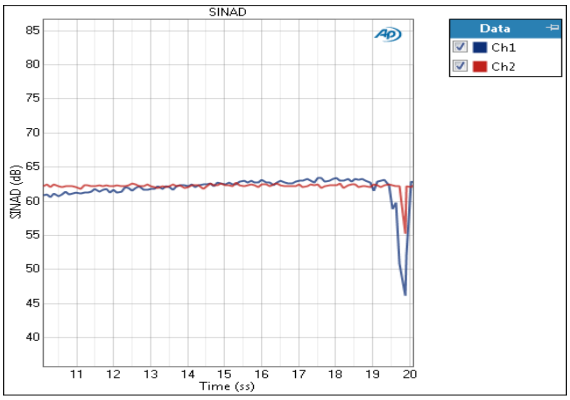
The Measurement Recorder: THD Level measurement result provides a record of THD level versus elapsed time for any audio signal. The illustration shows a 20-second record of a 10 kHz sine wave on two channels, with a 10 kHz EQ control adjusted from about –20 dB to about +12 dB during the course of the recording.

Units

Units available for THD Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*	*Jitter*
• s	• Vrms	• dBFS	• UI
	• dBV	• FS	• dBUI
	• dBu	• %FS	• s
	• dBrA	• dBrA	
	• dBrB	• dBrB	
	• dBSPL1	• dBSPL1	
	• dBSPL2	• dBSPL2	
	• dBm	• D	
	• W (watts)		
	• V		

SINAD



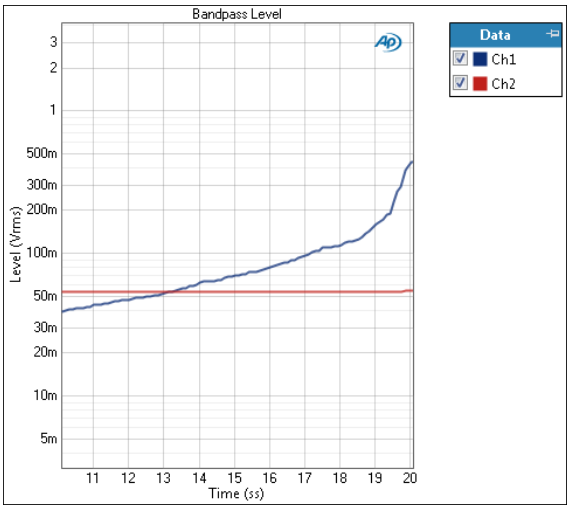
The Measurement Recorder: SINAD measurement result provides a record of the SINAD ratio versus elapsed time for any audio signal. The illustration shows a 20-second record of a 10 kHz sine wave on two channels, with a 10 kHz EQ control adjusted from about –20 dB to about +12 dB during the course of the recording.

Units

Units available for SINAD are:

X-axis	*Y-axis*
• s	• x/y
	• dB

Bandpass Level



The Bandpass Level measurement result provides a record of the Bandpass Level versus elapsed time for any audio signal. The illustration shows a 20-second record of a 10 kHz sine wave on two channels, with a 10 kHz EQ control adjusted from about –20 dB to about +12 dB during the course of the recording.

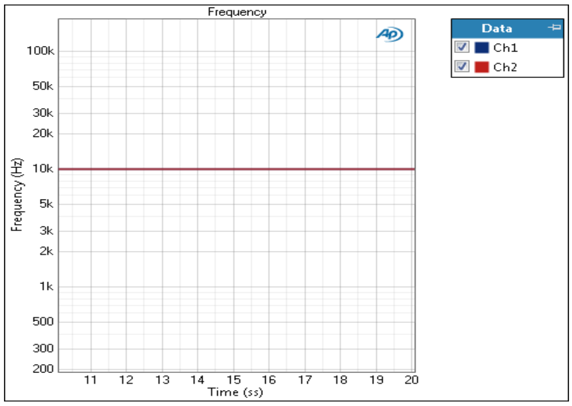
Units

Units available for Bandpass Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*	*Jitter*
• s	• Vrms	• dBFS	• UI
	• dBV	• FS	• dBUI
	• dBu	• %FS	• s
	• dBrA	• dBrA	
	• dBrB	• dBrB	
	• dBSPL1	• dBSPL1	
	• dBSPL2	• dBSPL2	
	• dBm	• D	
	• W (watts)		
	• V		

Frequency

The Measurement Recorder: Frequency measurement result will provide a record of frequency versus elapsed time for any audio signal. The illustration shows a 20-second record of a 10 kHz sine wave on two channels, with a 10 kHz EQ control adjusted from about –20 dB to about +12 dB during the course of the recording.

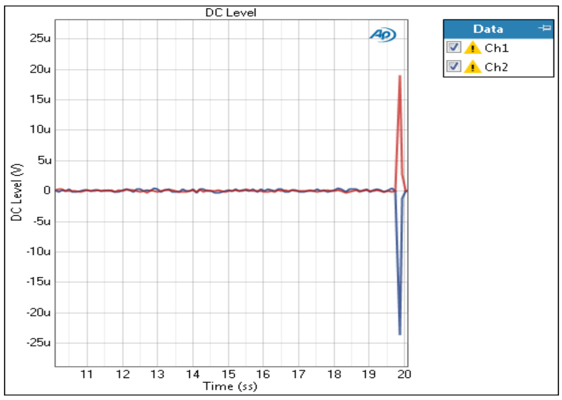


Units

Units available for Frequency are:

X-axis	*Y-axis*
• s	• Hz
	• dHz
	• F/R
	• %Hz

DC Level



The DC Level result is not available when the input is AC coupled.

The Measurement Recorder: DC Level measurement result will provide a record of DC level versus elapsed time for any signal. The Measurement Recorder does not require a specific test signal. It can be used with any audio signal within the input range of the analyzer.

Note: Although the Measurement Recorder result includes Generator and Filter controls, the DC Level measurement incorporates a very tight low-pass filter that rejects *any audio signal and makes the analysis filters irrelevant. Measurement Recorder: DC Level measurements will return the same results whether or not there is an audio signal in the DUT, or whether or not there is filtering applied.*

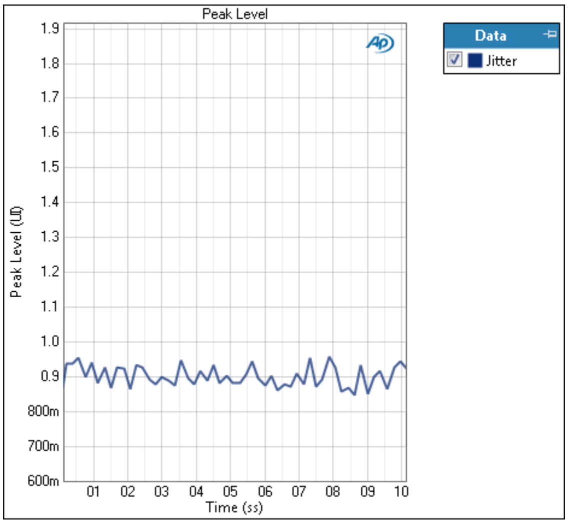
Go to page 240 for more information about DC Level.

Units

Units available for DC Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• s (seconds)	• V	• D • hex

Peak Level



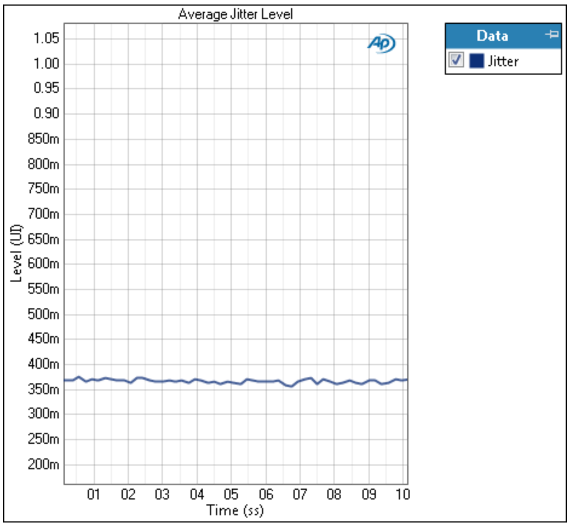
The **Measurement Recorder: Peak Level** result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is generator level; Y axis is the bandpass filtered DUT output level, scaled in peak units.

Units

Units available for Peak Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis (analog)*	*Y-axis (digital)*
• Vrms	• dBFS	• V	• D
• Vp	• FS		
• Vpp	• %FS		
• dBV	• dBrG		
• dBu		*Y-axis (jitter)*	
• dBrG		• UI	
• dBm		• dBUI	
• W (watts)		• s	

Average Jitter Level



The **Measurement Recorder: Average Jitter Level** result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is generator level; Y axis is the bandpass filtered DUT output jitter level, scaled in average units.

Units

Units available for Average Jitter Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis (jitter)*
• Vrms	• dBFS	• UI
• Vp	• FS	• dBUI
• Vpp	• %FS	• s
• dBV	• dBrG	
• dBu		
• dBrG		
• dBm		
• W (watts)		

Recording Audio to a File

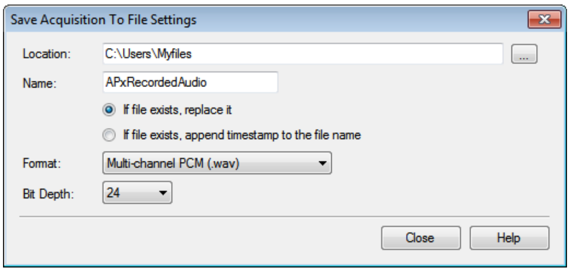
The Measurement Recorder and the Noise Recorder can optionally record acquired audio to a file on disk.

Note: APx timing constraints will sometimes cause a timeout error when saving to a network folder or a removable disk. We recommend saving audio files to a local disk.

On the Measurement or Noise Recorder panel, check the **Save to File** checkbox. Click **File Settings** to specify the file name, location and file format.

Save to File is not available when Input Configuration is set to File (Analog Units) or File (Digital Units).

Save Acquisition to File Settings dialog



Location

Enter or browse to the destination folder here.

Name

Name the output file here. Choose from one of these selections:

- If file exists, replace it
- If file exists, append timestamp to the file name

Generator Trigger

When Signal Generation is set to **Waveform**, there is always some delay between Measurement Recorder **Start** and the instant the waveform playback begins. When **Generator Trigger** is **Off** (unchecked), the file recording begins at the measurement **Start**. When **Generator Trigger** is **On** (checked), the file recording begins at the instant the waveform playback begins.

Format

Choose the file format in the **Format** field.

For both analog inputs and digital acquisitions, there are three linear file formats available:

Multiple Mono PCM (.wav)

If **Multiple Mono PCM (.wav)** (the default) is selected, each audio channel of the acquisition is saved as an individual monaural WAV file, using the Microsoft WAVE_FORMAT_PCM (Type 1, 24 bit) file format. For multichannel audio of channel count n, n mono WAV files are created, with the numeral n appended to the file name.

Multi-channel PCM (.wav)

If **Multi-channel PCM (.wav)** is selected, all the audio channels of the acquisition are saved in one multi-channel file, using the WAVE_FORMAT_PCM format. For 1 channel and 2 channel acquisitions, this selection records a standard WAVE_FORMAT_PCM (Type 1, 24 bit). For multichannel acquisitions, this selection records a multichannel WAVE_FORMAT_PCM file in a proprietary 24-bit format, compatible with Sony Sound Forge and other audio editing applications.

Extensible Multi-channel PCM (.wav)

If **Save as Extensible Multi-channel PCM (.wav)** is selected, all the audio channels of the acquisition are saved in one multichannel file, using the Microsoft standard WAVE_FORMAT_EXTENSIBLE format, at 24 bits.

32-bit audio

The DSIO receiver will accept 32-bit audio. In such a case, the audio can be saved in any of the above formats, but the bit depth will be 32 bit fixed point.

Bit Depth

This field is only available for analog acquisitions. The default Bit Depth (word length) is 24 bits. You can also select 16 bits.

Analog scaling and sample rates

When recording an analog acquisition, proprietary scaling metadata is embedded in the WAV file. The scaling factor relates the analog voltage to the digital level (Vrms/FS), and is used for scaling the display of values when using the APx500 file input configuration, or the MATLAB script apx_read_wave.

For analog acquisitions, the sample rate is set to be the same as the sample rate determined by the analyzer input configuration analog bandwidth setting, set in Signal Path Setup, as follows:

Bandwidth	Sample Rate
• 20 kHz	• 44.1 kS/s
• 22 kHz	• 48 kS/s
• 45 kHz	• 96 kS/s
• 90 kHz	• 192 kS/s
• 250 kHz	• 624 kS/s
• 500 kHz	• 1.248 MS/s
• 1 MHz	• 2.496 MS/s

Input bandwidths beyond 90 kHz require the BW52 Option; 1 MHz bandwidth requires input set to 1 channel.

The sample rate can be changed in the Advanced Settings dialog.

Duration for linear recordings

Windows WAV files are limited to 4 GB in size. For 24-bit stereo WAV files, this is approximately 1 hour at 192 kHz. Mono files and files with lower sample rates or bit depths can have longer durations. For longer recordings, APx automatically creates a sequential file (or a set of sequential files) providing a very long recording capability without loss of any data.

Note: The sequential file feature for very long recordings is resource-intensive, and requires a high performance level in terms *of PC CPU and hard disk speed, beyond the minimum recommended computer required to run APx500.*

Filtering

The Filtering set on the Measurement Recorder/Noise Recorder panel is applied to the measurement, but the audio is recorded without filtering. AC/DC coupling settings apply, however.

Bit depth and sample rate for digital acquisitions

For digital acquisitions, the record sample rate and bit depth are set to match that of the incoming digital signal.

Coded formats available for Measurement Recorder digital acquisitions

Coded audio formats are only available as Save to File selections for digital acquisitions in Measurement Recorder.

Coded audio (typically multichannel Dolby or DTS surround sound, coded in an IEC61937 bitstream) is saved in the incoming format. Only coded formats currently supported by the APx waveform generator are supported by Save to File. When connected to the HDMI Sink input, recordings cannot be made if HDCP is enabled at the playback device. See Compare Encoded Bitstream to Reference in Chapter 27 to evaluate bitstreams acquired with HDCP enabled.

First, select the expected coded format from the list:

- Dolby Digital (.ac3)
- Dolby Digital + (.ec3)
- DTS 5.1 Compact (.cpt)
- DTS 5.1 (.dts)
- DTS-HD (.cpt)
- Dolby TrueHD (.mlp)

When you click Start, Measurement Recorder waits until the selected coded bitstream is detected, and then begins analysis and saves the coded audio to the named file.

This pause accommodates DUTs such as Blu-ray Disc players, which often output PCM audio while idling between tracks.

If you choose

- Auto Detect Encoded

Measurement Recorder waits until the any of the listed coded bitstreams is detected before analyzing and saving.

Alternatively, you can select

- Raw Encoded Bitstream (.bin)

This saves the raw digital output of the receiver chip as a binary file for analysis using third-party software tools.

Duration for coded recordings

Coded file length is limited only by computer storage capacity, to the maximum duration of the Measurement Recorder acquisition of 7 days. Hundreds of gigabytes of storage are necessary for extended times.

Filtering

Coded waveforms are saved without any processing or modification.

More about Reading Rate

A reading is taken at the beginning of the record and at (approximately) the reading rate throughout the record. The final reading is taken at the reading instant that equals or exceeds the sweep end time.

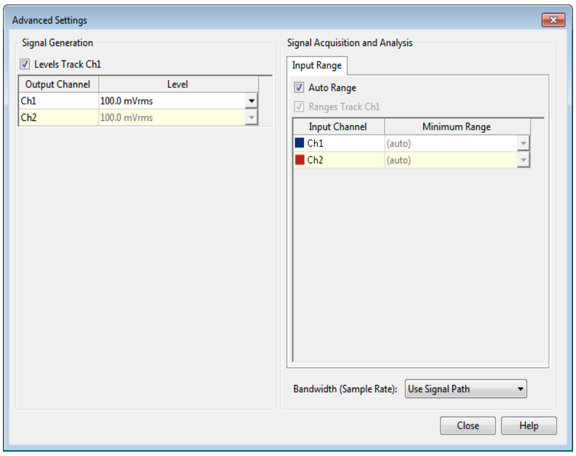
The reading rate also has an effect on analyzer performance at low frequencies. Each reading rate has a low-frequency point below which analyzer performance falls off. THD+N performance degrades rapidly below this point, with level accuracy degrading a little more slowly.

Measurement Recorder and Noise Recorder accuracy limits, by reading rate and frequency:

- at 20/sec > 60 Hz
- at 10/sec > 30 Hz
- at 5/sec > 15 Hz
- at 3/sec > 9 Hz
- at 2/sec > 6 Hz
- at 1/sec > 3 Hz
- at 0.5/sec > 1.5 Hz
- at 0.2/sec > 0.6 Hz
- at 0.1/sec > 0.3 Hz

Advanced Settings for Measurement Recorder

The default settings here are appropriate for most measurements. You may want to make minor adjustments for special situations.



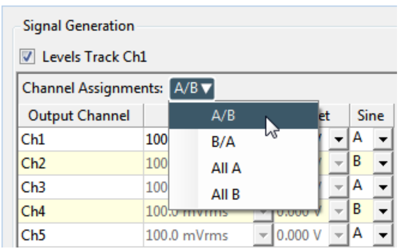
Signal Generation

If **Track first channel level** is checked (the default), the generator output level value for channel 1 is copied to the other channels, and output levels for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels.

To set individual generator output channel levels, uncheck the "Track first channel level" checkbox and enter values in the output channel Level fields.

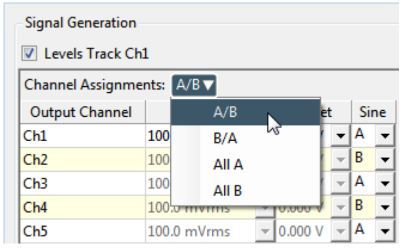
Set Channel Assignments for special waveforms

For **Split Sine** special waveform generation, you can set channel mapping. By default, channel A audio is mapped to odd numbered output channels, channel B audio to even numbered output channels. These assignments can be remapped by changing individual settings in the **Sine** column, or by selection one of several presets from the **Set Channel Assignments** menu.



Set Channel Assignments for generator waveforms

For stereo or multichannel generator waveform files, you can set channel mapping. By default, channel 1 audio is mapped to channel 1 output, channel 2 to channel 2, and so on. If the number of channels in the waveform file is less than the number of output channels, the waveform channels resume numbering at 1 and wrap to the next available output channel. These assignments can be remapped by changing individual settings in the **Wave Ch** column, or by selection one of several presets from the **Set Channel Assignments** menu.



Loop Waveform

By default, generator waveforms are looped in the generator, providing a continuous generator output. If Loop Waveform is unchecked, the generator waveform is played once, and then ends.

Signal Acquisition and Analysis

Input Range tab

Digital inputs do not require ranging, and the Input Range tab is unavailable when inputs are set to a digital format.

See page 551 for more information about ranging and autoranging.

You can set a fixed range for each analog input channel for the Measurement Recorder. If "Track first channel range" is checked (the default), the fixed input range setting for channel 1 is copied to the other channels, and range settings for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels.

To set individual analog input channel ranges, uncheck the "Track first channel range" checkbox and enter values in the input channel Range fields.

Fixed Notch Filter for THD+N Measurement

You can set a fixed notch filter for Measurement Recorder measurements. Check the **Fix Notch Filter for THD+N Measurement** checkbox and set a filter frequency in the **Notch Frequency** field. Notch frequency range depends upon the system input bandwidth setting.

- For input bandwidth = 90 kHz, notch frequencies can be in the range of 5 Hz to 80 kHz.
- For input bandwidth = 45 kHz, notch frequencies can be in the range of 5 Hz to 40 kHz.

- For input bandwidth = 22 kHz, notch frequencies can be in the range of 5 Hz to 20 kHz.

Bandwidth (Sample Rate)

This control is only available when the analyzer input configuration is set to Analog.

The Bandwidth (Sample Rate) control allows you to override the analog input bandwidth setting in Signal Path Setup. The input bandwidth set here operates only on the Measurement Recorder.

Certain applications that may use the files created here require files of specific sample rate. An example would be the recording of perceptual evaluation signals, which are often required to be at 8 kHz or 16 kHz sample rate. For analog inputs, the default sample rate of a recorded file is determined by the input bandwidth setting, normally set in Signal Path Setup. This control allows selection of other bandwidths/ sample rates. The analog input bandwidth (sample rate) settings, whether made here or in Signal Path Setup, result in channel-count constraints at high sample rates.

The bandwidth choices, followed by their corresponding sample rates, are:

- 2.75 kHz (6 kS/s)
- 3.5 kHz (8 kS/s)
- 5.5 kHz (12 kS/s)
- 7 kHz (16 kS/s)
- 11 kHz (24 kS/s)
- 20 kHz (44.1 kS/s)
- 22 kHz (48 kS/s)
- 40 kHz (88.2 kS/s)
- 45 kHz (96 kS/s)
- 80 kHz (176.4 kS/s)
- 90 kHz (192 kS/s)
- 250 kHz (624 kS/s) (requires the BW52 Option)
- 500 kHz (1.248 MS/s) (requires the BW52 Option)
- 1 MHz (2.496 MS/s) (requires the BW52 Option, set to 1 channel)
- Use Signal Path (the default)

Note that recording files with bandwidths of 250 kHz or greater (with the BW52 Option at sample rates at or above 624 kHz) requires a very fast PC and hard disk, due to the high data rates. Slower systems can suffer recording interruptions and failure.

58

Metadata Recorder
(Sequence Mode)

Introduction

For the APx515, this measurement requires a software option key. See page 166 for more information about software options.

The Metadata Recorder measurement records the state of selected metadata settings in the incoming digital audio signal. If one of the selected settings changes state during the acquisition, the transition instant and the new state are recorded and displayed.

This measurement is only available when the input is set to digital unbalanced, digital balanced, digital optical, or HDMI.

Operation

If you have not yet set up your test, first go to Signal Path Setup (Chapter 6).

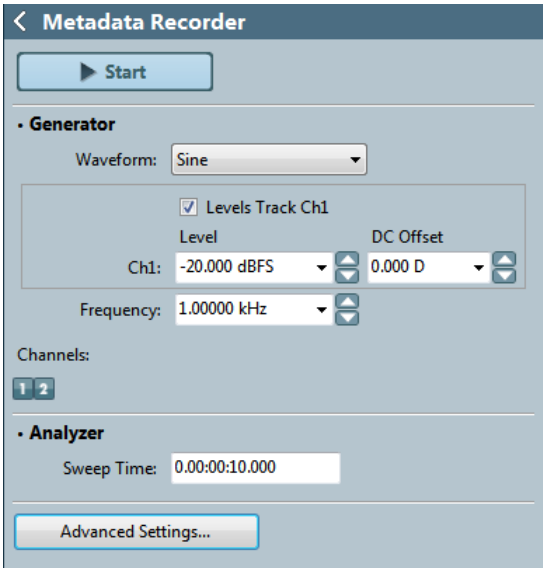
Generator

The typical configuration for this measurement is the closed-loop configuration, using the APx generator as a stimulus. See APx Generator Waveforms and Controls for detailed information about the using the APx Generator and setting Waveform, Level and Frequency.

Alternatively, you can make open-loop measurements, using either the External Source or File Input configuration.

Linear waveforms

Since the Metadata Recorder only reads metadata information, the audio in the transport stream is largely irrelevant. Chose the internal generator or a linear generator waveform. For a closed-loop test, you can set the metadata for the transport stream in Signal Path Setup > Output Configuration > Digital [Unbalanced] [Balanced] [Optical] [HDMI] > Settings > Status/User Bits or Audio InfoFrame.



Coded waveforms

For a coded waveform, you must choose a generator waveform of a coded type, such as AC3 or DTS. The coding metadata is embedded in the waveform file, and the transport stream Validity bit is set (indicating not linear audio). You can set other metadata for the transport stream as discussed above.

Analyzer

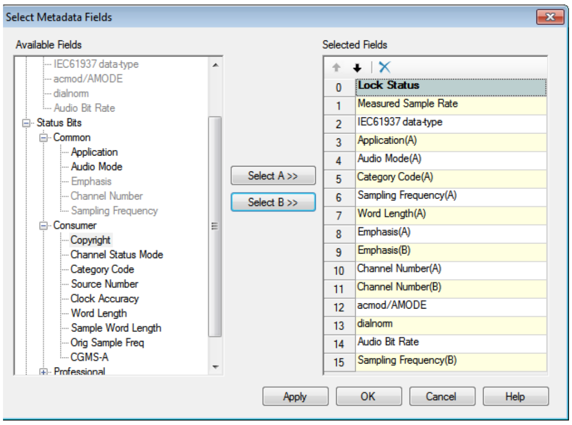
Acquiring the metadata

Set Sweep Time for the duration you would like to display in the log. The default is 10 s. Click Start. The initial state of each metadata field is displayed by a labeled bar. If a field changes state, a transition is indicated, and the label after the transition indicates the new state. The transition is marked with a time stamp referenced to the beginning of the sweep.

The X-axis of the Metadata Recorder is labeled in seconds, but the scale does not represent a linear flow of time. Instead, (when X-Axis Range is set to **Auto**) the total time of the sweep is shown across the breadth of the graph, and the graph is divided into segments left to right that correspond to the number of transitions. Each segment is marked with the instant of transition.

Select Fields (in the Results display)

You can choose up to sixteen metadata fields for display. Click **Select Fields** to open the selection dialog.



In a New Project, the default fields are

- Lock Status
- Measured Sample Rate
- IEC61937 data-type
- Application(A) [Status Bits]
- Audio Mode(A) [Status Bits]
- Category Code(A) [Status Bits]
- Sampling Frequency(A) [Status Bits]
- Word Length(A) [Status Bits]

Adding fields

Click **Select Fields...** to open the selection dialog.

Open a branch in the tree in the left panel and select a metadata field. In the right panel, select the row in which you would like the data to be displayed. Click the **Select>>** button to assign the field to the display. For Status Bits, choose the **Select A>>** or **Select B>>** button for the subframe you want to display, A or B.

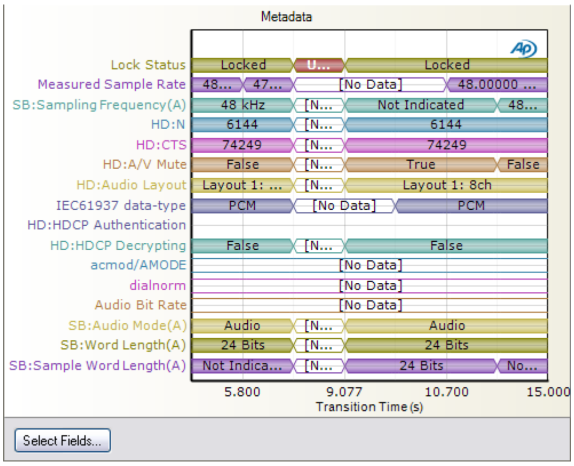
A default color is assigned to each row. Click the browser button at the right of the row to change the display color for that row.

You can remove an entry by clicking the **Delete** button at the top of the right panel. Use the **Up** and **Down** arrows to change the order of the fields.

Abbreviations used in the Metadata Recorder display

"SB" stands for status bits. "(A)" and "(B)" refer to the status bits subframe designation. "HD" stands for HDMI.

Metadata Recorder results



The initial state of each metadata field is displayed by a labeled bar. If a field changes state, a transition is indicated, and the label after the transition indicates the new state. The transition is marked with a time stamp referenced to the beginning of the sweep.

The X-axis of the Metadata Recorder is labeled in seconds, but the scale does not represent a linear flow of time. Instead, (when X-Axis Range is set to Auto) the total time of the sweep is shown across the breadth of the graph, and the graph is divided into segments left to right that correspond to the number of transitions. Each segment is marked with the instant of transition.

Units

Units available for Metadata Recorder are

- s

See Chapter 98 for more information about units of measurement.

More About Audio Metadata

> *Note: for this topic, the designation "IEC60958" represents all the largely compatible audio transport streams, including AES/EBU, AES3, S/PDIF, SMPTE 276M and so on.*

Metadata has been defined as "data about other data." Some digital audio files and digital transport streams carry metadata that describe aspects of the stream and the audio content.

Transport streams and metadata

IEC60958

This industry standard transport stream carries audio samples, embedded clock and metadata. For each subframe (or channel) of audio, there is a synchronizing/identification Preamble, followed by audio data, followed by a Validity Bit, a User Bit, a Channel Status Bit (normally just called a Status Bit) and a Parity Bit. The User Bits and Status Bits are collected over 192 frames, providing metadata in the form of a block of 384 user bits and Subframe A and Subframe B blocks of 192 channel status bits.

Coded audio streams

Low-bit rate coded audio such as Dolby Digital (AC3) and DTS Digital Surround can be embedded in the IEC60958 transport stream as defined in the standard IEC61937. In such cases, the Validity Bit is set, indicating a non-PCM signal. Additionally, the embedded IEC61937 audio carries its own metadata, specific to the coding format.

HDMI: Stream metadata

The HDMI transport stream carries a great deal of data and metadata. Although in audio test and measurement we are not concerned with much of the HDMI signal, the video formats and high-level stream metadata do affect a receiver's ability to correctly accept embedded audio, and these metadata settings and readings are exposed in APx500.

HDMI: Audio InfoFrame

HDMI carries metadata in InfoFrames, and APx500 can set and read Audio InfoFrame data.

Embedded audio stream metadata

Although audio on HDMI is packetized and multiplexed into a high-bit rate video stream, the format of the audio mimics IEC60958 audio and includes user bits, status bits and metadata related to coded audio.

EDID

HDMI also carries EDID, which is not directly related to the audio but affects how a downstream device may react to the transport stream.

HDCP

HDMI streams are often encrypted using HDCP encryption, indicated by a metadata flag.

Audio streams that do not include metadata

Analog audio does not carry any digital metadata. Also, pure PCM sampled audio data is simply a pulse train of samples, and does not carry any metadata. I2S, TDM and other chip-level protocols are of this type; thus, the signals passing in or out of the DSIO ports have no metadata.

Audio files and metadata

Linear files

Generally speaking, linear audio files such as WAV files do not contain metadata. There are housekeeping data that all files contain so that the computer can work with the files, but not metadata about the audio content.

That being said, there are variants on WAV files that contain channel allocation information or specialized broadcast production information. It is even possible to package coded audio with its attendant metadata in a WAV file. But common WAV files, and the WAV linear test signals we provide, do not carry metadata.

When such a file is routed out through a transport stream transmitter (e.g., IEC60958 or HDMI) metadata is added by the transmitter.

Coded audio files

Coded audio files such as AC3, DTS, MLP and DTSHD contain the proprietary metadata for the coding format. When such a file is routed out through a transport stream transmitter (e.g. IEC60958 or HDMI), the coding metadata is maintained as embedded IEC61937 metadata, and transport stream metadata is added by the transmitter.

IEC60958 Status Bits and User Bits

Overview

The family of digital interface formats variously known as IEC60958, AES3, AES/EBU, S/PDIF, and SMPTE 276M share a common history of development by standards committees made up of audio experts from around the world. The formats are, for the most part, compatible. The primary differences are in the physical interface and in the use and interpretation of channel status bits.

In practice, if the physical interfaces are compatible and the sample rate and bit depth are supported, a signal transmitted conforming to any of these standards will almost always be converted to audio in a receiving device.

Status Bits

The channel status information is carried in the Status Bits. Apart from the first two bits, the meaning of the bits is defined differently for the consumer and professional formats. See the tables at the end of this chapter.

User Bits

The user bit can be used to carry user-specific information.

Status bits and user bits over HDMI

Although the physical layer of the high-speed, packeted transport signal of HDMI is completely incompatible with any of the above interfaces, nonetheless the supported audio protocols at the input or output of HDMI transmitters or receivers are compliant with IEC60958 and the coded audio layer IEC61937. Channel status bits and user bits are supported within the audio streams carried by HDMI. The default for HDMI is the consumer application.

HDMI and HDMI Audio InfoFrame

Overview

The HDMI transport stream carries a great deal of information, including video data, audio data and extensive metadata, including several types of InfoFrames that describe various aspects of the data in the transport stream.

Of interest in audio test and supported in APx500 is the Audio InfoFrame, which contains information about the audio format, with additional channel allocation and downmix information for linear PCM audio.

“Refer to Stream Header”

Note that the CEA-861-D standard that defines Audio InfoFrame supports HDMI, but also other transport streams. Some fields of the Audio InfoFrame are considered redundant in the HDMI application and are required to be set to 0, “Refer to Stream Header”. “Stream Header” indicates the IEC60958 channel status bits and, for coded streams, the IEC61937 burst info data.

Audio InfoFrame fields

Coding Type (CT)

In HDMI use, audio Coding Type is considered redundant and is required to be set to 0 (“Refer to Stream Header”). For testing purposes, APx500 can ignore this requirement and transmit audio Coding Type indicators at any supported setting. Coding Type and Channel Count share Data Byte 1.

Channel Count (CC)

Audio channel count can be set to 0 (“Refer to Stream Header”), or to 2–8 channels for multichannel linear PCM use. For testing purposes, APx500 can transmit channel count indicators at any supported setting. Coding Type and Channel Count share Data Byte 1.

Sampling Frequency (SF)

In HDMI use, audio sampling frequency is considered redundant and is required to be set to 0 (“Refer to Stream Header”). For testing purposes, APx500 can ignore this requirement and transmit sampling frequency indicators at any supported setting. Sampling Frequency and Sample Size share Data Byte 2.

Sample Size (SS)

In HDMI use, audio sample size is considered redundant and is required to be set to 0 (“Refer to Stream Header”). For testing purposes, APx500 can ignore this requirement and transmit sample size indicators at any supported setting. Sampling Frequency and Sample Size share Data Byte 2.

Speaker Allocation (CA)

The speaker allocation field is provided for Layout 1 multichannel LPCM streams, which do not carry speaker data. Metadata is embedded in the HDMI signal indicating the desired mapping of channels 1–8 to one of 32 loudspeaker arrangements. For testing purposes, APx500 can transmit speaker allocation data at any supported setting. Speaker Allocation is carried in Data Byte 4.

Level Shift Value (LSV)

The level shift field is provided for multichannel linear PCM streams, which do not carry downmix data. For testing purposes, APx500 can transmit level shift data at any supported value. Level Shift Value and Downmix share Data Byte 5.

Downmix (DM)

The downmix inhibit field is provided for multichannel linear PCM streams, which do not carry downmix data. For testing purposes, APx500 can transmit downmix permitted (0) or prohibited (1). Level Shift Value and Downmix share Data Byte 5.

In APx500, outgoing Audio InfoFrame metadata can be set in the HDMI Output Settings dialog; see page 113. Incoming Audio InfoFrame metadata can be read in the Metadata Monitor: HDMI; see page 33.

Metadata tables

Status Bits consumer interpretation

See the latest edition of the standard IEC60958 Part 3 for more information.

<table>
<tr><th>BITS</th><th>LABEL</th><th colspan="3">INTERPRETATION</th></tr>
<tr><td>0</td><td>application: consumer or professional</td><td colspan="3">0: consumer; 1: professional</td></tr>
<tr><td>1</td><td>non-audio</td><td colspan="3">0: audio data is linear PCM samples
1: other than linear PCM samples</td></tr>
<tr><td>2</td><td>copyright</td><td colspan="3">0: asserted; 1: not asserted</td></tr>
<tr><td>3-5</td><td>emphasis</td><td colspan="3">000: Emphasis not indicated
100: CD-type emphasis</td></tr>
<tr><td>6-7</td><td>channel status mode</td><td colspan="3">00: mode zero; other values reserved</td></tr>
<tr><td>8-15</td><td>category code</td><td colspan="3">(bit 8 is LSB)</td></tr>
<tr><td>16-19</td><td>source number</td><td colspan="3">(bit 16 is LSB)</td></tr>
<tr><td>20-23</td><td>channel number</td><td colspan="3">(bit 20 is LSB)</td></tr>
<tr><td>24-27</td><td>sampling frequency</td><td colspan="3">0000: 44.1 kHz
0100: 48 kHz
1100: 32 kHz</td></tr>
<tr><td>28-29</td><td>clock accuracy</td><td colspan="3">10: Level I, ±50 ppm
00: Level II, ±1000 ppm
01: Level III, variable pitch shifted</td></tr>
<tr><td>30-31</td><td>reserved</td><td colspan="3"></td></tr>
<tr><td>32</td><td>word length
(field size)</td><td colspan="3">0: Maximum length 20 bits
1: Maximum length 24 bits</td></tr>
<tr><td>33-35</td><td>word length</td><td>
000:
101:
001:
010:
011:
100:</td><td>if bit 32 = 1
not indicated
24 bits
23 bits
22 bits
21 bits
20 bits</td><td>if bit 32 = 0
not indicated
20 bits
19 bits
18 bits
17 bits
16 bits</td></tr>
<tr><td>36-39</td><td>reserved</td><td colspan="3"></td></tr>
<tr><td>40-191</td><td>reserved</td><td colspan="3"></td></tr>
</table>

Status Bits professional interpretation

See the latest edition of the standard IEC60958 Part 4 for more information.

BITS	*LABEL*	*INTERPRETATION*
0	application: consumer or professional	0: consumer; 1: professional format
1	non-audio	0: audio data is linear PCM samples 1: other than linear PCM samples
2-4	emphasis	000: Emphasis not indicated 100: No emphasis 110: CD-type emphasis 111: J-17 emphasis
5	lock	0: not indicated 1: unlocked
6-7	sampling frequency	00: not indicated (or see byte 4) 10: 48 kHz 01: 44.1 kHz 11: 32 kHz

<table>
<tr><td>8-11</td><td>Channel mode

(SCDSR represents “single channel double sample rate”)</td><td colspan="3">0000: not indicated (default to 2 ch)
0001: 2 channel
0010: 1 channel (monophonic)
0011: primary / secondary
0100: stereo
0101: reserved for user applications
0110: reserved for user applications
0111: SCDSR (see byte 3 for ID)
1000: SCDSR (stereo left)
1001: SCDSR (stereo right)
1111: Multichannel (see byte 3 for ID)</td></tr>
<tr><td>12-15</td><td>user bit management</td><td colspan="3">0000: no indication
0001: 192-bit block as in channel status
0010: As defined in AES18
0011: user-defined
0100: as in IEC60958-3 (consumer)</td></tr>
<tr><td>16-18</td><td>use of aux sample word</td><td colspan="3">0000: not defined, audio max 20 bits
0001: used for main audio, max 24 bits
0010: used for coordination signal, audio max 20 bits
0011: user-defined</td></tr>
<tr><td>19-21</td><td>source word length</td><td>
000:
001:
010:
011:
100:
101:</td><td>if max = 24 bits
not indicated
23 bits
22 bits
21 bits
20 bits
24 bits</td><td>if max = 20 bits
not indicated
19 bits
18 bits
17 bits
16 bits
20 bits</td></tr>
<tr><td>22-23</td><td>alignment level</td><td colspan="3">00: not indicated
01: –20 dB FS
10: –18.06 dB FS</td></tr>
<tr><td>24-31</td><td>channel identification</td><td colspan="3">if bit 31 = 0 then channel number is 1 plus the numeric value of bits 24-30.
if bit 31 = 1 then bits 4–6 define a multichannel mode and bits 0–3 give the channel number within that mode.</td></tr>
<tr><td>32-33</td><td>digital audio reference signal (DARS)</td><td colspan="3">00: not a DARS
10: DARS grade 2 (+ / –10 ppm)
01: DARS grade 1 (+ / –1 ppm)</td></tr>
<tr><td>35-38</td><td>sampling frequency</td><td colspan="3">0000: not indicated
1000: 24 kHz
0100: 96 kHz
1001: 22.05 kHz
0101: 88.2 kHz
1101: 176.4 kHz
1111: User defined</td></tr>
<tr><td>39</td><td>sampling frequency scaling</td><td colspan="3">0: no scaling
1: apply factor of 1 / 1.001 to value</td></tr>
<tr><td>48-79</td><td>alphanumerical channel origin</td><td colspan="3">four-character label using 7-bit ASCII with no parity. Bits 55, 63, 71, 79 = 0.</td></tr>
<tr><td>80-111</td><td>alphanumerical channel destination</td><td colspan="3">four-character label using 7-bit ASCII with no parity. Bits 87, 95, 103, 111 = 0.</td></tr>
<tr><td>112-143</td><td>local sample address code</td><td colspan="3">32-bit binary number representing the sample count of the first sample of the channel status block.</td></tr>
<tr><td>144-175</td><td>time of day code</td><td colspan="3">32-bit binary number representing time of source encoding in samples since midnight.</td></tr>
<tr><td>176-183</td><td>reliability flags</td><td colspan="3">0: data in byte range is reliable
1: data in byte range is unreliable</td></tr>
<tr><td>184-191</td><td>CRCC</td><td colspan="3">00000000: not implemented
nnnnnnnn: error check code for bits 0–183</td></tr>
</table>

59

Modulated Noise
(Sequence Mode)

This measurement requires a software option key. See page 166 for more information about software options.

The Modulated Noise measurement is specifically designed to detect air leaks in a sealed loudspeaker/enclosure system. Modeling an air leak as a puff of amplitude-modulated noise, this measurement uses a low-frequency stimulus to excite the system, and applies filtering and AM detection to analyze the speaker output and provide a modulated noise to unmodulated noise ratio result. Using the Audio Precision implementation, a well-sealed system in a quiet room will have a modulated noise close to 0 dB; leaky enclosures will present modulated noise ratios of 5 dB or more, to as high as 30 dB.

See More about Loudspeaker Testing on page 185 for a more detailed discussion of using modulated noise in loudspeaker testing.

Operation

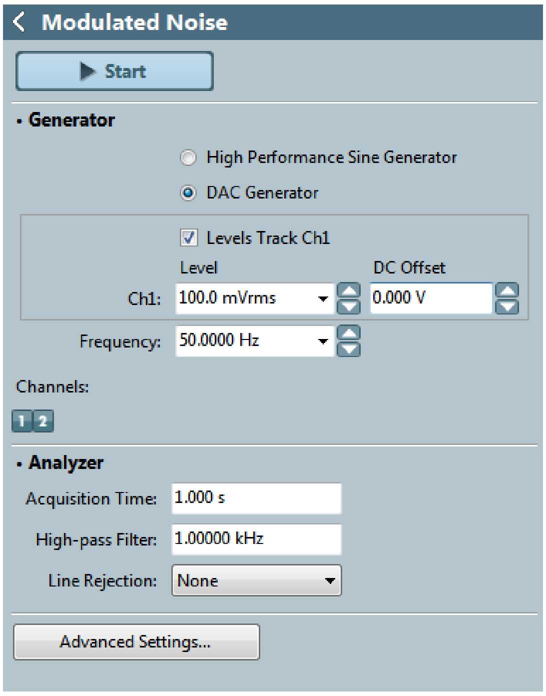
If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 6 for information about Signal Path Setup.

Generator

This measurement must be performed in the closed-loop configuration, using the APx generator as a stimulus.

Selecting the generator waveform

By default, the Modulated Noise measurement uses a 50 Hz sine wave at the frequency and level set in the Generator panel as the test signal. You can set other frequencies, but low frequencies are the most effective in detecting air leaks. It is important to drive the loudspeaker with enough level to get the air moving in the enclosure.



See Chapter 5 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 49 for a comparison of features.

Analyzer

The acquired signal is first comb-filtered, with a comb set at the generator frequency and its harmonics.

Acquisition Time

Set the acquisition time here.

High-pass filter

Use this filter to reduce environmental noise. The default corner frequency is 1 kHz, with a range of 10 Hz to 3 kHz. Adjust this filter to find the highest modulated noise ratio. When set, the local high-pass filter acts on the acquired audio, after comb filtering. Read about Modulated Noise comb filtering in More About Loudspeaker Testing on page 185.

Low pass, high pass and weighting Signal Path Filters settings (see page 545) made in Signal Path Setup > Input/Output will also affect this measurement. Unlike other measurements, setting the local high-pass filter in Modulated Noise does not override the high-pass filter in Signal Path Setup.

If a filter in Signal Path Setup is set to reduce or eliminate a frequency range that includes the Modulated Noise stimulus (set in Generator > Frequency), the Modulated Noise measurement may fail to make an acquisition.

Line Rejection

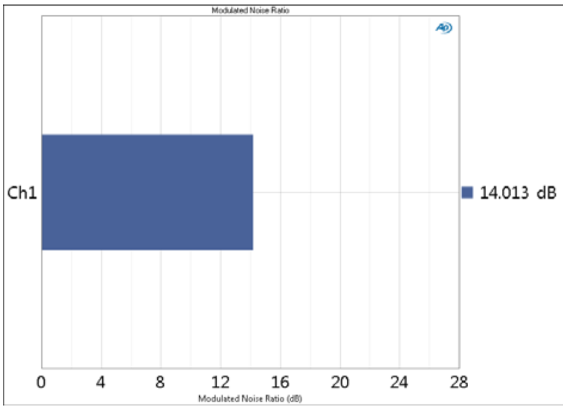
This optional filter will notch out the mains power line frequency and its harmonics, if necessary. Set the filter to the nominal mains frequency in your region, 50 Hz or 60 Hz.

Advanced Settings

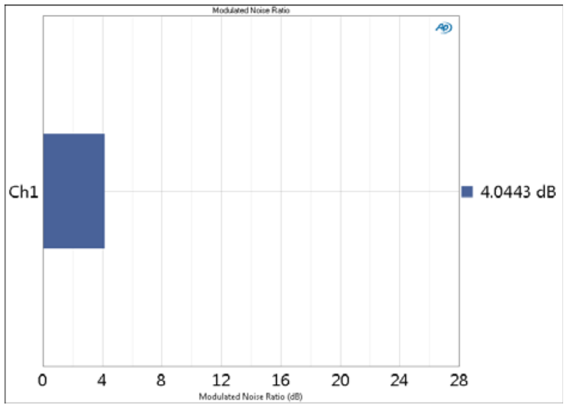
Advanced Settings makes available individual channel output level and offset controls, and input ranging and settling controls. See Advanced Settings for single value measurements on page 317.

See Chapter 98 for more information about units of measurement.

Modulated Noise Ratio



Loudspeaker enclosure with an air leak.



Same loudspeaker enclosure with air leak plugged.

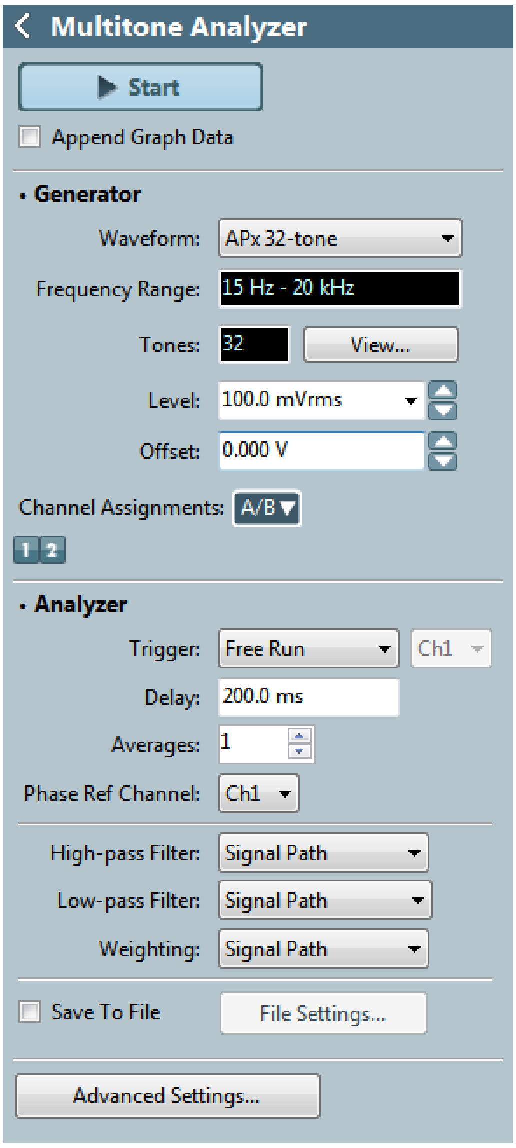
Units

Units available for Modulated Noise Ratio are

- x/y
- dB

60

Multitone Analyzer
(Sequence Mode)



Introduction

For the APx515, this measurement requires a software option key. See page 166 for more information about software options.

The Multitone Analyzer uses a special stimulus signal called a *multitone*. Multitone analysis is very fast, and provides a broad range of results.

The Multitone Analyzer results available in APx500 are:

- Level
- Gain
- Relative Level
- Deviation
- Phase
- Phase (at Tone)
- TD+N Ratio
- TD+N Level
- TD+N Spectrum
- Crosstalk
- Crosstalk (at Tone Pair)
- Total Level
- Tone Level
- Maximum Level (at Tone)
- Noise Density Spectrum
- Noise RMS Level
- Signal to Noise Ratio
- DC Level
- Frequency Shift
- FFT Spectrum
- Acquired Waveform

For more information about multitone, go to page 373.

Operation

If you have not yet configured the instrument I/O for your DUT and test, first go to Signal Path Setup (Chapter 6). When Generator and Analyzer (below) are set for your test, click Start.

Append Graph Data

Measured data are grouped in a Data Set, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the Append Graph Data box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 573.

Generator

Waveform

The APx generator can create a multitone directly from a multitone signal definition, or it can play an audio (WAV) file that contains a multitone.

Using a pre-defined multitone signal definition

A number of pre-defined multitone signal definitions are available for selection from the Waveform drop-down menu. The illustration at left shows the pre-defined definition “APx 32-tone” as the selection. Pre-defined multitone signal definitions cannot be edited.

When a pre-defined multitone signal definition is currently selected, you can read the multitone signal’s Min Frequency, Max Frequency and the number of Tones in the display fields. Click the View button to open the View Multitone Signal Definition dialog, where you can see the definition in detail, export the definition as a data file, or use it to create and save an audio (WAV) multitone file.

Creating and using user-defined multitone signal definitions

You can create custom multitone signal definitions. From the Waveform drop-down menu, choose Create New... to open the Create Multitone Signal Definition dialog (page 376). Choose a signal definition as a starting point, then click OK to continue to the Edit Multitone Signal Definition dialog (page 377), where you can edit the definition in detail, export the definition as a CSV or XLS data file, or use it to create and save an audio (WAV) multitone file. Click OK to save your definition in the project and add it to the Waveform menu.

Loading a multitone signal definition from a file on disk

You can import a multitone signal definition from a file on disk. From the Waveform drop-down menu, choose Browse for file... and select a CSV or XLS multitone signal definition file using the file browser.

Using a linear audio file on disk as a generator waveform

You can use APx’s generator waveform feature with the multitone analyzer, much as you can with other measurements. From the Waveform drop-down menu, choose Browse for file... and select a linear audio (WAV) file using the file browser. For proper operation and analysis, the audio contained in the file must be a valid, well-formed multitone. For a linear (WAV) file, APx500 extracts a multitone signal definition from the audio in the file. This definition is required to provide the frequency, level and phase information necessary for analysis. The extracted definition can be examined by clicking the View button to open the View Multitone Signal Definition dialog (page 376). When the multitone is played by the APx generator, the original audio file is used for signal generation, not the extracted definition. See Generator Waveform files on page 163.

Using a coded audio file on disk as a generator waveform

Coded audio files (ac3, dts, etc.) can be used as generator waveform files. The file must contain coded audio derived from a valid and well-formed multitone signal. Additionally, the multitone signal definition data used to create the original multitone must be available to APx500. When a coded multitone file is selected, a new field called Signal Definition appears in the settings panel. Use this selector to choose a multitone signal definition that corresponds to the multitone in the coded file. This definition is required to provide the frequency, level and phase information necessary for analysis. See Generator Waveform files on page 163.

Description

This field is only displayed when a generator waveform is currently loaded in the generator. It displays information about the waveform file.

Signal Definition

This field is displayed in External Source configuration, or when a coded generator waveform is currently loaded in the generator. It provides a selector to choose a multitone signal definition that corresponds to the multitone in use (either the externally sourced multitone, or the multitone in the coded file). See Edit Multitone Signal Definition on page 377. This definition is required to provide the frequency information necessary for analysis.

Min Frequency / Max Frequency / Tones

These fields display information about the multitone definition currently loaded in the generator. See View Multitone Signal Definition (page 376) and Edit Multitone Signal Definition (page 377).

Level

Set the generator output level in this field.

The Level control in an APx measurement sets the level of the generator output to the rms value entered in the Level field. With a sine wave, the result is what you might expect: the generator outputs a single tone at 1 Vrms.

However, a multitone waveform is the sum of two or more sine waves of different frequency and phase, and possibly different level. These tones are summed, resulting in a complex waveform whose crest factor (page 381) will be higher than that of a sine wave. The Level control sets the amplitude of the complex waveform, but the individual component stimulus tone levels in a multitone signal are typically well below the rms level of the complex waveform.

Channel

The generator will output a multitone to the DUT on the selected generator channels (page 162).

Start

The Start button runs the measurement. For closed loop operation, Start generates the multitone signal and prepares the analyzer for acquisition. For External Source (open loop) measurements, Start prepares the analyzer for acquisition, awaiting Trigger.

Analyzer

Trigger

By default, the Multitone Analyzer Trigger is set to Free Run, which is the untriggered condition. In many cases, especially closed-loop configurations, this setting will provide fast, reliable and highly accurate multitone measurements. However, some use cases or testing environments will require other trigger configurations.

- Signal
  This trigger mode is the most discriminating. The incoming audio on a specified channel is compared to the multitone signal definition in memory, with selectable degrees of discrimination (see Channel, Delay, Match % and SNR, below). The analysis is only triggered when a match between the incoming audio and the multitone signal definition is accomplished, after any trigger Delay.
  Tones that are very close together (separated by only a few bins) are more difficult to trigger on.
- Generator
  This trigger mode begins analysis at the moment the internal generator outputs a signal, after any trigger Delay.
- Level
  This trigger mode begins analysis when the incoming audio on a specified channel exceeds the specified level, after any trigger Delay.
- Free Run (the default)
  This trigger has no conditions. Analysis begins when the Start button is clicked, after any trigger Delay.

Channel

For Signal and Level triggering, select the triggering channel here.

Delay

The trigger Delay inserts time between a successful trigger event and the actual start time of the acquisition. The range is 0.000 s to 1.000 s; default is 200.0 ms.

Level

For Level triggering, set a threshold Level here.

Match %

For Signal triggering, set the desired degree of matching here. The range is from 25 % (rough match) to 100 % (exact match).

SNR

For Signal triggering, set the required signal-to-noise ratio here. The range is from 20 dB to 100 dB.

Averages

For non-periodic signals such as noise, averaging multiple acquisitions can provide a more useful view. This control allows you to specify how many acquisitions to average. The default is 1; maximum is 8.

Phase Ref Channel

Select the reference channel for the phase measurement here. The phase of the other channels is calculated relative to the reference channel.

Filters

Local high-pass, low-pass and weighting filters are available for this measurement. The low-pass and high-pass filters, if selected, are applied to the entire measured signal and affect all the results. The weighting filter, if selected, affects only the distortion and distortion plus noise results. See Measurement Filters on page 547 for detailed information about the filters available locally for some measurements.

Low pass, high pass and weighting Signal Path Filters settings (see page 545) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass or high-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active. A local weighting filter setting does not override any Signal Path filter settings; the local weighting filter is applied in addition to any active Signal Path filters.

Save to File

You can record the multitone audio signal to a file as the signal is being acquired. Click **Save to File** before you run the multitone measurement. For multitone, only the frequency-shifted FFT record is saved, not the entire acquisition. See Recording Multitone Audio to a File on page 381 for detailed information.

File Settings

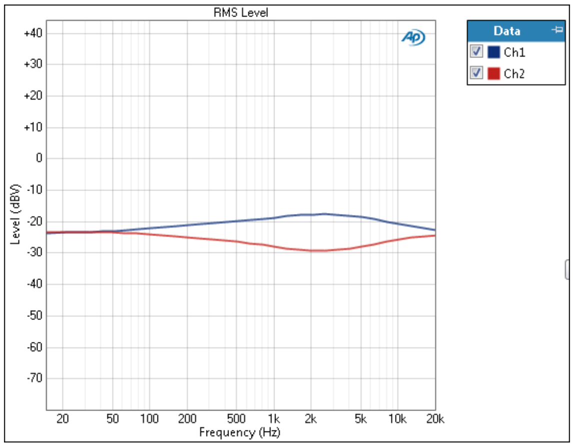
When saving a multitone acquisition to a file, click **File Settings** to specify the file name and location. For subsequent acquisitions, data are overwritten to the same file name, with no warning. To keep multiple acquisitions, rename the file for each acquisition.

Advanced Settings

If you'd like to adjust analyzer ranging or settling parameters, click **Advanced Settings**, where you will also find multitone-specific settings including A-B signal mapping, trigger timeout, maximum frequency shift and skirt width. See Chapter 98 for more information about units of measurement.

A multitone result provides a limited number of measured data points (one for each tone in the stimulus, as you can see in the graph data grid (page 573). Values between these data points implied by the plotted curve are interpolated for display.

Multitone Analyzer: Level



The **Multitone Analyzer: Level** result uses the multitone method to provide a graphical display of the level vs. frequency (frequency response) for each channel of the DUT. The rms value of each tone is measured, and a response curve is plotted through these values.

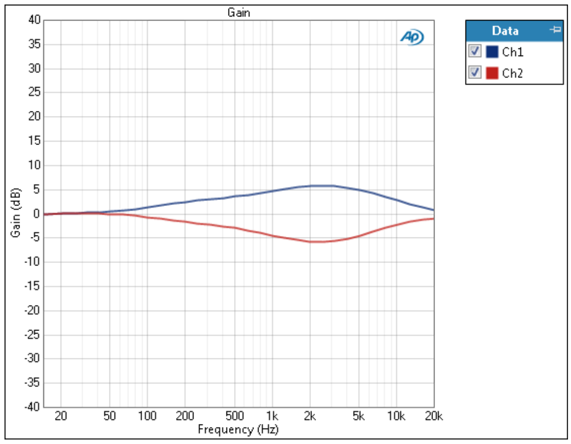
The **Skirt Width** control (page 381) in **Advanced Settings** can affect the values reported in this result.

Units

Units available for Level results are

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz	• Vrms	• dBFS
• dHz	• dBV	• FS
• F/R	• dBu	• %FS
• %Hz	• dBrA	• dBrA
	• dBrB	• dBrB
	• dBSPL1	• dBSPL1
	• dBSPL2	• dBSPL2
	• dBm	
	• W (watts)	

Multitone Analyzer: Gain



Gain measurements are not available when External Source is selected as the output configuration, when File is selected as the input configuration, or when a square wave or generator waveform file is selected as the test signal.

The **Multitone Analyzer: Gain** result uses the multitone method to provide a graphical display of the gain vs. frequency (gain response) for each channel of the DUT. The rms value of each tone is measured, and a response curve is plotted through these values.

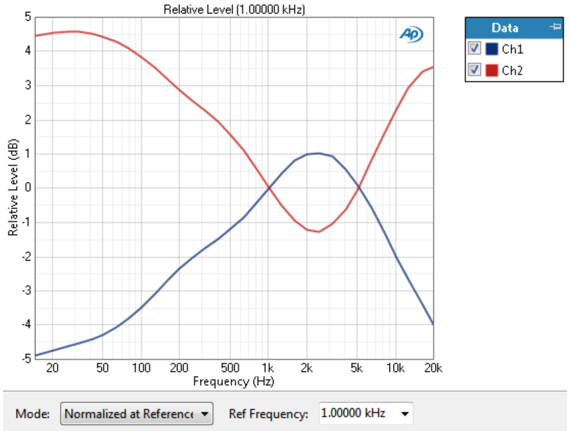
The **Skirt Width** control (page 381) in **Advanced Settings** can affect the values reported in this result.

Units

Units available for Multitone: Gain are

X-axis	*Y-axis (same-domain)*	*Y-axis (cross-domain)*
• Hz	• x/y	• FS/Vrms
• dHz	• %	• dB(FS/Vrms)
• F/R	• ppm	or
• %Hz	• dB	• Vrms/FS
		• dB(Vrms/FS)

Multitone Analyzer: Relative Level



The **Multitone Analyzer: Relative Level** result uses the multitone method to provide a graphical display of the level vs. frequency (frequency response) for each channel of the DUT, relative to the measured level at a reference frequency. This enables you to specify a frequency that will be set as 0 dB, and to view the response in relation to that frequency. The rms value of each tone is measured, and a response curve is plotted through these values.

The **Skirt Width** control (page 381) in **Advanced Settings** can affect the values reported in this result.

Additional Controls for Relative Level

Ref Frequency

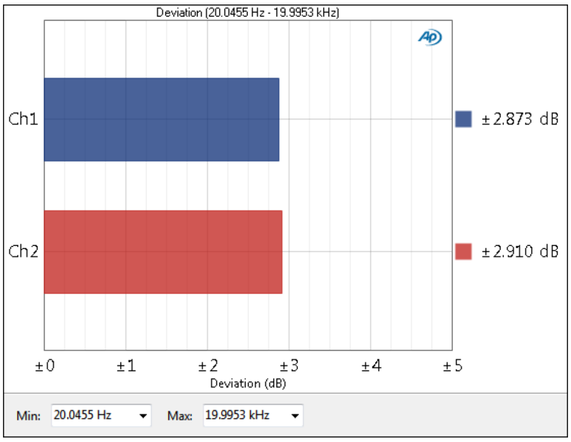
Enter a reference frequency here. The measured level at the frequency you choose is set as 0 dB on the graph, and the DUT response is plotted in relation to this reference. You can change the **Ref Frequency** at any time and the graph will immediately be redrawn to reflect the new setting. A midband frequency such as 1 kHz is often set as the reference frequency, and is the default here.

Units

Units available for **Multitone: Relative Level** are

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Multitone Analyzer: Deviation



The **Multitone Analyzer: Deviation** result uses the multitone method to provide the frequency deviation (the total range of frequency variation) of each DUT channel, displayed as a meter (single value) result. You can specify a minimum and maximum frequency to define the range to be considered in the deviation measurement.

The **Skirt Width** control (page 381) in **Advanced Settings** can affect the values reported in this result.

Additional Controls for Deviation

Min Frequency / Max Frequency

Deviation measurements are often limited to a certain frequency range; the default here is 20 Hz to 20 kHz. You can change set the range across which deviation is calculated by entering new values into the **Min Frequency** and **Max Frequency** fields. You can change this range at any time and the meter bars will immediately be redrawn to reflect the new settings.

A note about data sets and the Deviation result

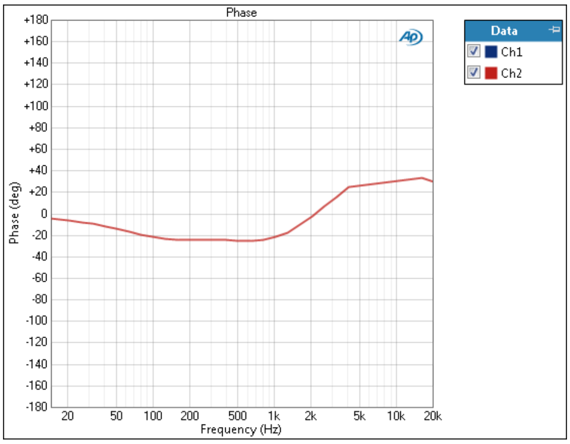
You can append or import graph data to Multitone measurements, but the Deviation result only shows the deviation in the latest acquired data. Other data sets cannot be selected for Deviation.

Units

Units available for **Multitone: Deviation** results are

- dB

Multitone Analyzer: Phase



The **Multitone Analyzer: Phase** result uses the multi-tone method to provide a graphical display of the inter-channel phase vs. frequency (interchannel phase response) for each channel of the DUT. The phase value of each tone is measured and compared to the phase value of the reference channel at the same tone, and a response curve is plotted through these values.

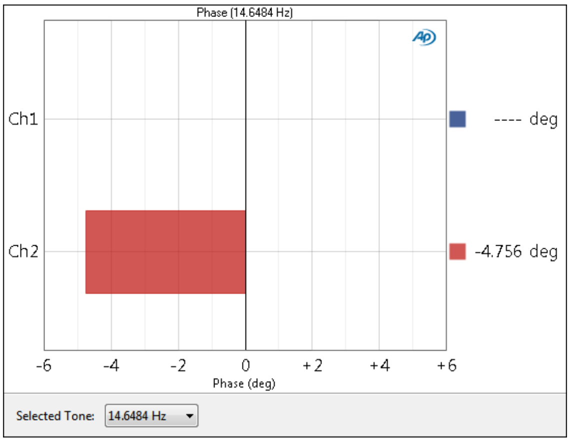
Read more about phase on page 300.

Units

Units available for **Multitone: Phase** are

X-axis	*Y-axis*
• Hz	• deg
• dHz	• rad
• F/R	
• %Hz	

Multitone Analyzer: Phase (at Tone)



The **Multitone Analyzer: Phase at (Tone)** result uses the multitone method to provide a graphical display of the interchannel phase at any tone in the multitone, for each channel of the DUT. The phase value of the selected tone is measured and compared to the phase value of that tone in the reference channel.

Read more about phase on page 300.

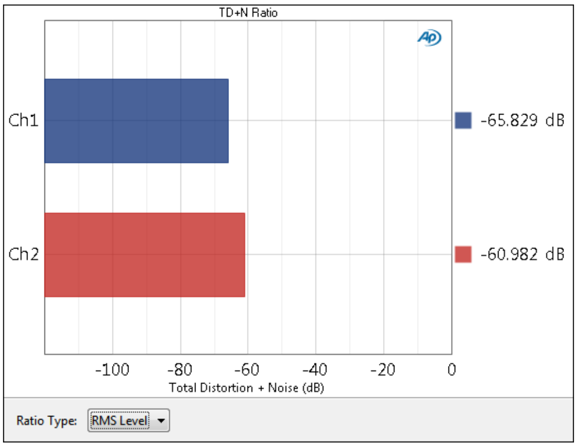
Units

Units available for **Multitone: Phase (at Tone)** are

- deg
- rad

See Chapter 98 for more information about units of measurement.

Multitone Analyzer: TD+N Ratio



Operation

See the Multitone Analyzer main topic on page 361 for controls and settings common to all multitone measurements.

The **Multitone Analyzer: TD+N Ratio** result uses the multitone method to provide total distortion + noise ratio results. How this is measured depends upon the setting of **Ratio Type.**

The **Skirt Width** control (page 381) in **Advanced Settings** can affect the values reported in this result.

TD+N Ratio (Ratio Type set to RMS Level)

This result is the ratio of two intermediate results.

First, the rms distortion + noise level (the sum of the rms distortion + noise across the multitone bandwidth) is determined for each DUT channel. Tone bins and bins in the current **Skirt width** are excluded from this result. All other bins, odd and even are summed. This provides a total distortion + noise (TD+N) level result. In a second operation, all the bins are summed to provide a total rms level result. The total rms level is divided by the TD+N level, to produce a TD+N ratio result.

TD+N Ratio (Ratio Type set to Max Level)

This result is the ratio of two intermediate results.

First, the rms distortion + noise level is determined in the same way as described above, providing a TD+N level result. In a second operation, the selected tone bin and the bins in that tone's current **Skirt width** are summed to provide an rms level result. This result is then scaled (the attenuation applied in the multitone signal creation process is factored out), providing a maximum level value. This maximum level value is divided by the TD+N level, producing an alternative TD+N ratio result.

Ratio Type:

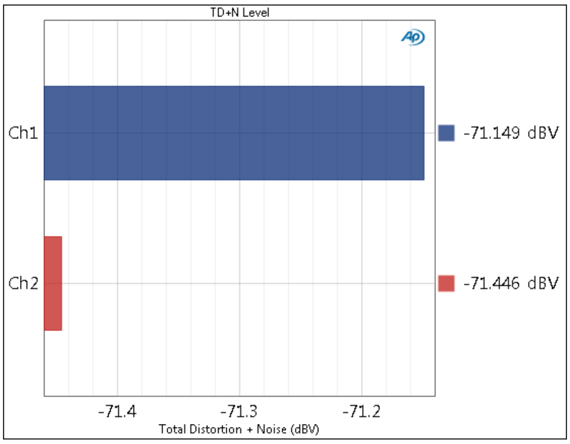
Choose the **Ratio Type** (described above) in this field.

Units

Units available for **Multitone: TD+N Ratio** results are:

- x/y
- %
- ppm
- dB

Multitone Analyzer: TD+N Level



The **Multitone Analyzer: TD+N Level** result uses the multitone method to provide the rms total distortion + noise level (the sum of the rms distortion + noise across the multitone bandwidth) for each DUT channel, displayed as a meter (single value) result. To produce a total distortion + noise result, tone bins and bins in the current **Skirt width** are excluded from measurement. All other bins, odd and even, are summed.

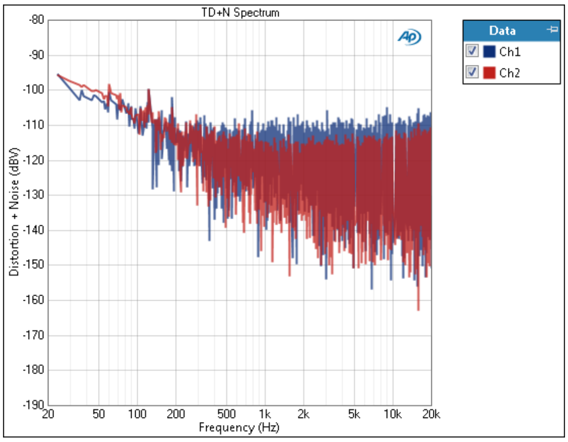
The **Skirt Width** control (page 381) in **Advanced Settings** can affect the values reported in this result.

Units

Units available for **Multitone: TD+N Level** are

Analog	*Digital*
• Vrms	• dBFS
• dBV	• FS
• dBu	• %FS
• dBrA	• dBrA
• dBrB	• dBrB
• dBSPL1	• dBSPL1
• dBSPL2	• dBSPL2
• dBm	
• W (watts)	

Multitone Analyzer: TD+N Spectrum



The Multitone Analyzer: TD+N Spectrum result provides an FFT spectrum display of the acquired signal. All bins, odd and even, that are not tone bins nor within the current **Skirt width** are plotted in this display. This removes the stimulus tones and reports a total distortion + noise result.

This result provides approximately the full number of measured data points (one for each non-tone bin in the FFT analysis). The graph data grid (page 573) initially shows the number of data points on the graph display, but can be set to show All Points.

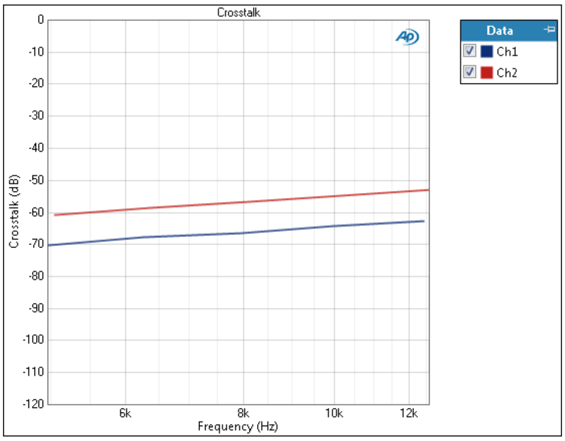
The **Skirt Width** control (page 381) in **Advanced Settings** can affect the values reported in this result.

Units

Units available for Multitone: TD+N Spectrum are

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz	• Vrms	• dBFS
• dHz	• dBV	• FS
• F/R	• dBu	• %FS
• %Hz	• dBrA	• dBrA
	• dBrB	• dBrB
	• dBSPL1	• dBSPL1
	• dBSPL2	• dBSPL2
	• dBm	
	• W (watts)	

Multitone Analyzer: Crosstalk



The **Multitone Analyzer: Crosstalk** result uses the multitone method to provide a graphical display of the crosstalk vs. frequency (crosstalk response) for each channel of the DUT. The crosstalk of each tone pair is measured, and a response curve is plotted through these values.

The **Crosstalk** results require that specific crosstalk tone exist in the multitone signal. If there are no such tones in the stimulus signal, Crosstalk results will not be available. See **Multitone Crosstalk Measurements** on page 375 for more information.

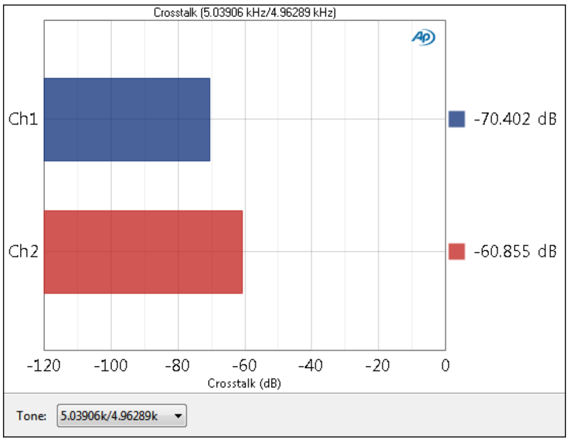
The **Skirt Width** control (page 381) in **Advanced Settings** can affect the values reported in this result.

Units

Units available for **Multitone: Crosstalk** are

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Multitone Analyzer: Crosstalk (at Tone Pair)



The **Multitone Analyzer: Crosstalk (at Tone Pair)** result uses the multitone method to provide the crosstalk level for a selected tone pair for each DUT channel, displayed as a meter (single value) result.

The **Crosstalk** results require that specific crosstalk tone exist in the multitone signal. If there are no such tones in the stimulus signal, Crosstalk results will not be available. See **Multitone Crosstalk Measurements** on page 375 for more information.

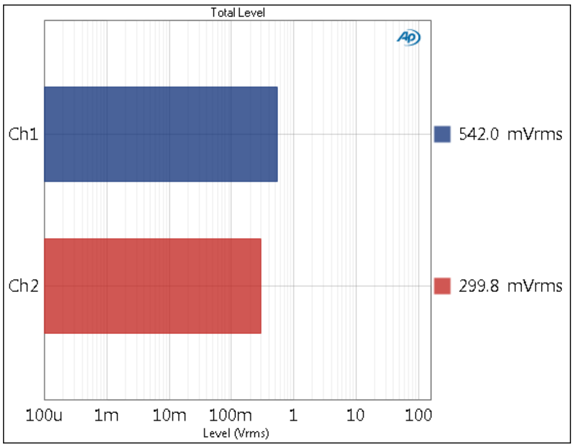
The **Skirt Width** control (page 381) in **Advanced Settings** can affect the values reported in this result.

Units

Units available for **Multitone: Crosstalk (at Tone Pair)** are:

- x/y
- %
- ppm
- dB

Multitone Analyzer: Total Level



The **Multitone Analyzer: Total Level** result uses the multitone method to provide the total rms level (the sum of the rms levels of all tones and their skirts) for each DUT channel, displayed as a meter (single value) result.

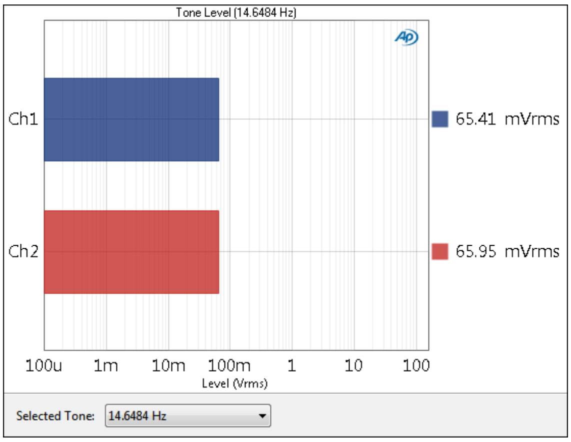
The **Skirt Width** control (page 381) in **Advanced Settings** can affect the values reported in this result.

Units

Units available for **Multitone: Total Level** are

Analog	*Digital*
• Vrms	• dBFS
• dBV	• FS
• dBu	• %FS
• dBrA	• dBrA
• dBrB	• dBrB
• dBSPL1	• dBSPL1
• dBSPL2	• dBSPL2
• dBm	
• W (watts)	

Multitone Analyzer: Tone Level



The **Multitone Analyzer: Tone Level** result uses the multitone method to provide the rms level for a selected tone for each DUT channel, displayed as a meter (single value) result.

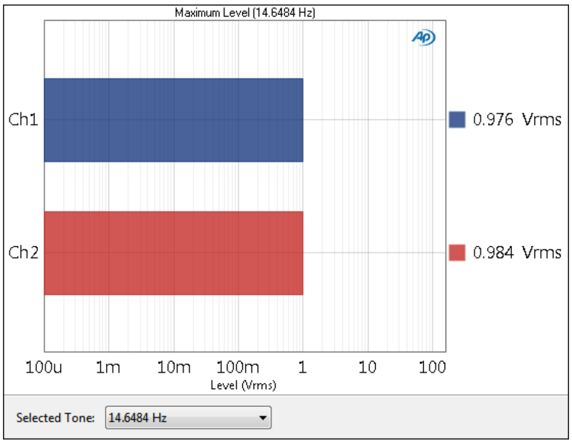
The **Skirt Width** control (page 381) in **Advanced Settings** can affect the values reported in this result.

Units

Units available for **Multitone: Tone Level** are

Analog	*Digital*
• Vrms	• dBFS
• dBV	• FS
• dBu	• %FS
• dBrA	• dBrA
• dBrB	• dBrB
• dBSPL1	• dBSPL1
• dBSPL2	• dBSPL2
• dBm	
• W (watts)	

Multitone Analyzer: Maximum Level (at Tone)



The **Multitone Analyzer: Maximum Level (at Tone)** result uses the multitone method to provide the calculated maximum level for a selected tone for each DUT channel, displayed as a meter (single value) result.

The **Skirt Width** control (page 381) in **Advanced Settings** can affect the values reported in this result.

Explanation of Max Level

Since a multitone waveform is a complex waveform, the sum of several or many individual tones, none of the individual tones can be at the maximum level of the waveform. In other words, the crest factor of the waveform will always be greater than the crest factor for a sine wave (1.414).

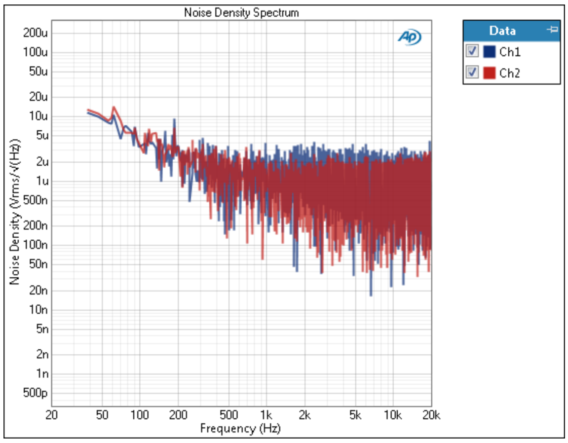
To provide a close estimation of a signal to noise ratio measurement made in the conventional method, the selected tone bin and the bins in that tone's current **Skirt width** are summed to provide an rms level result. This result is then scaled (the attenuation applied in the multitone signal creation process is factored out), providing a maximum level value.

Units

Units available for **Multitone: Maximum Level (at Tone)** are

Analog	*Digital*
• Vrms	• dBFS
• dBV	• FS
• dBu	• %FS
• dBrA	• dBrA
• dBrB	• dBrB
• dBSPL1	• dBSPL1
• dBSPL2	• dBSPL2
• dBm	
• W (watts)	

Multitone Analyzer: Noise Density Spectrum



The **Multitone Analyzer: Noise Density Spectrum** result provides an FFT spectrum display of the acquired signal. All odd-numbered bins that are not within the current Skirt width are plotted in this display; even numbered bins are not reported, ignoring stimulus tones and distortion products. This produces a noise-only result. This result is scaled for display in the accepted unit for noise density results: V/root(Hz) for analog inputs, or FS/root(Hz) for digital inputs.

> *This result provides approximately 1/2 the full number of measured data points (one for each odd-numbered bin in the FFT analysis). The graph data grid (page 573) initially shows the number of data points on the graph display, but can be set to show All Points.*

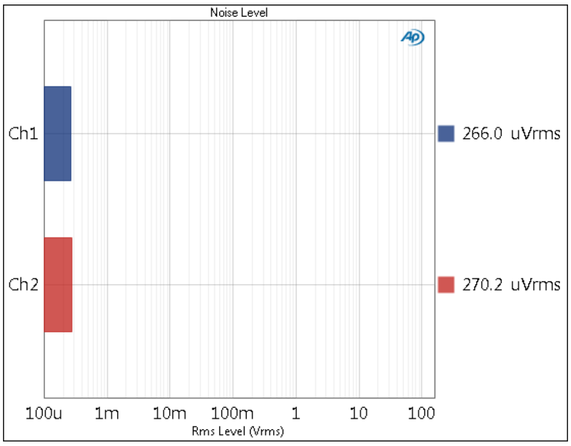
The **Skirt Width** control (page 381) in **Advanced Settings** can affect the values reported in this result.

Units

Units available for **Multitone: Noise Density Spectrum** are

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz	• V/root(Hz)	• FS/root(Hz)
• dHz		
• F/R		
• %Hz		

Multitone Analyzer: Noise Level



The **Multitone Analyzer: Noise Level** result uses the multitone method to provide the rms noise level (the sum of the rms noise across the multitone bandwidth) for each DUT channel, displayed as a meter (single value) result. To produce a noise-only result, even-numbered bins and bins in the current **Skirt width** are excluded from measurement. Instead, such bins are assigned the values in the adjacent odd-numbered bins. All bins are summed.

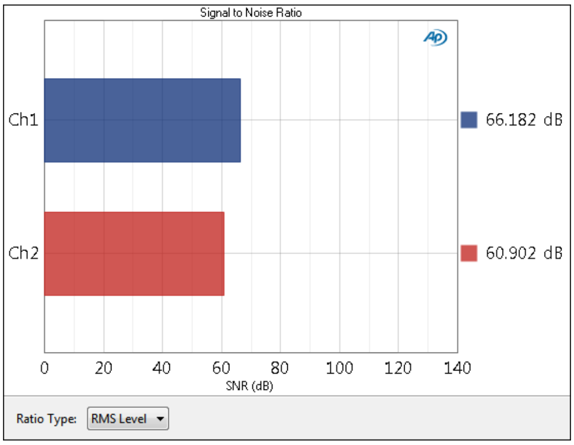
The **Skirt Width** control (page 381) in **Advanced Settings** can affect the values reported in this result.

Units

Units available for **Multitone: Noise Level** are

Analog	*Digital*
• Vrms	• dBFS
• dBV	• FS
• dBu	• %FS
• dBrA	• dBrA
• dBrB	• dBrB
• dBSPL1	• dBSPL1
• dBSPL2	• dBSPL2
• dBm	
• W (watts)	

Multitone Analyzer: Signal to Noise Ratio



Operation

See the Multitone Analyzer main topic on page 361 for controls and settings common to all multitone measurements.

The **Multitone Analyzer: Signal to Noise Ratio** result uses the multitone method to provide signal to noise ratio results. How this is measured depends upon the setting of **Ratio Type**.

The **Skirt Width** control (page 381) in **Advanced Settings** can affect the values reported in this result.

Signal to Noise Ratio (Ratio Type set to RMS Level)

This result is the ratio of two intermediate results.

First, the rms noise level (the sum of the rms noise across the multitone bandwidth) is measured for each DUT channel. To produce a noise-only result, even-numbered bins and bins in the current **Skirt Width** are excluded from measurement. Instead, the excluded bins are assigned the values in the adjacent odd-numbered bins. All bins are summed. This provides a total rms noise level result.

In a second operation, all the bins are summed to provide a total rms level result.

The total rms level is divided by the total noise level, to produce a signal to noise ratio result.

Signal to Noise Ratio (Ratio Type set to Max Level)

This result is the ratio of two intermediate results.

First, the rms noise level (the sum of the rms noise across the multitone bandwidth) is measured for each DUT channel. in the same way as described above, providing a total rms noise level result.

In a second operation, the selected tone bin and the bins in that tone's current **Skirt width** are summed to provide an rms level result. This result is then scaled the attenuation applied in the multitone signal creation process is factored out), providing a maximum level value.

This maximum level value is divided by the total noise, producing an alternative signal to noise ratio result.

See **Maximum Tone Level** on page 370 for an explanation of the calculations that provide maximum level.

Ratio Type

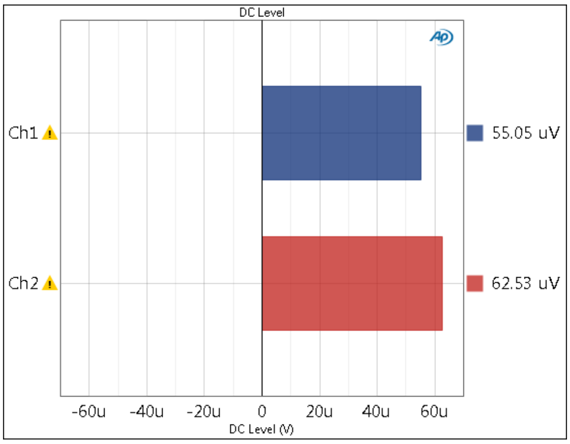
Choose the **Ratio Type** (described above) in this field.

Units

Units available for **Multitone: Signal to Noise Ratio** are:

- x/y
- dB

Multitone Analyzer: DC Level



The **Multitone Analyzer: DC Level** result uses the multitone method to measure dc level in the presence of signal for each DUT channel, displayed as a meter (single value) result. If **AC Coupled** is selected in the analog input configuration settings in Signal Path Setup, the Multitone Analyzer: DC Level result is not available. Change the setting to **DC Coupled** and start the measurement again.

The **Skirt Width** control (page 381) in **Advanced Settings** can affect the values reported in this result.

The analog measurement range for DC Level depends upon the instrument model:

- for an APx515, it is from approximately ±10 μV to ±120 V.
- for an APx52x or 555 family analyzer, it is from approximately ±10 μV to ±160 V.
- for an APx58x, it is from approximately ±10 μV to ±115 V.

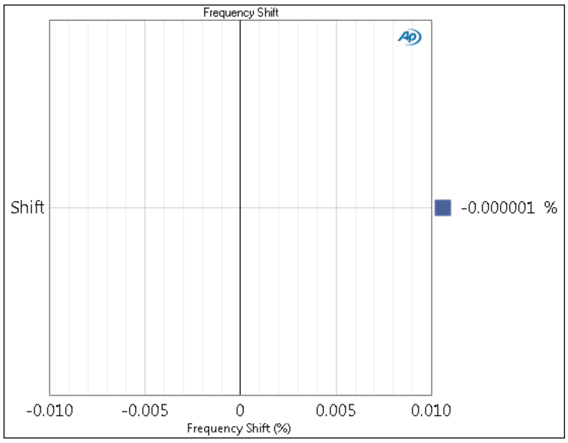
See More about DC Level on page 240.

Units

Units available for **Multitone: DC Level** are

Analog signals	***Digital signals***
• V	• D
	• hex

Multitone Analyzer: Frequency Shift



Operation

See the Multitone Analyzer main topic on page 361 for controls and settings common to all multitone measurements.

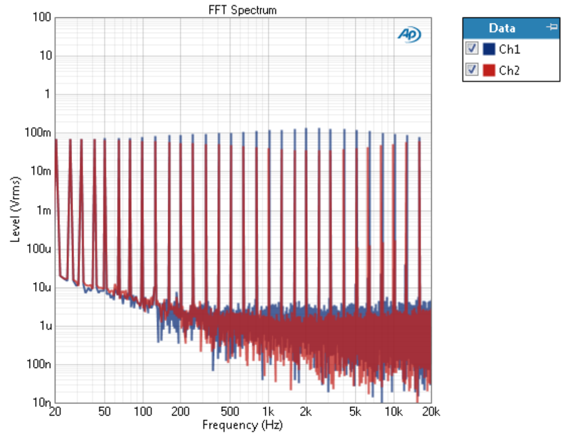
Multitone Analyzer: Frequency Shift is a diagnostic result that reports the degree of frequency shifting necessary to bring the signal acquired from the DUT into the precise match required for synchronous (windowless) FFT analysis.

Units

Units available for **Multitone: Frequency Shift** are:

- %
- ppm

Multitone Analyzer: FFT Spectrum



The **Multitone Analyzer: FFT Spectrum** result provides an FFT spectrum display (a frequency domain display) of the acquired signal. Every bin is plotted in this display. For a multitone stimulus, this will show the individual tones rising from a floor of distortion + noise.

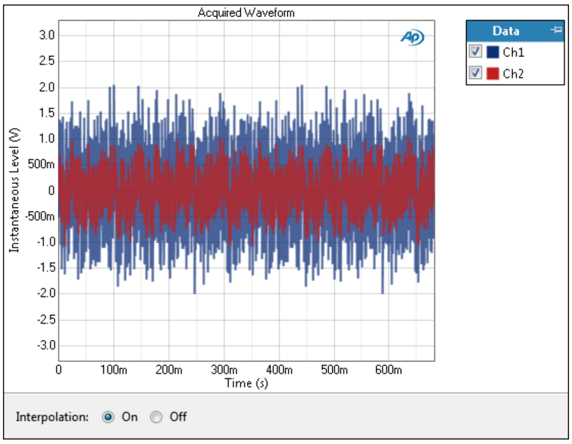
This result provides the full number of measured data points (one for each bin in the FFT analysis). The graph data grid (page 573) initially shows the number of data points on the graph display, but can be set to show All Points.

Units

Units available for **Multitone: FFT Spectrum** are

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz	• Vrms	• dBFS
• dHz	• dBV	• FS
• F/R	• dBu	• %FS
• %Hz	• dBrA	• dBrA
	• dBrB	• dBrB
	• dBSPL1	• dBSPL1
	• dBSPL2	• dBSPL2
	• dBm	
	• W (watts)	

Multitone Analyzer: Acquired Waveform



The **Multitone Analyzer: Acquired Waveform** result provides a time domain display of the acquired signal.

Units

Units available for **Multitone: Acquired Waveform** are

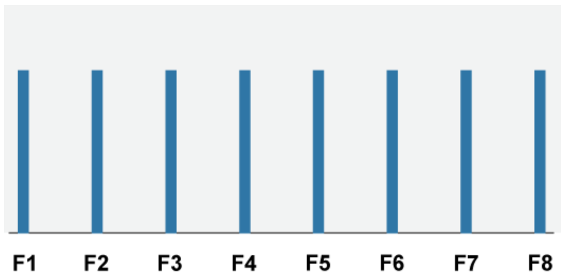
Analog signals	*Digital signals*
• V	• D
	• hex

More about multitone

Multitone testing uses FFT analysis with a special stimulus waveform. A multitone stimulus signal is a complex waveform, the combination of many sine waves. There are typically from 3 to 30 or more tones in a multitone signal; to the ear, it sounds like a dissonant organ chord. The analysis can provide frequency response, phase response, crosstalk response, distortion response, noise response and other results.

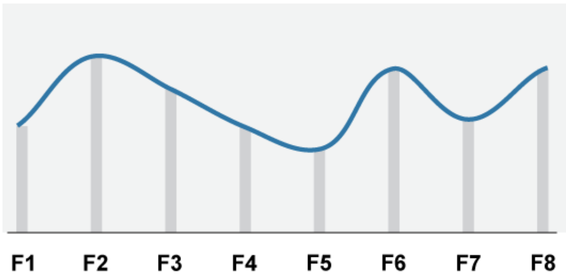
Multitones are useful for fast testing applications, and since a multitone stimulus can be brief and unobtrusive, multitone can be considered for in-service testing.

The example below shows a diagram representing a frequency domain view of an 8-tone multitone signal.



The multitone signal is applied to the DUT and an acquisition of the device's output is made for analysis.

The FFT data can be interpreted in a number of ways, extracting amplitude, phase, and other results. Multitone can provide detailed, accurate and fast measurements using a short multitone “burst.”



Multitone: Level displays a frequency response curve that plots the level across all of the tones at analysis. The amplitudes in the FFT bins that correspond to the multitone stimulus tones are shown as a light gray lines. The dark line represents the frequency response curve.

APx500 Multitone implementation

To produce useful results, the APx multitone analyzer must perform a windowless FFT on the acquired multitone audio signal. This requires that the analyzer be provided with the precise frequency values for each of the tones in the multitone stimulus signal. For multitones generated in the APx generator from a multitone signal definition, this information is available within the system. For playback of generator waveforms or for external source, multitone signal definition information must be made available to the analyzer.

FFT analysis

The multitone is analyzed in the frequency domain. The frequency and phase results of both tone bins and non-tone bins are used to derive the various measurement results.

Windowless FFT

Windowed FFTs filter the FFT record in the time domain using amplitude window functions to eliminate the effect of any discontinuity at the record ends. However, windowed FFTs provide limited frequency resolution.

Current technology has made it trivial to create stimulus signals with exactly the same sample rate and length as the FFT, permitting FFT analysis without using windows in most use cases. This method produces FFT results of very high frequency resolution and none of the distortions created by window leakage.

Bins, bandwidth and bin width

A frequency-domain FFT result contains amplitude and phase values in frequency bins. The number of bins is equal to one-half of the FFT record length: an FFT length of 8192 samples would produce a result with 4096 bins.

The bandwidth of the result is equal to one-half of the sample rate: an FFT record of a sample rate of 48 kHz would produce a result with a 24 kHz bandwidth.

The bin width is equal to the bandwidth divided by the number of bins: the examples above would produce a result with bin widths of 5.86 Hz. The amplitude and phase value of each bin would be the sum of the amplitudes and phases all frequencies that fall into that bin.

As another example, an FFT record of a length of 48000 samples and a sample rate of 48 kHz would have 24000 bins across a bandwidth of 24 kHz. Each bin would be 1 Hz wide.

Multitone and even and odd bins

Tones in an Audio Precision multitone stimulus signal are always set on even-numbered bins. This has the desirable effect of causing any distortion products to fall into even bins as well. At analysis time, all bins can be measured, or only tone bins, or only non-tone bins, or only odd bins. This enables the analysis software to report level (tone bins), distortion plus noise (all non-tone bins), noise (all odd bins) and so on. When reporting only the odd bins, you would correctly expect that the noise would be about half of the value of the true noise, as only half the bins are used. APx500 solves this by reporting values for all bins, using the value of the adjacent odd bin for each ignored even bin.

Skirts

A DSP generated and analyzer multitone is very precise, and a view of the results in loopback reveals zero stimulus energy outside of any FFT bin. But real world DUTs can exhibit a variable frequency shift, commonly jitter or flutter. Even when the multitone signal is properly frequency-shifted to match the bin centers, jitter or flutter will spread stimulus tone energy around the tone frequency, creating skirts surrounding the FFT bin(s) in analysis.

For multitone results based on rms level results, APx500 includes the energy in the skirts by including adjacent bins. For multitone results based on noise results, APx500 rejects the energy in the skirts by excluding adjacent bins. By default, the skirt width is 3.00 % of the tone frequency. For a 1 kHz tone, this would be a skirt width of 30 Hz.

Signal Definition

Multitone signal definition creation

A well-formed multitone waveform requires tones that are on specific frequency centers (for windowless FFTs, below) and with optimized phase relationships (for a low crest factor). These will vary with the chosen Sample Rate and Length. APx500 provides a comprehensive multitone creation capability within the Multitone Measurement environment. A multitone signal definition can be created, imported or extracted from an audio file. This definition can be loaded directly into the APx generator, or it can be exported as definition data or used to create a multitone WAV file.

Signal Generation

Output of an internally generated multitone (closed loop)

The APx generator can output a multitone from a multitone signal definition, or by playing a multitone audio file using APx500's generator waveform capability.

Output of a generator waveform multitone (from a WAV file)

When the APx generator is outputting a multitone by playing a generator waveform WAV file, a multitone signal definition is extracted from the audio in the WAV file to provide the analyzer with the necessary frequency center information.

Output of a generator waveform multitone (from a coded audio file)

APx500 does not incorporate decoders to open or analyze coded audio files. Therefore, when the APx generator is outputting a multitone by playing a generator waveform coded audio file, a multitone signal definition cannot be extracted from the audio in the file. A multitone signal definition containing frequency center information that corresponds to the audio in the coded file must be provided.

Signal Analysis

Acquisition

The multitone is acquired and the frequencies of the tones are identified. These are compared to the frequencies in the current multitone signal definition. The acquired audio is sample rate converted (SRC) to shift the pitch of the acquired multitone until the frequencies exactly match the frequencies in the multitone signal definition. This is required for windowless FFT analysis.

External Source configuration (open loop)

When APx500 is making a multitone measurement in external source configuration, a multitone signal definition containing frequency center information that corresponds to the audio being played in the DUT must be provided.

Triggering

The analysis of the multitone does not begin until the trigger function is satisfied. Four trigger modes and a trigger delay allow you to optimize the acquisition and analysis.

Results

Level

FFT analysis provides level and phase results for each frequency bin. In APx500 multitone measurements, these data are used in a number of ways.

- The rms level of every bin is reported. This provides the FFT spectrum result.
- The rms level of every non-tone bin is reported, with skirt exclusion. This provides the TD+N (total distortion + noise) result, which is also used as a source to derive the TD+N Ratio and TD+N Level results.
- The rms level of each stimulus tone bin is reported, with skirt inclusion. From these are derived the Level, Gain, Relative Level, Deviation, Total RMS Level, RMS Level (at Tone), Maximum Level (at Tone) results, and the signal component of Signal to Noise Ratio.
- The rms level of each odd bin is reported, with skirt exclusion. From these are derived Noise RMS Level, Noise Density Spectrum and the noise component of Signal to Noise Ratio.
- The instantaneous voltage level in the zero (0) Hz bin is reported as DC Level.

Multitone Crosstalk Measurements

APx multitone signals that have not been specifically configured for crosstalk measurements are mono; that is, the same multitone signal is applied to all APx output channels. Crosstalk measurements, however, require that one or more tones exist in each channel that do not exist in another, so that the degree of crosstalk can be observed and measured.

APx allows you to modify a mono multitone signal definition to create two related multitones that contain crosstalk tones; that is, that have one or more tones unique to each signal. The resultant multitone signal definitions are called “A” and “B”.

In APx, you have the choice of measuring crosstalk at one specified frequency or tone, or at all tones. APx accomplishes this by adding a tone 0.5 % higher than the specified tone(s) and then deleting the original specified tone(s), for signal A; and by adding a tone 0.5 % lower than the specified tone(s) and then deleting the original specified tone(s), for signal B.

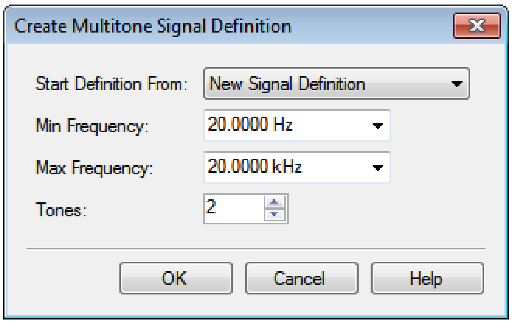
View Multitone Signal Definition

See **Edit Multitone Signal Definition** dialog on page 377 for detailed discussions of the fields in both the **View Multitone Signal Definition** and the **Edit Multitone Signal Definition** dialogs. The fields in these two dialogs and the descriptions of their content are the same.

The **View Multitone Signal Definition** dialog allows you to view a multitone signal definition that cannot be edited. You can optionally export the signal definition or create an audio file (WAV file) from the definition. Multitone signal definitions that cannot be edited include preset multitone signal definitions, signal definitions extracted from audio files, and signal definitions associated with External Source stimulus signals in open loop testing.

Create a Multitone Signal Definition

Also see **Edit Multitone Signal Definition** on page 377.



This dialog opens from **Signal Generation > Waveform > Create New...** on the Multitone Analyzer settings panel. The selections made here can be edited in more detail or changed entirely in the following dialog, **Edit Multitone Signal Definition** (page 377).

Start Definition From

Select an existing multitone definition from this list to use as a starting point for your new multitone.

New Signal Definition is the default, which provides a simple, 2-tone definition as a starting point. **Sample Rate** is set to 48 kHz for an analog signal path, or to the current sample rate for a digital signal path. **Length** is set to 8192 samples, **Min Frequency** to approximately 20 Hz and **Max Frequency** to approximately 20 kHz. The actual frequencies are constrained by the **Sample Rate** and **Length** defined in **New Signal Definition**. You can edit the Min **Frequency**, **Max Frequency** and **Tones** fields as described below.

Min Frequency

The minimum frequency in the definition selected in **Start Definition From** is shown here.

If **Start Definition From** is set to **New Signal Definition**, you can edit the frequency in this field to establish a new **Min Frequency** for the definition you are creating. Note that the available frequencies are constrained by the **Sample Rate** and **Length** defined in **New Signal Definition**.

Max Frequency

The maximum frequency in the definition selected in **Start Definition From** is shown here. If **Start Definition From** is set to **New Signal Definition**, you can edit the frequency in this field to establish a new **Max Frequency** for the definition you are creating. Note that available frequencies depend upon the **Sample Rate** and **Length** settings in the definition selected in **Start Definition From**.

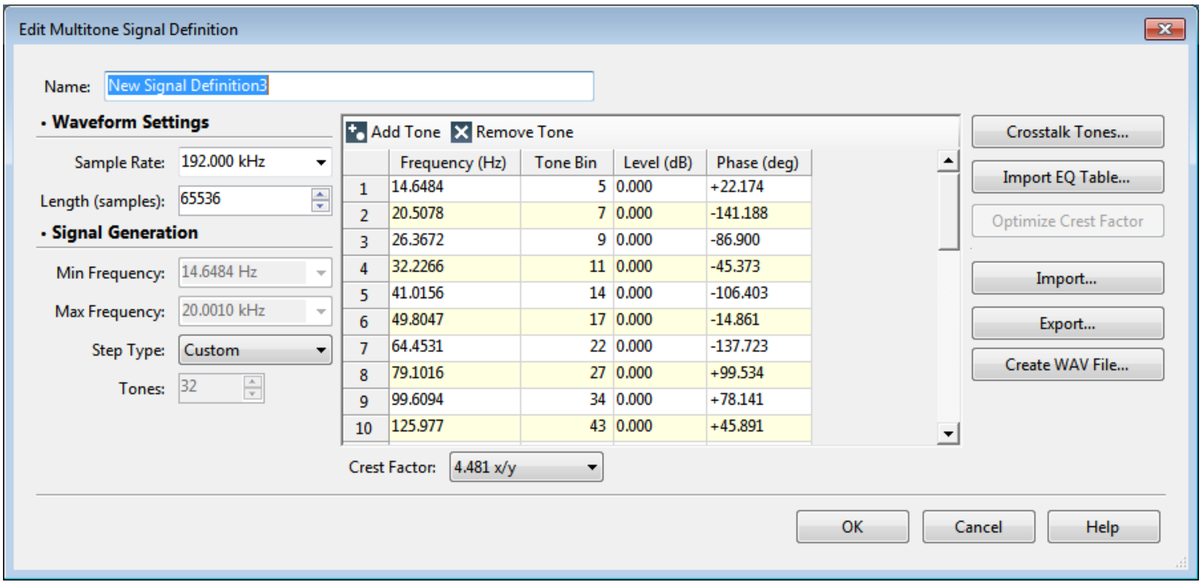
Tones

The number of tones in the definition selected in **Start Definition From** is shown here. If **Start Definition From** is set to **New Signal Definition**, you can edit this field to change the number of tones for the definition you are creating.

OK

When you have made your selections, click **OK** to open the next dialog, **Edit Multitone Signal Definition**.

Edit Multitone Signal Definition



This dialog allows you to edit a multitone signal definition. You can optionally import a signal definition, export the signal definition or create an audio file (WAV file) from the definition.

APx500 does not allow all multitone signal definitions to be edited.

Name

Enter a unique name for your multitone signal definition here.

Waveform Settings

Sample Rate

Enter the sample rate for your multitone here.

Length (samples)

Enter the length of the multitone signal here.

Signal Generation

Values entered in these fields will immediately be propagated into the tone description grid, to the right.

Min Frequency

Enter the minimum desired frequency for your multitone signal here.

Max Frequency

Enter the maximum desired frequency for your multitone signal here.

Step Type

The **Step Type** determines the spacing between the tones in the multitone signal. Choices are

- Logarithmic
  Logarithmic will logarithmically space the number of tones set in Tones across the frequency range set in Min and Max Frequency.
- Linear
  Linear will linearly space the number of tones set in Tones across the frequency range set in Min and Max Frequency. When the Step Type is Linear, the Step Size field (below) becomes available.
- Custom
  When any tone has been manually edited in the grid (added, deleted, or the frequency is changed), the Step Type becomes Custom, and the Min Frequency, Max Frequency and Tones fields become unavailable.

Tones

Enter the number of tones here.

Step Size

When the **Step Type** is **Linear**, the **Step Size** field becomes available. **Step Size** is equal to the range between **Min** and **Max Frequency** divided by the number of **Tones**. Editing the value of any of these three variables will change the values of the other two.

Tone Description Grid

The grid in the center of the dialog lists each tone in the multitone definition, sorted by frequency. When the information is available, the Tone Bin, Level and Phase are also listed.

Crest Factor

This field shows the crest factor (page 381) of the current multitone definition.

Crosstalk Tones...

When creating or editing a multitone definition, you can create tones on alternate channels that are slightly offset in frequency, for the purpose of measuring crosstalk. This button opens the Create Crosstalk Tones dialog. See Creating Multitone Crosstalk Tones on page 381 for a detailed discussion.

Import EQ Table...

See Generator Equalization on page 170 for general information on this subject.

You can equalize a multitone by entering various levels in the Level column in the Edit Multitone Signal Definition dialog. You can also import an existing generator equalization table file and apply it to the current multitone signal definition. Click Import EQ Table... and use the file browser to select a compatible generator EQ table in *.xls or *.csv format.

The level applied to tones that are between EQ points in the file will be set to an interpolated value.

Import EQ Table operations are additive. Reset all tone level values to 0 dB before re-importing to prevent addition.

Optimize Crest Factor

When you have chosen all the tones and optionally edited levels, added crosstalk tones, and applied an EQ curve, click Optimize Multitone. This control applies a set of algorithms to find phase settings for each tone that result in a multitone waveform with a lower crest factor (page 381). Optimization proceeds for 1 second, and uses the optimized waveform if it shows an improved crest factor. Subsequent applications of Optimize Multitone will re-optimize the waveform, but there is rarely any improvement over the optimization achieved in the first pass. Multitone waveforms with a great number of tones may show little or no improvement in crest factor after optimization.

Note: if a low crest factor (page 381) is important to your application, consider designing a multitone with fewer tones.

Import...

Click Import... to open a multitone signal definition from a file on disk. Use the file browser to select a CSV or XLS multitone signal definition file.

Export...

Your multitone signal definition is added to the current project as an attached project item. You can also export the signal definition as a CSV or XLS file. Click Export... to open a Save File browser.

Create WAV File....

You can use the current multitone signal definition to create a multitone audio file, in the mono or stereo linear PCM WAV format. Click Create WAV File... and enter the Duration (up to 5 minutes) and Bit Depth (either 16-bit or a 24-bit) in the dialog box. Click OK to open a file browser.

Multitone Crosstalk Measurements

APx multitone signals that have not been specifically configured for crosstalk measurements are mono; that is, the same multitone signal is applied to all APx output channels. Crosstalk measurements, however, require that one or more tones exist in each channel that do not exist in another, so that the degree of crosstalk can be observed and measured.

APx allows you to modify a mono multitone signal definition to create two related multitones that contain crosstalk tones; that is, that have one or more tones unique to each signal. The resultant multitone signal definitions are called “A” and “B”.

In APx, you have the choice of measuring crosstalk at one specified frequency or tone, or at all tones. APx accomplishes this by adding a tone 0.5 % higher than the specified tone(s) and then deleting the original specified tone(s), for signal A; and by adding a tone 0.5 % lower than the specified tone(s) and then deleting the original specified tone(s), for signal B.

This results in one or more crosstalk “tone pairs” between the two signals, offset slightly around the original tone centers.

Operation

The first task is to create a multitone with crosstalk tones, as described above. This is done in the Edit Multitone Signal Definition dialog. First, design the multitone definition with tones, sample rate, etc. according to your testing requirements.

Then click Crosstalk Tones... to open the Create Crosstalk Tones dialog and specify the tones you want. Close the dialog.

Observe that there are now an “A” and a “B” tab for the signal definition grid, each listing the tones in the two multitones. Optimize the crest factor and click OK to save the definition in the project.

Results

Two multitone crosstalk results are available.

Crosstalk result

The **Crosstalk** result requires at least two crosstalk tone pairs in the stimulus multitone, and provides a crosstalk ratio vs. frequency result. See Multitone Analyzer: Crosstalk (page 368), and the Multitone Analyzer main topic (page 361) for more detailed operational information.

Crosstalk (at Tone Pair) result

The Crosstalk (at Tone Pair) result provides a crosstalk ratio value for the selected tone pair. See Multitone Analyzer: Crosstalk (at Tone Pair) (page 369) and the Multitone Analyzer main topic (page 361) for more detailed operational information.

Removing or Changing Crosstalk Tones

If you open a multitone signal definition that already has A/B crosstalk tone pairs, you will not be able to add more tone pairs. If you click **Crosstalk Tones...** from the **Edit Multitone Signal Definition** dialog (page 377), you will see that the only available option is **Remove Crosstalk Tones**. Accepting this option will delete the "B" signal definition, leaving a mono multitone definition that is not configured for crosstalk measurement.

Note that accepting this option does not return the specified crosstalk tones to their original frequencies, but instead keeps the "A" definition, in which the specified crosstalk tones were shifted up by 0.5 %.

Mapping the A and B multitones to APx output channels

The APx generator can output a maximum of two different signals simultaneously. When the output format supports more than two signals, these two signals can be arbitrarily mapped to the output channels using controls in Advanced Settings. The default mapping is the "A" multitone signal to odd channels, and the "B" multitone signal to even channels.

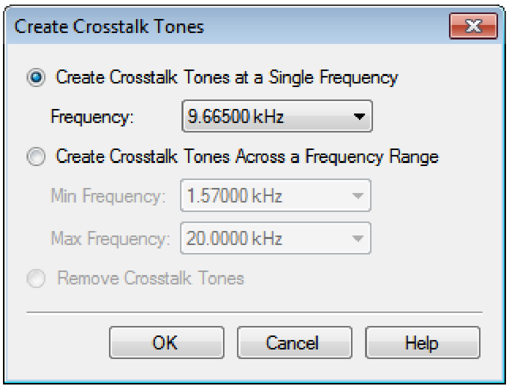
Creating Multitone Crosstalk Tones

In APx500, multitone definitions and waveforms that do NOT contain crosstalk tones are monaural, single-channel definitions or waveforms.

For crosstalk measurements, the APx500 multitone implementation requires a stereo, two-channel definition or waveform. For each frequency of interest, a pair of tones is created (one tone slightly above the frequency of interest, and one tone slightly below). The upper tone of the tone pair is placed in the "A" channel, and the lower in the "B" channel.

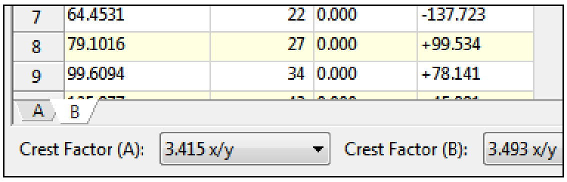
Create Crosstalk Tones dialog

In the **Edit Multitone Signal Definition** dialog (page 377), click **Crosstalk Tones...** The **Create Crosstalk Tones** dialog will open.



Create Crosstalk Tones at a Single Frequency

Use the first selection to create a pair of crosstalk tones at a single frequency of interest. Choose the frequency from the list of available tones in the current multitone signal definition. The single-channel multitone signal definition will be converted to a two-channel signal definition. (When you return to the previous dialog, you will notice that an "A" and a "B" tab have been added to bottom of the tone definition grid.) The tone you select will be removed from the definition, and a crosstalk tone pair will be created. The upper tone will be placed in the A channel, and the lower tone will be placed in the B channel.



Create Crosstalk Tones Across a Frequency Range

Use this selection to create a pair of crosstalk tones at every frequency within a range of interest. Set the range minimum and maximum frequencies using the **Min** and **Max** lists, which show the available tones in the current multitone signal definition. The single-channel multitone signal definition will be converted to a two-channel signal definition. (When you return to the previous dialog, you will notice that an "A" and a "B" tab have been added to bottom of the tone definition grid.} The tones you select will be removed from the definition, and a crosstalk tone pair will be created for each tone. The upper tones of each pair will be placed in the A channel, and the lower tones will be placed in the B channel.

“Not enough FFT bins” error

At low frequencies for short multitone FFT lengths, there will not enough FFT bins available to create a crosstalk tone pair. To create a successful multitone with crosstalk tones, either increase the **Length**, or specify crosstalk tones of higher frequencies.

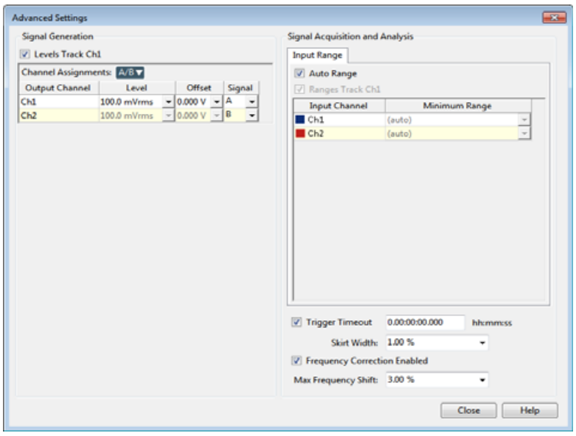
Remove Crosstalk Tones

If a multitone signal definition already has crosstalk tones on the A and B channels, new crosstalk tones cannot be added without first removing the existing crosstalk tones. Selecting **Remove Crosstalk Tones** deletes the B channel, resulting in a monaural, single-channel multitone signal definition.

Note that removing the existing crosstalk tones does NOT return the tones to their original frequencies, as were specified in the original multitone signal definition. That information is not carried forward in the definition. Instead, the tone(s) that had been shifted up slightly (the A channel crosstalk tones) become the tones in the new definition.

Advanced Settings for the Multitone Analyzer

The default settings here are appropriate for most measurements. You may want to make minor adjustments for special situations.



Also see More About Multitone and APx500 Multitone implementation on page 373.

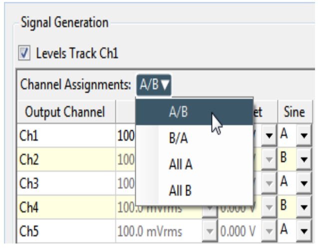
Signal Generation

If **Track first channel** level is checked (the default), the generator output level value for channel 1 is copied to the other channels, and output levels for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels.

To set individual generator output channel levels, **uncheck** the **Track first channel level** checkbox and enter values in the output channel **Level** fields.

Set Channel Assignments for crosstalk multitones

For multitones that include crosstalk tones, you can also set channel mapping. By default, crosstalk multitones map channel A audio to odd output channels, and channel B audio to even output channels. These assignments can be remapped by changing individual settings in the **Signal** column, or by selection one of several presets from the **Set Channel Assignments** menu.



Set Channel Assignments for generator waveforms

For stereo generator waveform multitone files, you can set channel mapping. By default, channel 1 audio is mapped to channel 1 output, and channel 2 to channel 2. These assignments can be remapped by changing individual settings in the **Wave Ch** column.

File Playback

For generator waveform file playback, you can view the length of the file, and you can adjust the playback start position. Select **Seconds** or **Samples** for the **Length** and **Start Position** units. Enter a new **Start Position** if desired.

Signal Acquisition and Analysis

Input Range tab

Digital inputs do not require ranging, and the Input Range tab is unavailable when inputs are set to a digital format.

If **Auto Range** is checked (the default), each analog channel determines its input range automatically, based on the level of the input signal. If the input signal level changes beyond a ranging threshold, **Auto Range** will cause the input ranging circuits to move up or down for proper ranging.

Read more about ranging in Chapter 92.

If **Auto Range** is unchecked, you can set a fixed range for each analog input channel. If **Track first channel range** is checked (the default), the fixed input range setting for channel 1 is copied to the other channels, and range settings for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels.

To set individual analog input channel ranges, uncheck the Track first channel range checkbox and enter values in the input channel Range fields.

Advanced Settings for Multitone does not have a Settling tab.

Trigger Timeout

When a Trigger selection other than Free Run (untriggered) is set and Trigger Timeout is not set, the APx500 software will wait indefinitely for a valid trigger condition. When Trigger Timeout is set, the operation will wait for the duration set in the Trigger Timeout field, and then will cancel the measurement with a notice. Minimum time is 0.1 s, maximum time is 7 days.

Enter the Trigger Timeout following this pattern:

d:hh:mm:ss.s

where d = days, hh = hours, mm = minutes, ss = seconds. Days and fractional seconds are optional.

Skirt Width

Flutter, jitter or other variable frequency shift in the device or transmission channel will create sidebands of energy around each of the tones, called skirts. Some small amount of energy is also spread into the skirts as an artifact of the multitone frequency correction. For correct rms level measurements, energy in the skirts must be added to the energy in the tone. For correct noise measurements, energy in the skirts must be excluded from the measurement.

The Skirt Width control specifies the width around the tone that is to be considered as skirt energy. The default (and maximum) is 3 %.

Frequency Correction Enabled

In practical devices and transmission channels, audio signals can undergo frequency changes (sample rate changes). APx multitone measurements, which use unwindowed FFTs for analysis, must frequency shift the acquired waveform so that it matches the internal reference exactly. This checkbox enables frequency correction.

By default, Frequency Correction is checked (on) for digital inputs. The multitone measurement will run faster with Frequency Correction disabled.

Max Frequency Shift

When Frequency Correction is enabled, the Max Frequency Shift field is available.

The value in Max Frequency Shift represents the maximum percentage of shift allowed. If the input signal requires a shift beyond this value, the measurement will fail. The default value is 3 %; the range is 0 % to 5 %.

Crest Factor

The crest factor of a waveform is the ratio of its peak value to its rms value. Crest factor can be expressed as a ratio or fraction (4:1 or 4/1, for example) or as a decibel (12.041 dB, in this case). When expressed as a ratio or fraction, crest factor is often reduced to lowest terms (4, in this case).

In APx500, the amplitude of a multitone waveform is set to the value in the generator Level setting field. The higher the waveform's crest factor, the lower the level of the component tones within the waveform.

Typical crest factors:

WAVEFORM	RATIO	DECIBEL
square wave	1	0 dB
sine wave	1.414	3.01 dB
triangle wave	1.732	4.77 dB
ambient noise (typical)	3	10 dB
speech (typical)	4	12 dB
APx 32-tone multitone	4.481	13.028 dB
some musical instruments (typical)	8–10	18–20 dB
fast transient (such as a gunshot)	30 or more	30 dB or more

Recording Multitone Audio to a File

This topic discusses the Save to File feature when invoked from the Multitone Analyzer. See Recording Audio to a File on page 347 for a discussion of the Save to File feature when invoked from the Measurement Recorder.

Overview

The Measurement Recorder and the Noise Recorder can optionally record acquired audio to a file on disk.

Transform record

As explained in APx500 Multitone Implementation on page 374, at analysis time a multitone signal must match the multitone signal definition in sample length and in sample rate. This often requires frequency (sample rate) shifting and truncation of the raw input acquisition. It is this truncated and frequency-shifted audio, the audio actually used for analysis, that is recorded to file; not the raw audio acquisition. This audio is also be referred to by the terms *transform record* or *FFT record*.

Note: APx timing constraints will sometimes cause a timeout error when saving to a network folder or a removable disk. We recommend saving audio files to a local disk.

On the Measurement or Noise Recorder panel, check the Save to File checkbox. Click File Settings to specify the file name, location and file format.

Save to File is not available when Input Configuration is set to File (Analog Units) or File (Digital Units).

Save Acquisition to File Settings dialog

Location

Enter or browse to the destination folder here.

Name

Name the output file here. Choose from one of these selections:

- If file exists, replace it
- If file exists, append timestamp to the file name

Format

Choose the file format in the Format field.

For both analog inputs and digital acquisitions, there are three linear file formats available:

Multiple Mono PCM (.wav)

If Multiple Mono PCM (.wav) (the default) is selected, each audio channel of the acquisition is saved as an individual monaural WAV file, using the Microsoft WAVE_FORMAT_PCM (Type 1, 24 bit) file format. For multichannel audio of channel count n, n mono WAV files are created, with the numeral n appended to the file name.

Multichannel PCM (.wav)

If Multichannel PCM (.wav) is selected, all the audio channels of the acquisition are saved in one multichannel file, using the WAVE_FORMAT_PCM format. For 1 channel and 2 channel acquisitions, this selection records a standard WAVE_FORMAT_PCM (Type 1, 24 bit). For multichannel acquisitions, this selection records a multichannel WAVE_FORMAT_PCM file in a proprietary 24-bit format, compatible with Sony Sound Forge and other audio editing applications.

Extensible Multichannel PCM (.wav)

If Save as WAVE_FORMAT_EXTENSIBLE is selected, all the audio channels of the acquisition are saved in one multichannel file, using the Microsoft standard WAVE_FORMAT_EXTENSIBLE format, at 24 bits.

Bit Depth

This field is only available for analog acquisitions. The default Bit Depth (word length) is 24 bits. You can also select 16 bits.

Analog scaling and sample rates

When recording an analog acquisition, proprietary scaling metadata is embedded in the WAV file. The scaling factor relates the analog voltage to the digital level (Vrms/FS), and is used for scaling the display of values when using the APx500 file input configuration.

For analog acquisitions, the sample rate is set to be the same as the sample rate determined by the analyzer input configuration analog bandwidth setting, set in Signal Path Setup, as follows:

Bandwidth	Sample Rate
22 kHz	48 kHz
45 kHz	96 kHz
90 kHz	192 kHz
250 kHz	624 kHz
500 kHz	1.248 MHz
1 MHz	2.496 MHz

Input bandwidths beyond 90 kHz require the BW52 Option; 1 MHz bandwidth requires input set to 1 channel.

The sample rate can be changed in the Advanced Settings dialog.

Duration for linear recordings

Windows WAV files are limited to 4 GB. For 24-bit stereo WAV files, this is approximately 1 hour at 192 kHz. Mono files and files with lower sample rates or bit depths can have longer durations. For longer recordings, APx automatically creates a sequential file (or a set of sequential files) providing a very long recording capability without loss of any data.

Note: The sequential file feature for very long recordings is resource-intensive, and requires a high performance level in terms of PC CPU and hard disk speed, beyond the minimum recommended computer required to run APx500.

61

Noise (RMS)
(Sequence Mode)

The Noise (RMS) measurement provides a single-value result that shows the noise present at the output of each DUT channel, using an rms detector. Typically, the Noise measurement is used without a stimulus signal (**Waveform: None**), but a Noise signal or Generator Waveform files are also available.

The Noise (RMS) measurement available in APx500 is:

- Noise: RMS Level

Preparing for a Noise Measurement

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 6 for information about Signal Path Setup.

To measure the residual noise in a DUT, select **None** as the waveform, which will run the measurement with no stimulus signal. We recommend disconnecting any devices from the DUT inputs (including the APx generator, if connected), and terminating the DUT inputs with a low resistance, typically 0 Ω to a few hundred ohms.

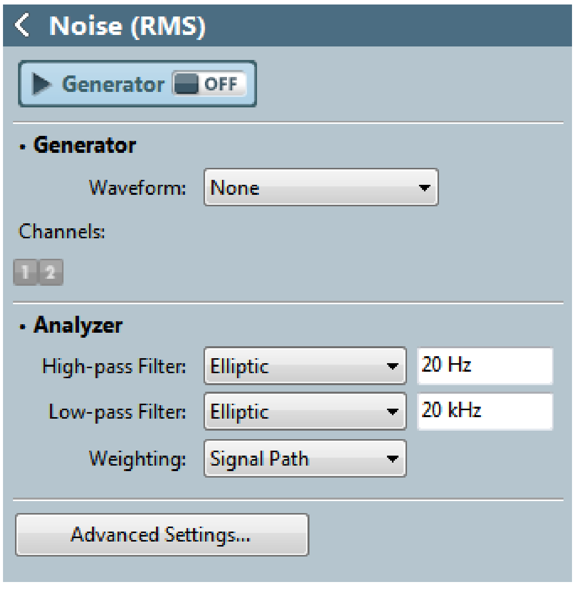
To test using a noise stimulus, select **Noise** or **Browse for File**, and connect the generator outputs to the DUT inputs.

Generator

If you choose to make a noise measurement with a stimulus signal, these signals are available. When **Waveform** is set to **Noise**, these choices will be available from the **Noise Shape** selector:

- White
- Pink
- IEC 60268-1
- BS EN 50332-1

See **More About Noise Signals** on page 385 for descriptions of these signals.



Alternatively, you can select a generator waveform file as the test signal, using the file browser in the Waveform drop-down menu.

See Chapter 5 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

Level

Set the generator level here. Note that with Noise signals the generator level values are scaled differently; see **More About Noise Signals** on page 385.

Analyzer

This measurement can be configured to measure jitter in Signal Path Setup > Input > Measure. Read detailed information about jitter generation and measurement beginning on page 60.

Filters

Noise measurements are typically made within a limited passband. The default setting for noise level measurements uses a 20 kHz low-pass filter and a 20 Hz high-pass filter.

Local high-pass, low-pass and weighting filters are available for this measurement. The low-pass and high-pass filters, if selected, are applied to the entire measured signal and affect all the results. The weighting filter, if selected, affects only the distortion and distortion plus noise results. See Measurement Filters on page 547 for detailed information about the filters available locally for some measurements.

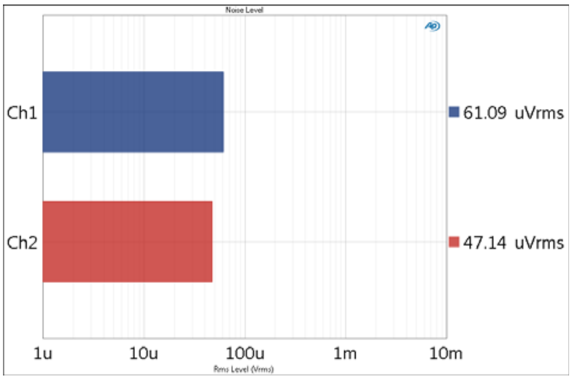
Low pass, high pass and weighting Signal Path Filters settings (see page 545) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass or high-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active. A local weighting filter setting does not override any Signal Path filter settings; the local weighting filter is applied in addition to any active Signal Path filters.

Advanced Settings

Advanced Settings makes available individual channel output level and offset controls, and input ranging and settling controls. See Advanced Settings, next.

See Chapter 98 for more information about units of measurement.

Noise (RMS) Level



Noise (RMS) Level provides a single-value meter result, displaying the rms noise level from each DUT channel, as measured at each analyzer input.

Units

Units available for Noise (RMS) Level are

Analog Signals

- Vrms
- dBV
- dBu
- dBrA
- dBrB
- dBSPL1
- dBSPL2
- dBm
- W (watts)

Digital Signals

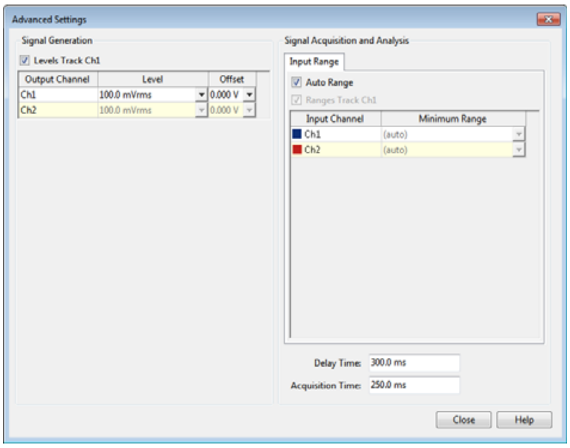
- dBFS
- FS
- %FS
- dBrA
- dBrB
- dBSPL1
- dBSPL2

Jitter signals

- UI
- dBUI
- s

Advanced Settings for Noise Measurement

The default settings here are appropriate for most measurements. You may want to make minor adjustments for special situations.



Signal Generation

If Waveform is set to None, no signal generation settings will be available.

If Waveform is set to Noise, the Output Channel level grid will be available.

If Track first channel level is checked (the default), the generator output level value for channel 1 is copied to the other channel(s), and output levels for channel(s) greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channel(s).

To set individual generator output channel levels, uncheck the “Track first channel level” checkbox and enter values in the output channel Level field(s).

Signal Acquisition and Analysis

Input Range tab

Digital inputs do not require ranging, and the Input Range tab is unavailable when inputs are set to a digital format.

If **Auto Range** is checked (the default), each analog channel determines its input range automatically, based on the level of the input signal. If the input signal level changes beyond a ranging threshold, **Auto Range** will cause the input ranging circuits to move up or down for proper ranging.

See page 551 for more information about ranging and autoranging.

If **Auto Range** is unchecked, you can set a fixed range for each analog input channel. If **Track first channel range** is checked (the default), the fixed input range setting for channel 1 is copied to the other channels, and range settings for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels.

To set individual analog input channel ranges, uncheck the **Track first channel range** checkbox and enter values in the input channel Range fields.

Delay Time

When the noise measurement is run from the Navigator or as part of a sequence, the acquisition is delayed for the time set here. This pause can avoid signal disruptions that may occur in the DUT when changing measurements. The Delay Time range is 0.0 s to 10.0 s; the default setting is 300.0 ms.

Acquisition Time

Since noise is aperiodic, a very short acquisition may display a result that is well above or below the average noise level. In the Noise: RMS Level measurement the signal is acquired over a period of time between 100 ms and 10 s, set in Acquisition Time. Longer acquisitions provide more repeatable results, but update more slowly. The default acquisition time is 250 ms.

More About Noise Signals

The Noise signals in APx500 are generated in DSP. They have the following characteristics:

White noise

White noise has a power spectrum ideally constant per unit bandwidth from just above DC to half the sampling rate. The crest factor is between 3 and 5, and the probability distribution is close to Gaussian. The generation period is 2^{32-1} samples long, which is about a day at 48 kHz.

Pink noise

Pink noise is generated from white noise using a filter, and inherits the statistics of the white noise. Pink noise has a power spectrum ideally constant per fractional bandwidth from just above DC to half the sampling rate. The signal power drops off below 10 Hz. The ideal pink power spectrum is maintained ±1.0 dB in the range 2×10^{-4} SR to 0.45 SR.

IEC 60268-1 noise

Noise according to the IEC 60268-1 standard is generated from pink noise passed through a weighting filter. IEC 60268-1 has a spectrum that mimics program material, including voice and music.

BS EN 50332-1 noise

BS EN 50332-1 is IEC 60268-1 noise with a reduced crest factor, accomplished by applying soft clipping. The crest factor is between 1.8 and 2.2. BS EN 50332-1 noise supports headphone testing.

Setting Generator levels when using noise signals

The RMS units assume a sine wave, which has a crest factor of the square root of 2 (approximately 1.414). None of the noise signals have this crest factor, so actual noise values will not be equal to the generator level in RMS-based units.

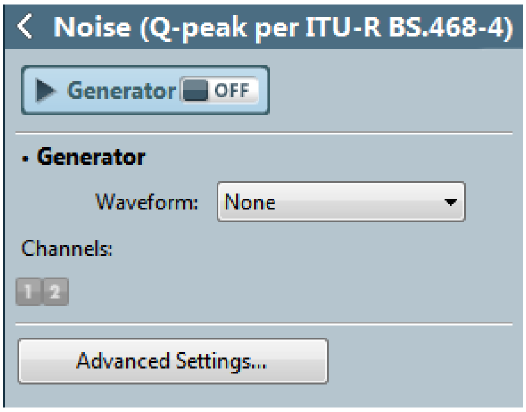
White noise is generated using a method that guarantees that the peak value is equal to the generator level in peak units; Vp and Vpp will accurately specify the noise level. Note that for digital outputs there is no generator level peak unit.

The filters used in pink noise, IEC 60268-1 noise and BS EN 50332-1 noise alter the signal so that the maximum output cannot be guaranteed in the same way that the white noise can. Signal levels for these noise types will not be equal to the generator level setting for any units, but will be bounded by the Vp setting.

62

Noise (Q-peak)
(Sequence Mode)

Noise (Q-Peak) Level view provides a single-value measurement of the noise level present at the output of each DUT channel, using a Q-peak detector and the CCIR/ITU-R BS 468-4 weighting curve. Use this measurement if you are required to provide CCIR/ITU-R BS 468-4 results.



Preparing for a Noise Measurement

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 6 for information about Signal Path Setup.

To measure the residual noise in a DUT, select **None** as the waveform, which will run the measurement with no stimulus signal. We recommend disconnecting any devices from the DUT inputs (including the APx generator, if connected), and terminating the DUT inputs with a low resistance, typically 0 ohms to a few hundred ohms.

To test using a noise stimulus, select **Noise** or **Browse for File**, and connect the generator outputs to the DUT inputs.

Generator

Typically, the Noise (Q-Peak) measurement is used without a stimulus signal (**Waveform: None**), but a Noise signal or Generator Waveform files are also available.

If you choose to make a noise measurement with a stimulus signal, these signals are available. When **Waveform** is set to **Noise**, these choices will be available from the **Noise Shape** selector:

- White
- Pink
- IEC 60268-1
- BS EN 50332-1

See **More About Noise Signals** on page 385 for descriptions of these signals.

Alternatively, you can select a generator waveform file as the test signal, using the file browser in the Waveform drop-down menu.

See Chapter 5 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

Level

Set the generator level here. Note that with Noise signals the generator level values are scaled differently; see **More About Noise Signals** on page 385.

Analyzer

The average Q-peak level across the default **Acquisition Time** of 250 ms is shown on the meter bar display. To adjust the **Acquisition Time** or the **Delay Time** (which sets the time of the beginning of the acquisition), go to **Advanced Settings**.

This measurement can be configured to measure jitter in Signal Path Setup > Input > Measure. Read detailed information about jitter generation and measurement beginning on page 60.

Filtering

A fixed CCIR/ITU-R BS 468-4 weighting curve filter is used for Noise (Q-peak) measurements.

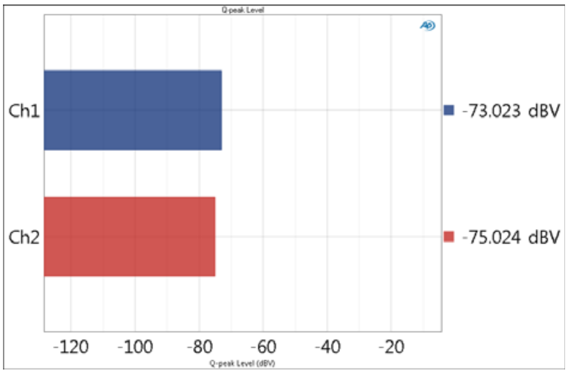
Low pass, high pass and weighting input filter settings made in Signal Path Setup > Input/Output > Filters will also affect this measurement. See Chapter 91 for more information about filtering.

Advanced Settings

Advanced Settings makes available individual channel output level and offset controls, and input ranging and settling controls. See Advanced Settings on page 384.

See Chapter 98 for more information about units of measurement.

Q-Peak Level



Noise (Q-Peak) Level provides a single-value meter result, displaying the Q-Peak noise level from each DUT channel, as measured at each analyzer input.

Units

Units available for Noise (Q-peak) are

Analog Signals

- Vrms
- dBV
- dBu
- dBrA
- dBrB
- dBSPL1
- dBSPL2
- dBm
- W (watts)

Digital Signals

- dBFS
- FS
- %FS
- dBrA
- dBrB
- dBSPL1
- dBSPL2

63

Noise Recorder (RMS)
(Sequence Mode)

Introduction

For the APx515, this measurement requires a software option key. See page 166 for more information about software options.

Noise Recorder (RMS) is a diagnostic tool that provides a record of an RMS noise measurement versus elapsed time. The record can be very long (up to one week), providing a means of monitoring the output of a device under test over an extended period of time. Typically, the Noise Recorder is used without a stimulus signal (Waveform: None), but a Noise signal or Generator Waveform files are also available.

The Noise Recorder result available in APx500 is:

- RMS Level

Recording an audio file to disk

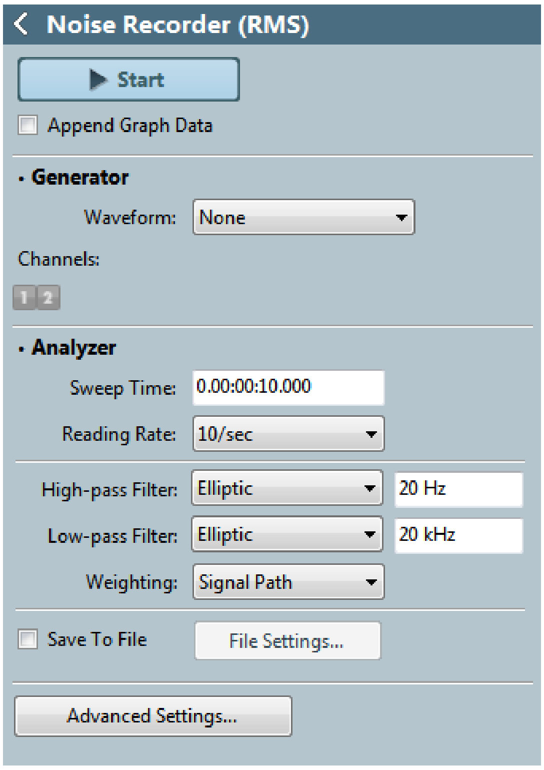
The Noise Recorder also provides a means of recording the audio acquisition to a file. See Recording Audio to a File on page 347 for more information.

Preparing for a Noise Recorder measurement

If you have not yet set up your test, first go to Signal Path Setup (Chapter 6).

To measure the residual noise in a DUT, select None as the waveform, which will run the measurement with no stimulus signal. We recommend disconnecting any devices from the DUT inputs (including the APx generator, if connected), and terminating the DUT inputs with a low resistance, typically 0 ohms to a few hundred ohms.

To test using a noise stimulus, select Noise or Browse for File, and connect the generator outputs to the DUT inputs.



Operation

When Waveform is set to None, no stimulus is generated. Click Start and the RMS level of the noise in the DUT output is recorded.

When Waveform (below) is set to Noise or File, click Start. The generator will output the stimulus signal to the DUT on the selected generator channels at the level set in the Signal Generation panel. The RMS level of the noise in the DUT output is recorded.

Append Graph Data

Normally, the current graph data for all results is deleted each time you start a new measurement. If the Append box is checked, the current data is kept in memory as a Data Set (see page 165), and the new measurement data is appended as a new data set. You can append many Data Sets.

Generator

Typically, noise measurements are made with no stimulus signal. Once connections to the DUT outputs are made, the noise level is measured and displayed.

However, signal generation is available.

When Waveform is set to Noise, these choices will be available from the Noise Shape selector:

- White
- Pink
- IEC 60268-1
- BS EN 50332-1

See More About Noise Signals on page 385 for descriptions of these signals.

Alternatively, you can select a generator waveform file as the test signal, using the file browser in the Waveform drop-down menu. See Generator Waveforms on page 163.

Level

Set the generator level here. Note that with Noise signals the generator level value are scaled differently; see More About Noise Signals on page 385.

Analyzer

Sweep Time

Sweep Time sets the length of the measurement record. Minimum time is 0.1 s, maximum time is 7 days.

Enter the Sweep Time following this pattern:

d:hh:mm:ss.s

where d = days, hh = hours, mm = minutes, ss = seconds. Days and fractional seconds are optional.

Reading Rate

Reading Rate sets a maximum number of times the input signal will be read per second. In most cases the number of readings will be very close to this value. However, the rate is dependent upon PC performance and channel count, and may be lower. See More About Reading Rate on page 349.

Filters

Low-pass, weighting and high-pass filters are available for this measurement. In the case of the Noise Recorder, all three filters (if applied) act on the RMS Level result.

Noise measurements are typically made within a limited passband. The default setting for noise level measurements uses a 20 kHz low-pass filter and a 20 Hz high-pass filter.

Low pass, high pass and weighting Signal Path Filters settings (see page 545) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass or high-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active. A local weighting filter setting does not override any Signal Path filter settings; the local weighting filter is applied in addition to any active Signal Path filters.

Save to File

Click this option to save the acquired audio to a file on disk. See Recording Audio to a File on page 347.

Running the measurement in File Input configuration

See Analyzing Audio Files in Chapter 19 for more information about File Input configuration.

If you have chosen File for an input configuration in Signal Path Setup, first click the File List... button to add input file(s) to a list for analysis.

If your DUT is to be recording the WAV file, select a generator waveform and turn the generator ON, as described above. Configure your DUT to save a WAV file(s) whose file name(s) and location match the name specified in the File List.

If you are measuring an existing file (not recording from the generator but working in External Source), be sure that the file name(s) and location match the name(s) specified in the File List.

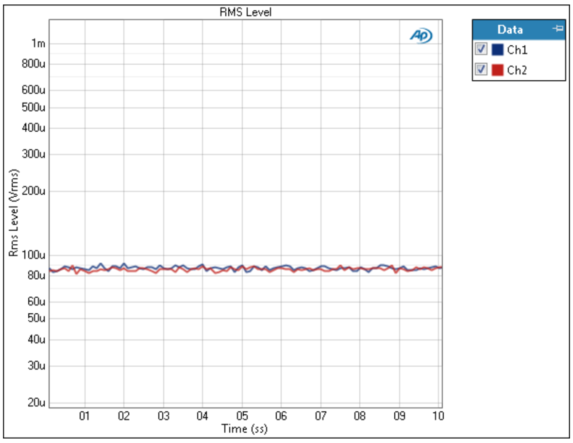
Finally, click the Analyze button. The first settled reading (typically within the first 0.10 second of the file) is shown on the meter bar display. To take a reading later in the file, go to the Settling tab in Advanced Settings and increase the Delay Time to the desired point in the file.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, see Advanced Settings for the Noise Recorder on page 391.

Noise Recorder: RMS Level

The Measurement Recorder: Level measurement result will provide a record of RMS level versus elapsed time for any audio signal. The illustration shows a 10-second record of a noise signal on two channels.



Units

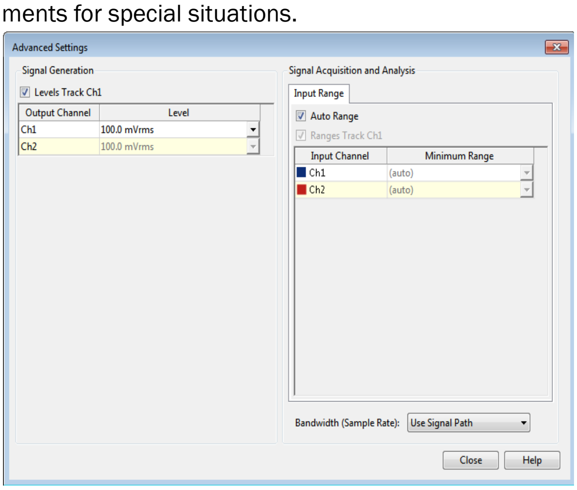
Units available for Measurement Recorder: Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• s	• Vrms	• dBFS
	• dBV	• FS
	• dBu	• %FS
	• dBrA	• dBrA
	• dBrB	• dBrB
	• dBSPL1	• dBSPL1
	• dBSPL2	• dBSPL2
	• dBm	
	• W (watts)	

See Chapter 98 for more information about units of measurement.

Advanced Settings for Noise Recorder

The default settings here are appropriate for most measurements. You may want to make minor adjustments for special situations.



Signal Generation

If Waveform is set to **None**, no signal generation settings will be available.

If Waveform is set to **Noise**, the Output Channel level grid will be available.

If **Track first channel level** is checked (the default), the generator output level value for channel 1 is copied to the other channel(s), and output levels for channel(s) greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channel(s).

To set individual generator output channel levels, uncheck the "Track first channel level" checkbox and enter values in the output channel Level field(s).

Signal Acquisition and Analysis

Input Range tab

Digital inputs do not require ranging, and the Input Range tab is unavailable when inputs are set to a digital format.

You can set a fixed range for each analog input channel for the Measurement Recorder. If "Track first channel range" is checked (the default), the fixed input range setting for channel 1 is copied to the other channels, and range settings for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels.

To set individual analog input channel ranges, uncheck the "Track first channel range" checkbox and enter values in the input channel Range fields.

See page 551 for more information about ranging and autoranging.

Bandwidth (Sample Rate)

This control is only available when the analyzer input configuration is set to Analog.

The Bandwidth (Sample Rate) control allows you to override the analog input bandwidth setting in Signal Path Setup. The input bandwidth set here operates only on the Noise Recorder.

Certain applications that may use the files created here require files of specific sample rate. For analog inputs, the default sample rate of a recorded file is determined by the input bandwidth setting, normally set in Signal Path Setup. This control allows selection of other bandwidths/sample rates. The analog input bandwidth (sample rate) settings, whether made here or in Signal Path Setup, result in channel-count constraints at high sample rates.

The bandwidth choices, followed by their corresponding sample rates, are:

- 2.75 kHz (6 kS/s)

- 3.5 kHz (8 kS/s)
- 5.5 kHz (12 kS/s)
- 7 kHz (16 kS/s)
- 11 kHz (24 kS/s)
- 20 kHz (44.1 kS/s)
- 22 kHz (48 kS/s)
- 40 kHz (88.2 kS/s)
- 45 kHz (96 kS/s)
- 80 kHz (176.4 kS/s)
- 90 kHz (192 kS/s)
- 250 kHz (624 kS/s) (requires the APx555 or the BW52 Option)
- 500 kHz (1.248 MS/s) (requires the APx555 or the BW52 Option)
- 1 MHz (2.496 MS/s) (requires the APx555 or the BW52 Option, set to 1 channel)
- Use Signal Path (the default)

Note that recording files at bandwidths above 90 kHz (sample rates above 192 kHz) requires a very fast PC and hard disk, due to the high data rates. Slower systems can suffer recording interruptions and failure.

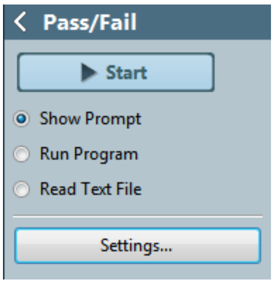
64

Pass/Fail
(Sequence Mode)

Pass/Fail is not a measurement, but a Navigator node designed to provide an opportunity for the operator or some external device to insert a **Pass** or **Fail** result in a sequence, unrelated to APx audio measurement results. The **Pass/Fail** result is included in the Report.

Whether or not this step presents a **Pass, Fail** or **Cancel** result, the Sequence continues.

Pass/Fail is only useful in an automated sequence.



Methods of Response

There are 4 methods of response to the Pass/Fail step in a sequence.

1. The Operator responds to a condition (an indicator light in the DUT, for example) by clicking the **Pass**, **Fail** or **Cancel** button on the prompt displayed on the computer screen.
2. An **Aux Control In** condition is met (a specific pin goes high, for example) for **Pass**, **Fail** or **Cancel**. This could be in response to the operator pushing a physical button, or could be a signal from another device involved in the test. A temperature-sensing device, for example, could send an **Aux Control In Fail** signal if the DUT temperature were out of range. **Aux Control In** conditions for **Pass**, **Fail** or **Cancel** can be set in **Project/Sequence Properties > Dialog Button Aux Control** (see page 482).
3. The step calls a program, which runs to produce a text string. If the resulting text string matches the expected result, the step reports **Pass**; in any other case (wrong result, illegal path, etc.) the step reports **Fail**.
4. The step opens a text file that contains a text string. If the text string matches the expected result, the step reports **Pass**; in any other case (wrong text, illegal path, etc.) the step reports **Fail**.

Operation

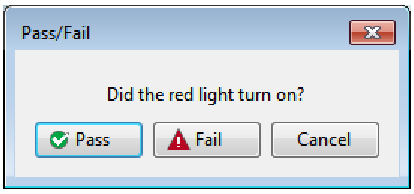
Open **Pass/Fail** and place a check mark on its node to enter it as a sequence step. Move the node to the desired position in the sequence.

Choose the method of response

- Show Prompt
- **Run Program**, or
- Read Text File

Click **Settings** to configure the response.

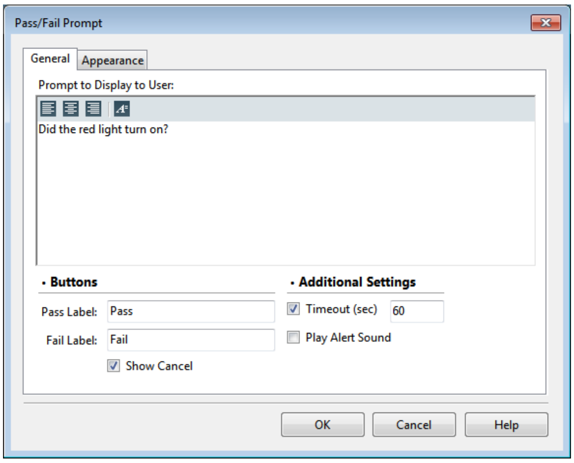
Show Prompt settings



Example prompt

General Tab

Prompt to Display to User



Add any text you want to appear in the prompt here. You can also paste formatted text from the Windows clipboard into the text entry field. Long text can be scrolled in the prompt.

The toolbar at the top of the text entry area provides left, center and right alignment buttons. The font style button opens a dialog where you can select a new font, change point size, color, weight, and so on.

Pass Button

By default, the **Pass** button is labeled “Pass”. You can change the button label by setting the label checkbox and entering new text in the label field.

Fail Button

By default, the **Fail** button is labeled “Fail”. You can change the button label by setting the label checkbox and entering new text in the label field.

Show Cancel

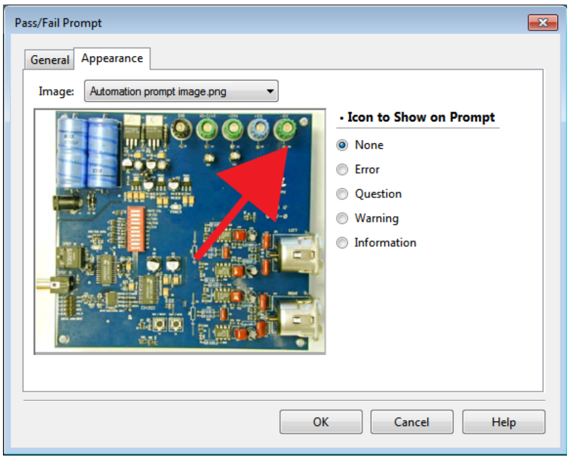
By default, there in no **Cancel** button in the prompt. If you would like to add a **Cancel** button, set the checkbox here.

Close Dialogs using Aux Control

The operator can use **Aux Control In** commands to close the prompts dialogs during production test, enabling the use of physical devices rather than clicking on the computer screen. To configure **Aux Control In** to respond, right-click on the Project node to open the **Project/Sequence Properties** dialog (page 482).

Appearance tab

Image



You can place an image at the top of the user prompt. This could be a photograph or diagram with additional visual information to help the user.

Beneath the Image area, click the **Browse...** button and navigate to an image file (page 501). Select the image and click **Open** to place the image in the prompt. When an image is used, the image file is also attached to the project file.

Larger images will cause the prompt window size to increase to fill the available screen space, and will be scaled to fit in the prompt window.

Click **Delete** to remove the image from the prompt. Image files attached to the project but not currently in use can be removed by navigating to the dialog at **File > Manage Attached Project Items...** (see page 24).

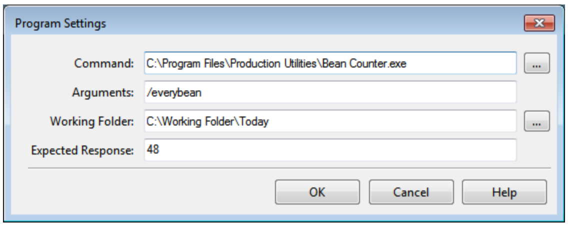
Icon to Show on Prompt

Check one of the following (or **None**) to place a standard Windows attention icon on the prompt.

- None
-  Error
-  Question
-  Warning
-  Information

Run Program

When configured to **Run Program**, the **Pass/Fail** step calls a program, which runs to produce a text string. If the resulting text string matches the expected result, the step reports **Pass**; in any other case (wrong result, illegal path, etc.) the step reports **Fail**.



Command:

Enter in the path and command for the external program here, or browse to the program using the browse button.

Arguments:

Enter in the command arguments here, if any.

Working Folder:

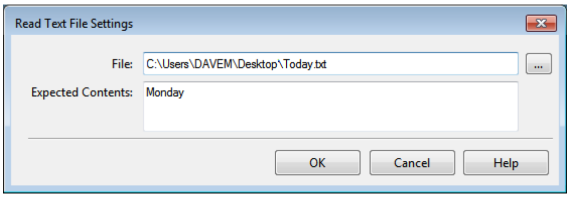
Enter in the working folder here, if necessary, or browse to the folder using the browse button.

Expected Response

Enter the exact text you expect as a response.

Read Text File

When configured to **Read Text File**, the **Pass/Fail** step opens a text file. If the resulting text string in the file matches the expected result, the step reports **Pass**; in any other case (wrong text string, illegal path, etc.) the step reports **Fail**.



File:

Enter in the path and filename for the text file here, or browse to the file using the browse button.

Expected Contents

Enter the exact text you expect to be in the file.

65

PESQ
(Sequence Mode)

This measurement requires a software option key. See page 166 for more information about software options.

PESQ (Perceptual Evaluation of Speech Quality) is a method of grading the quality of an audio communications channel, resulting in a MOS (Mean Opinion Score). See page 402 for more information about PESQ.

The PESQ measurement uses a speech sample stimulus signal that is passed through the DUT and compared with the original speech signal serving as a reference. A MOS result and a number of diagnostic cause analysis results are provided.

The PESQ measurement results available in APx500 are:

- MOS
- PESQ
- MOS vs Time
- Average Delay
- Delay vs Time
- Acquired Waveform
- Reference Waveform

Operation

If you have not yet set up your test, first go to Chapter 6 “Signal Path Setup.” When Generator and Analyzer settings (below) are set for your test, click **Start**.

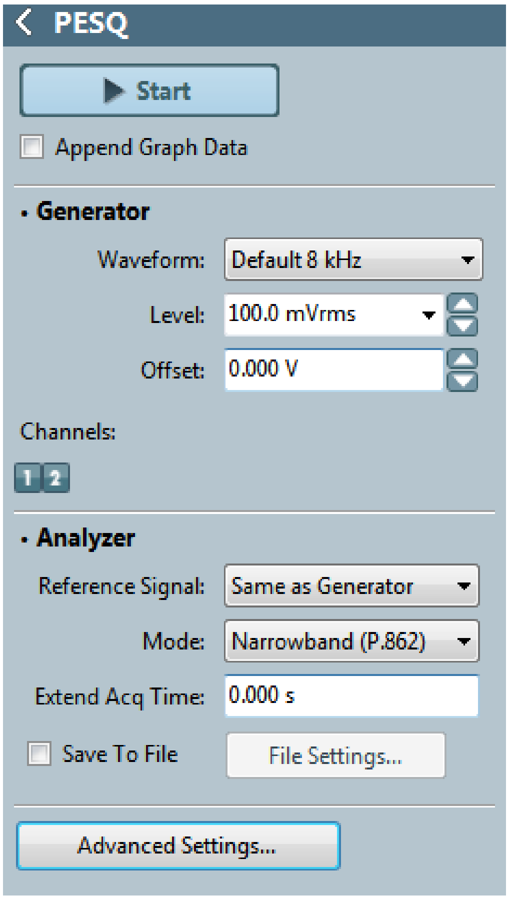
Append Graph Data

Normally, the current graph data for all results is deleted each time you start a new measurement. If the **Append** box is checked, the current data is kept in memory as a **Data Set** (see page 165), and the new measurement data is appended as a new data set. You can append many **Data Sets**.

Generator

Waveform

For the PESQ measurement, the APx generator can play a speech sample that conforms to PESQ requirements. APx provides one speech sample at a sample rate of 8 kHz and one at 16 kHz embedded in the PESQ measurement. Select one of these, or browse for a waveform file. An additional 32 speech samples are installed as .WAV waveforms during the installation of APx500.



Conforming waveforms are mono WAV files at a sample rate of 8 kHz or 16 kHz. Content should be speech samples. Length is typically 5 to 10 seconds; maximum length is 35 seconds.

Description

This field displays information about the current waveform.

Level

Set the generator output level in this field.

The Level control sets the level of the generator output to the value entered in the Level field, for a maximum level in the waveform file. See Generator Waveform Files for a general discussion of generator level when playing waveform files.

Channel

The generator will output the speech samples to the DUT on the selected generator channels.

Start

The Start button runs the measurement. For closed loop operation, Start outputs the speech sample signal and prepares the analyzer for acquisition. For External Source (open loop) measurements, Start prepares the analyzer for acquisition, awaiting a speech sample from the DUT.

Analyzer

Filters

No local low pass, high pass or weighting filters are available for this measurement. However, low pass, high pass and weighting filter settings made in Signal Path Setup > Input/Output will affect this measurement. Read more about APx Filters in Chapter 91.

Reference Signal

in Closed Loop configuration...

In closed loop configuration, the Reference Signal is usually the Generator Waveform that is currently selected.

In situations where you have externally modified or EQ'd a Reference Waveform (equalized to compensate for loudspeaker response, for example), you have the option to select an alternative Reference Signal (the unmodified reference waveform, for example) here.

in External Source configuration...

When in External Source (open loop) configuration, you must choose a reference speech sample file here. The speech sample file (or compensated speech sample file) being played from the DUT must correspond to the Reference Signal chosen here.

Mode

Conforming PESQ speech sample files have an 8 kHz sample rate for testing narrowband channels, and a 16 kHz sample rate for testing wideband channels. The Mode setting is forced to Wideband for 16 kHz sample rate reference files. For 8 kHz sample rate reference files, the Mode should be set to Narrowband. In mixed narrowband/wideband environments, set the Mode selector to Wideband, even if the reference file has a sampling rate of 8 kHz.

Extend Acq. Time

The acquisition time is set to the length of the speech sample (selected either in Waveform, for closed loop, or Reference Signal for External Source (open loop), plus the time set in this field. If you suspect you are getting poor results because the acquisition length is truncating the input signal, add time here.

This field is not available if Input Configuration is set to File. In File Input, the acquisition time is set to the length of the input file. In either case, the maximum acquisition length is 35 seconds.

Save to File

You can record the PESQ audio signal to a file as the signal is being acquired. Click Save to File before you run the PESQ measurement. See Recording Audio to a File on page 347 for more information.

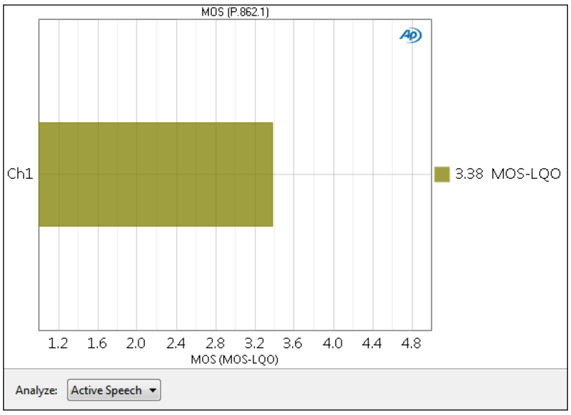
File Settings

When saving a PESQ acquisition to a file, click File Settings to specify the file name and location.

Advanced Settings

If you'd like to adjust analyzer ranging parameters, click Advanced Settings, where you will also find a PESQ-specific setting to defeat auto adjustment of leading / trailing silence for backward-compatible PESQ conformance testing. See Advanced Settings for PESQ on page 401. Read more about Units in Chapter 98.

PESQ: MOS result



The **PESQ: MOS** result is the primary result for the PESQ measurement. The PESQ score is mapped to MOS-LQO using P.862.1 for narrowband analysis, or P.862.2 for wideband analysis.

Additional Control for MOS result

Analyze

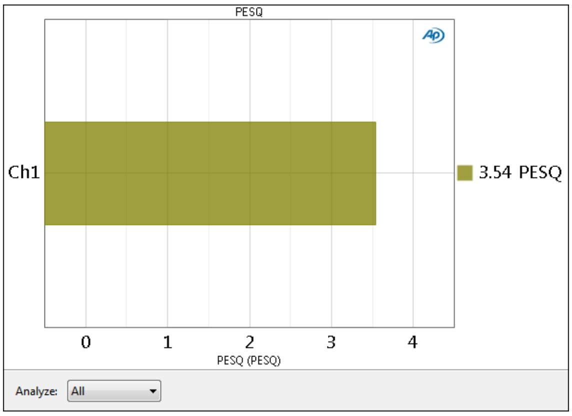
- All
  Analyzes the entire speech sample. This mode is supported by the ITU-T P.862 standard for MOS results.
- Active Speech
  Analyzes only the speech portions of the sample.
- Silence
  Analyzes only the relative silence between the speech portions.

Units

Units for PESQ: MOS are:

- MOS-LQO

PESQ: PESQ result



The **PESQ: PESQ** result shows the PESQ score for narrowband analysis. The PESQ score should be treated as an intermediate result, and the MOS scores reported instead.

Additional Control for MOS result

Analyze

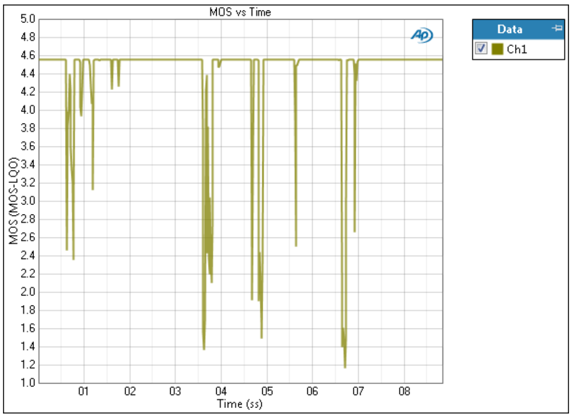
- All
  Analyzes the entire speech sample. This mode is supported by the ITU-T P.862 standard for MOS results.
- Active Speech
  Analyzes only the speech portions of the sample.
- Silence
  Analyzes only the relative silence between the speech portions.

Units

Units for PESQ: PESQ results are:

- PESQ

PESQ: MOS vs Time result



The PESQ: MOS vs Time is useful for cause analysis. The result shows a graph of the MOS score versus elapsed time.

Units

Units for PESQ: Delay vs Time are:

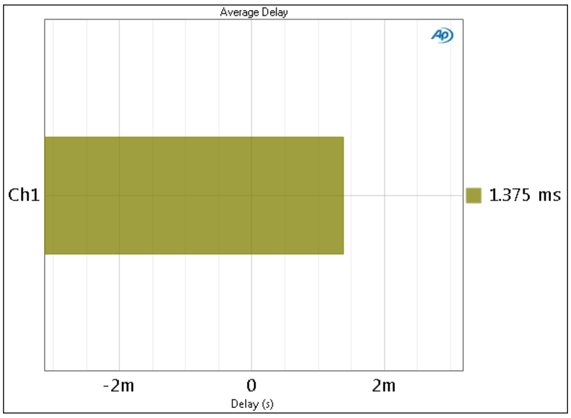
X-axis

- s (seconds)

Y-axis

- MOS-LQO

PESQ: Average Delay result



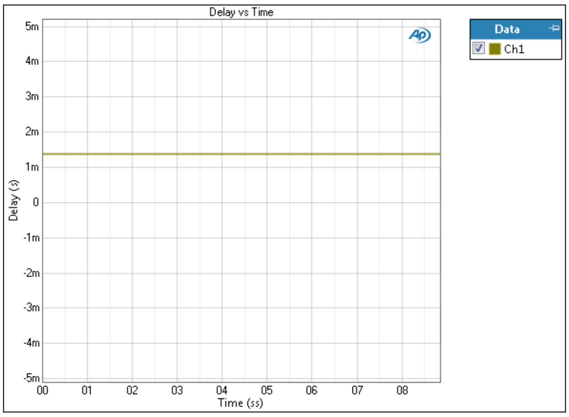
The PESQ: Average Delay result is useful for cause analysis. The result shows the average signal delay through the DUT or system.

Units

Units for PESQ: Average Delay are:

- s (seconds)

PESQ: Delay vs Time result



The PESQ: Delay vs Time result is useful for cause analysis. The result shows signal delay time versus elapsed time.

Units

Units for PESQ: Delay vs Time are:

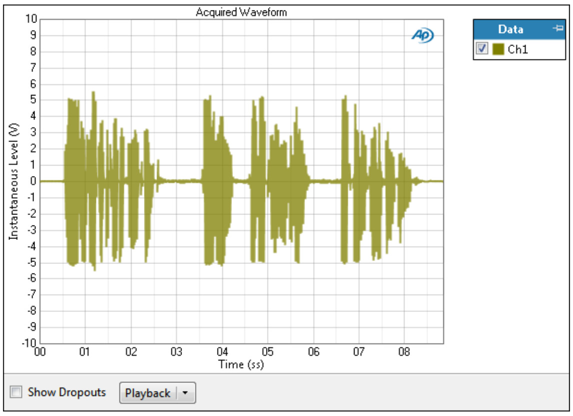
X-axis

- s (seconds)

Y-axis

- s (seconds)

PESQ: Acquired Waveform result



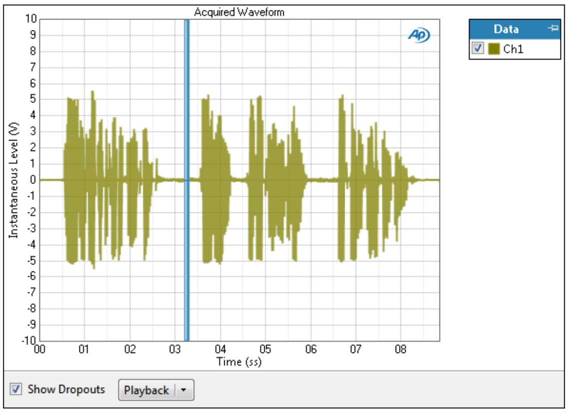
The PESQ: Acquired Waveform result is useful for cause analysis. The result shows a time-domain view of the acquired speech sample waveform.

Additional Control for Acquired Waveform result

Show Dropouts

APx compares the acquired waveform to the reference waveform. A signal dropout is defined as a location where there is signal in the reference waveform but no signal in the acquired waveform.

When the Show Dropouts checkbox is checked (the default), APx highlights dropout locations on the graph.



Playback

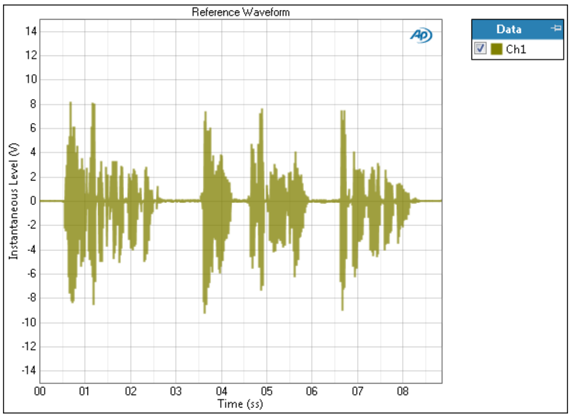
You can play the PESQ measurement Acquired Waveform through the PC Windows audio. Click the Playback menu to list the available acquired channels, and play the desired channel by selecting it.

Units

Units available for PESQ: Acquired Waveform are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• s (seconds)	• V	• D • hex

PESQ: Reference Waveform result



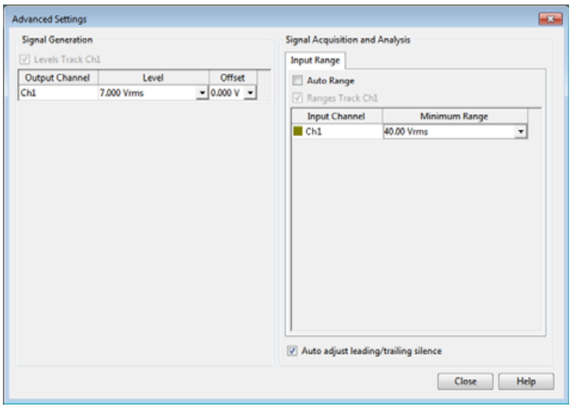
The PESQ: Reference Waveform result is result is useful for cause analysis. The result shows a time-domain view of the reference speech sample waveform.

Units

Units for PESQ: Reference Waveform are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• s (seconds)	• V	• D • hex

Advanced Settings for PESQ



The default settings here are appropriate for meter (single value) measurements under most conditions. You may want to make minor adjustments for special situations.

Signal Generation

If **Track first channel level** is checked (the default), the generator output level value for channel 1 is copied to channel 2, and the level for channel 2 cannot be edited. Any changes made to channel 1 are reproduced in channel 2.

To set individual generator output channel levels, uncheck the “Track first channel level” checkbox and enter values in the output channel Level fields.

File Playback

For generator waveform file playback, you can view the length of the file, and you can adjust the playback start position. Select Seconds or Samples for the Length and Start Position units. Enter a new Start Position if desired.

Signal Acquisition and Analysis

Input Range tab

Digital inputs do not require ranging, and the Input Range tab is unavailable when inputs are set to a digital format.

If **Auto Range** is checked (the default), each analog channel determines its input range automatically, based on the level of the input signal. If the input signal level changes beyond a ranging threshold, Auto Range will cause the input ranging circuits to move up or down for proper ranging.

Auto Range and PESQ

PESQ and other perceptual audio measurements use voice samples as test signals, and may benefit from the use of fixed ranging. See “When autoranging isn’t suitable” on page 552.

If **Auto Range** is unchecked, you can set a fixed range for each analog input channel. If **Track first channel range** is checked (the default), the fixed input range setting for channel 1 is copied to the other channels, to channel 2, and the range setting for channel 2 cannot be edited. Any changes made to channel 1 are reproduced in channel 2.

To set individual analog input channel ranges, uncheck the “Track first channel range” checkbox and enter values in the input channel Minimum Range fields.

Auto adjust leading/trailing silence

The PESQ implementation will automatically adjust the length of the leading and trailing silence of a speech sample to ensure that the length conforms to the ITU recommendations. When this checkbox is checked (the default), the automatic length adjustment is ON.

For PESQ conformance testing, switch this feature OFF to remain backward compatible.

More about PESQ and perceptual audio testing

Overview

PESQ (Perceptual Evaluation of Speech Quality), like POLQA, is a method of grading the quality of an audio communications channel. Perceptual audio evaluation methods were developed to enable evaluation of audio quality in channels of compromised performance, where conventional measurements such as THD+N may provide results that are not useful.

Perceptual audio evaluation techniques are based on subjective experiments with listeners, and do not use simple periodic waveforms or sweeps as most APx measurements do; consequently, the perceptual evaluation results cannot be stated as physical values such as volts or hertz. Instead, listeners' opinions of quality (or algorithms designed to respond similarly) are averaged and scored.

PESQ and POLQA are both methods of generating MOS results without actually using a room full of listeners. Instead, pre-recorded speech samples are passed through a DUT or system and are compared with the original speech sample. An algorithm evaluates the difference between the reference audio and the degraded audio, and produces a result.

Channels of compromised audio quality

Many speech communications channels have limited audio capabilities, when compared to full fidelity channels optimized for music transmission. The limited capabilities of speech channels may provide other benefits: lower cost, lower bandwidth, lower power requirements, and so on.

Most of the I/O interfaces offered in an APx analyzer have very high fidelity capabilities, but in some cases the configuration for a low data rate DUT may make PESQ or POLQA-type evaluations valuable. Passing a stimulus signal through a telephone network, for example, will likely degrade the signal substantially, as will low data rate codecs. The Bluetooth HSP and HFP profiles, in particular, are of moderate fidelity. Bluetooth manufacturers often use PESQ or POLQA to characterize Bluetooth devices operating in these profiles.

MOS (Mean Opinion Score)

Perceptual audio evaluations are typically made with a number of listeners, who provide opinions about the quality of audio samples they have listened to. Typically, many audio samples, many listeners and many iterations are used, and the results are averaged into a MOS (Mean Opinion Score). MOS is expressed in a unitless scale from 1 to 5, representing the average opinion of quality, as shown in this table:

MOS	*Quality*	*Impairment*
5	Excellent	Imperceptible
4	Good	Perceptible but not annoying
3	Fair	Slightly annoying
2	Poor	Annoying
1	Bad	Very annoying

MOS results in APx are reported as MOS-LQO (LQO stands for Listening-only Quality, Objective). For PESQ results, these reference the PESQ-to-MOS mapping: P.862.1 for narrowband, and P.862.2 for wideband.

PESQ ITU-T P.862 (2001)

The PESQ algorithm is defined in ITU-T Recommendation P.862 (2001).

Narrowband / Wideband testing

Narrowband has a bandwidth of 300–3400 Hz and uses a filter in analysis to emulate the characteristics of a telephone handset. Digital files for PESQ narrowband have a sampling rate of 8 kHz. Narrowband PESQ results are mapped to MOS-LQON according to P.862.1.

Wideband has a bandwidth of 50–7000 Hz and uses flat filtering. Digital files for PESQ wideband have a sampling rate of 16 kHz. Wideband PESQ results are mapped to MOS-LQOW according to P.862.2. The mapping algorithm in P.862.2 is slightly different from the algorithm in P.862.1.

In either case, the PESQ result itself should be considered an intermediate result, and the MOS score the primary result.

The standard forbids mixing of wideband and narrowband results. If wideband networks have to be compared to narrowband networks, then the wideband version of PESQ has to be used in both cases.

See http://www.itu.int/rec/T-REC-P.862/ for the latest ITU standard.

The PESQ score is available for the entire signal, or for the active speech parts of the signal only, or for the silent periods only.

Speech samples

PESQ evaluates speech samples. You can record your own samples, or obtain speech sample files from other sources. The ITU has created a number of samples designed to cover the range of speech sounds uttered by men and women in a range of languages. These are available at http://www.itu.int/net/itu-t/sigdb/genaudio/Pseries.htm and http://www.itu.int/rec/T-REC-P.501-200912-S/en .

See the ITU P501E documentation at the second link for detailed information about creating and using speech samples.

For PESQ, APx500 provides one speech sample at a sample rate of 8 kHz and one at 16 kHz embedded in the measurement. An additional 32 speech samples are installed as .WAV files during the installation of APx500.

The PESQ algorithm has been extensively tested against a number of European and Asian languages and performs well. For detailed information, refer to the ITU publication ITU-T-REC-P862 3-200711.pdf.

Averaging

Averaging multiple speech samples across multiple iterations provides the most reliable MOS results. APx500 provides a PESQ (Averaged) measurement that enables you to average up to 64 speech samples or iterations.

Other results

The primary PESQ results in APx500 are the MOS (P.862.1) result and the MOS Average (P.862.1) result.

Other results are provided that are unsupported by ITU-T P.862. These results may be helpful in diagnostics and cause analysis in the DUT or system.

PESQ (Averaged)
(Sequence Mode)

This measurement requires a software option key. See page 166 for more information about software options.

PESQ (Perceptual Evaluation of Speech Quality) is a method of grading the quality of an audio communications channel, resulting in a MOS (Mean Opinion Score). See page 402 for more information about PESQ.

This measurement provides a means to average a number of PESQ measurements, producing a single MOS Average result per channel. Best practice recommends at least two male and two female voice samples, each averaged several times. Some testers will use many different voice samples, repeated and averaged many times.

The PESQ (Averaged) measurement uses a speech sample stimulus signal that is passed through the DUT and compared with the original speech signal serving as a reference. A MOS result and a number of diagnostic cause analysis results are provided.

The PESQ (Averaged) measurement is not available in External Source (open loop) configuration, or when Input Configuration is set to File.

The PESQ (Averaged) measurement results available in APx500 are:

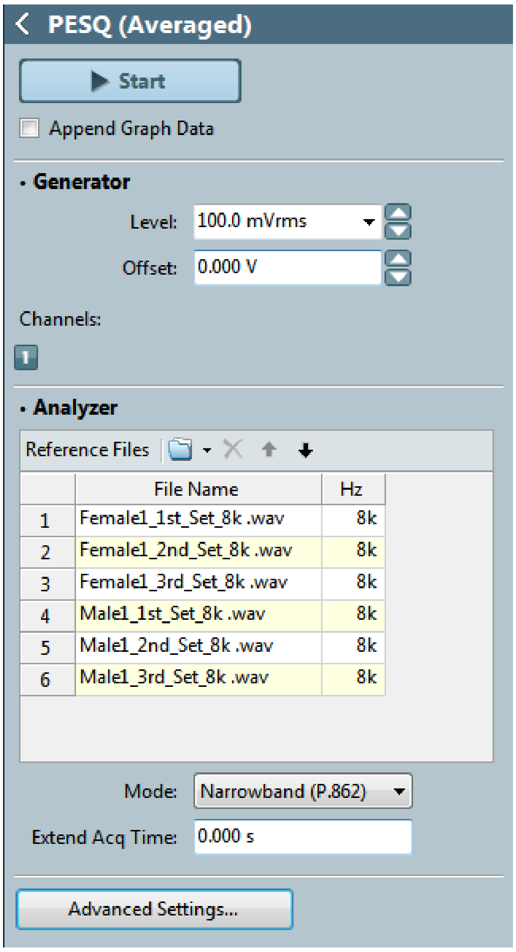
- MOS Average
- File MOS

Operation

If you have not yet set up your test, first go to Chapter 6 “Signal Path Setup.”

Start

When Signal Generation and Signal Analysis (below) are set and you have imported all your reference files, click Start. The files will be played through the DUT, one at a time.



Append Graph Data

Normally, the current graph data for all results is deleted each time you start a new measurement. If the **Append** box is checked, the current data is kept in memory as a **Data Set** (see page 165), and the new measurement data is appended as a new data set. You can append many **Data Sets**.

Generator

Level

Set the generator output level in this field.

The Level control sets the level of the generator output to the value entered in the Level field, for a maximum level in the waveform file. See Chapter 14 for a general discussion of generator level when playing waveform files.

Generator Channels

The generator will output the speech samples to the DUT on the selected generator channels.

Reference Files

For a PESQ measurement, a reference file in the APx is played through the generator, and the output of the DUT or communications channels under test is compared to the reference file. For multiple averaging iterations with the same file or with different files, browse to enter the files in the grid here. These reference files will be played and analyzed in the order shown in the grid. The minimum number of files is 2; the maximum number of files is 64.

Audio Precision provides 32 speech samples, installed as .WAV waveforms during the installation of APx500.

Conforming waveforms are mono WAV files at a sample rate of 8 kHz or 16 kHz. Content should be speech samples. Length is typically 5 to 10 seconds; maximum length is 35 seconds.

Use the folder icon to open a file browser. You can add files already in the project, or you can browse for files on disk. If you choose files of mixed sampling rates, the analysis Mode will be constrained to Wideband.

Select a file to delete it, or to move it up or down in the list using the arrow buttons.

Analyzer

Filters

No local low pass, high pass or weighting filters are available for this measurement. However, low pass, high pass and weighting filter settings made in Signal Path Setup > Input/Output will affect this measurement. Read more about APx Filters in Chapter 91.

As the generator moves through the Reference Files list, the analyzer acquires the signal and matches the audio received to the reference files, one at a time. The PESQ algorithm assigns a PESQ score, which is then mapped to a MOS score. A running average of the MOS scores is displayed, with the final value being the average of the scores of all the reference files.

Mode

Conforming PESQ speech sample files have an 8 kHz sample rate for testing narrowband channels, and a 16 kHz sample rate for testing wideband channels. The Mode setting is forced to Wideband for 16 kHz sample rate reference files. For 8 kHz sample rate reference files, the Mode should be set to Narrowband. In mixed narrowband/wideband environments, set the Mode selector to Wideband, even if the reference file has a sampling rate of 8 kHz. For files of mixed sampling rates, the Mode will be constrained to Wideband.

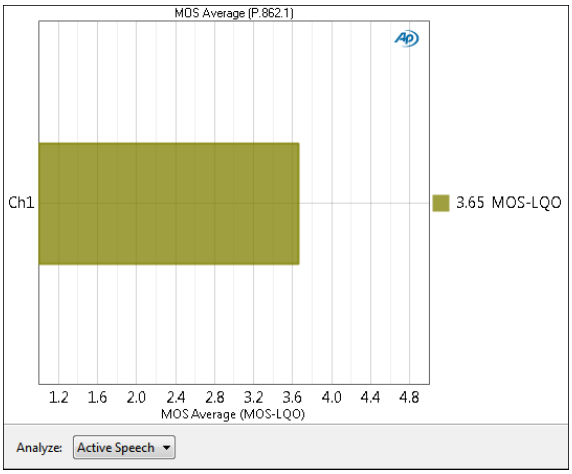
Extend Acq. Time:

The acquisition time is set to the length of the speech sample (selected either in Waveform, for closed loop, or Reference Signal for External Source (open loop), plus the time set in this field. If you suspect you are getting poor results because the acquisition length is truncating the input signal, add time here.

Advanced Settings

If you'd like to adjust analyzer ranging parameters, click Advanced Settings, where you will also find a PESQ-specific setting to defeat auto adjustment of leading / trailing silence for backward-compatible PESQ conformance testing. See Advanced Settings for PESQ on page 401. Read more about Units in Chapter 98.

PESQ (Averaged): MOS Average result



The PESQ: MOS Average result is the primary result for the PESQ (Averaged) measurement. The PESQ score is mapped to MOS-LQO using P.862.1 for narrowband analysis, or P.862.2 for wideband analysis. This result provides an average of all the MOS scores of the speech samples in the Reference Files list.

Additional Control for MOS result

Analyze

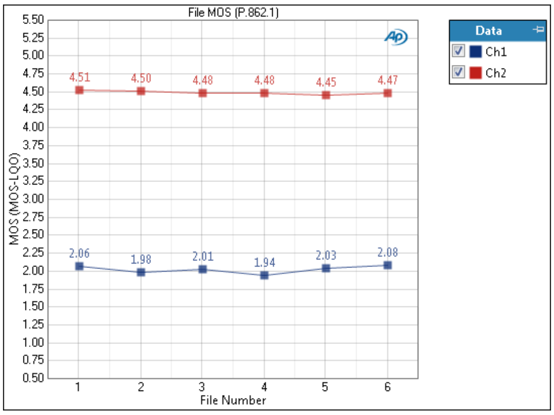
- All
  Analyzes the entire speech sample. This mode is supported by the ITU-T P.862 standard for MOS results.
- Active Speech
  Analyzes only the speech portions of the sample.
- Silence
  Analyzes only the relative silence between the speech portions.

Units

Units for PESQ (Averaged): MOS Average are:

- MOS-LQO

PESQ (Averaged): File MOS result



The PESQ (Averaged): File MOS result is useful for cause analysis. The result shows the MOS score for each of the files in the Reference Files list.

Additional Control for MOS result

Analyze

- All
  Analyzes the entire speech sample. This mode is supported by the ITU-T P.862 standard for MOS results.
- Active Speech
  Analyzes only the speech portions of the sample.
- Silence
  Analyzes only the relative silence between the speech portions.

Units

Units for PESQ (Averaged): File MOS are:

X-Axis

- File number

Y-Axis

- MOS-LQO

67

POLQA (Sequence Mode)

This measurement requires a software option key. *See page 166 for more information about software options.*

POLQA (Perceptual Objective Listening Quality Assessment) is a method of grading the quality of an audio communications channel, resulting in a MOS (Mean Opinion Score). See page 413 for more information about POLQA.

The POLQA measurement uses a speech sample stimulus signal that is passed through the DUT and compared with the original speech signal serving as a reference. A MOS result and a number of diagnostic cause analysis results are provided.

The POLQA measurement results available in APx500 are:

- MOS (LC IRS P.863)
- POLQA G.107
- MOS vs Time
- Average Delay
- Delay vs Time
- Acquired Waveform
- Reference Waveform

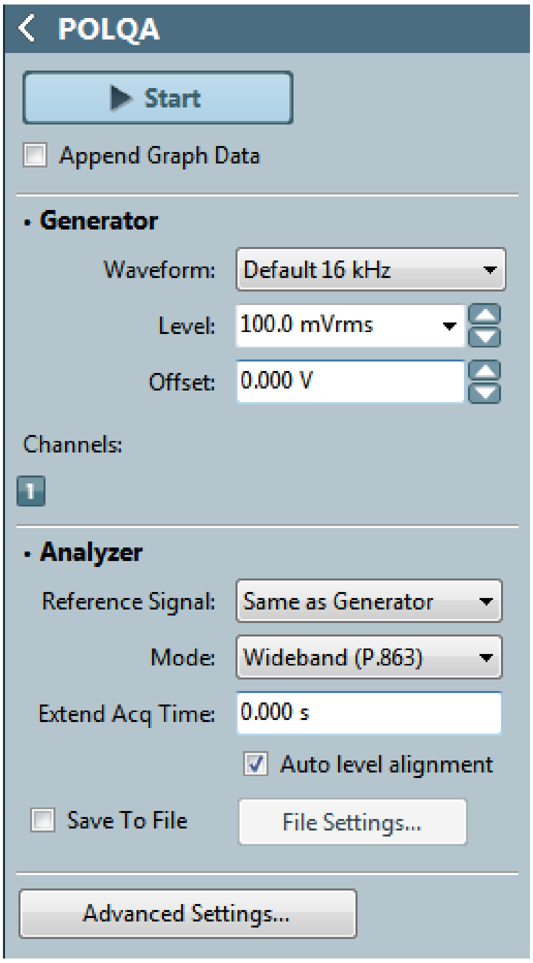
Operation

If you have not yet set up your test, first go to Chapter 6 “Signal Path Setup.”

When Signal Generation and Signal Analysis (below) are set for your test, click **Start**.

Append Graph Data

Normally, the current graph data for all results is deleted each time you start a new measurement. If the **Append** box is checked, the current data is kept in memory as a **Data Set** (see page 165), and the new measurement data is appended as a new data set. You can append many **Data Sets**.



Generator

Waveform

For the POLQA measurement, the APx generator can play a speech sample that conforms to POLQA requirements. APx provides one speech sample at each of three sample rates: 8 kHz, 16 kHz and 48 kHz, embedded in the POLQA measurement. Select one of these, or browse for a waveform file. An additional 48 speech samples are installed as .WAV waveforms during the installation of APx500.

Conforming waveforms are mono WAV files at a sample rate of 8 kHz, 16 kHz or 48 kHz. Content should be speech samples. Length is typically 5 to 10 seconds; maximum length is 12 seconds.

Description

This field displays information about the current waveform.

Level

Set the generator output level in this field.

The Level control sets the level of the generator output to the value entered in the Level field, for a maximum level in the waveform file. See Chapter 14 for a general discussion of generator level when playing waveform files.

Channel

The generator will output the speech samples to the DUT on the selected generator channels.

Start

The Start button runs the measurement. For closed loop operation, Start outputs the speech sample signal and prepares the analyzer for acquisition. For External Source (open loop) measurements, Start prepares the analyzer for acquisition, awaiting a speech sample from the DUT.

Analyzer

Note: the SW-POLQA-2 iButton only provides POLQA analysis on input channels 1 and 2. To authorize more channels, contact your Audio Precision representative.

Filters

No local low pass, high pass or weighting filters are available for this measurement. However, low pass, high pass and weighting filter settings made in Signal Path Setup > Input/Output will affect this measurement. Read more about APx Filters in Chapter 91.

Reference Signal

in Closed Loop configuration...

In closed loop configuration, the Reference Signal is usually the Generator Waveform that is currently selected.

In situations where you have externally modified or EQ'd a Reference Waveform (equalized to compensate for loudspeaker response, for example), you have the option to select an alternative Reference Signal (the unmodified reference waveform, for example) here.

in External Source configuration...

When in External Source (open loop) configuration, you must choose a reference speech sample file here. The speech sample file (or compensated speech sample file) being played from the DUT must correspond to the Reference Signal chosen here.

Note: the use a reference file whose sample rate differs from the target file is not standards compliant, and an error will be thrown. To suppress this error, set the Auto Resampling checkbox in Advanced Settings.

Mode

Conforming POLQA speech sample files have an 8 kHz sample rate for testing Narrowband (P.863) channels, a 16 kHz sample rate for testing Wideband (P.863) channels, and a 48 kHz sample rate for testing Super-Wideband (P.863) channels. The mode is selected automatically by the POLQA measurement based on the sample rate contained in the reference file.

A conforming POLQA speech file is also optimized for an Active Speech Level (ASL) of –26 dBov ASL, using the ITU-T P.56 meter. See Level alignment for POLQA on page 415.

Extend Acq. Time

The acquisition time is set to the length of the speech sample (selected either in Waveform, for closed loop, or Reference Signal for External Source (open loop), plus the time set in this field. If you suspect you are getting poor results because the acquisition length is truncating the input signal, add time here; 10 s is the maximum.

This field is not available if Input Configuration is set to File. In File Input, the acquisition time is set to the length of the input file. In either case, the maximum acquisition length is 35 seconds.

Save to File

You can record the POLQA audio signal to a file as the signal is being acquired. Click Save to File before you run the POLQA measurement. See Recording Audio to a File on page 347 for more information.

File Settings

When saving a POLQA acquisition to a file, click File Settings to specify the file name and location.

Auto level alignment

A key difference between POLQA and PESQ is in sensitivity to signal level. PESQ ignores level, but POLQA will produce poor MOS scores if the active speech level (ASL) presented to the algorithm is not matched (within a certain range) to the ASL of the reference file, which itself must be ASL-optimized for POLQA.

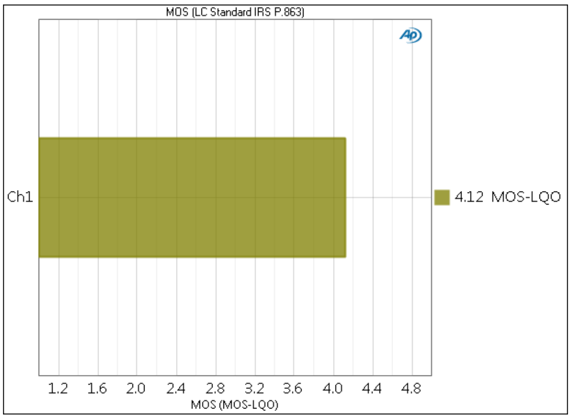
APx provides an Auto level alignment checkbox to defeat this level sensitivity. By default, **Auto level alignment** is **ON**, which makes good MOS scores easier to obtain. This is useful if your test requirements do not consider signal level, or if temporarily defeating level sensitivity is helpful in diagnosis. However, when **Auto level alignment** is **ON**, the APx POLQA measurement is not compliant with the ITU-T P.863 standard.

See **Level alignment for POLQA** on page 415 for information on adjusting levels to optimize MOS scores while remaining standards-compliant.

Advanced Settings

If you'd like to adjust analyzer ranging parameters, click Advanced Settings, where you will also find POLQA-specific settings for Auto Resampling. See Advanced Settings on page 413. Read more about Units in Chapter 98.

POLQA: MOS (LC IRS P.863) result



The **POLQA: MOS** result is the primary result for the POLQA measurement. The POLQA score is mapped to MOS-LQO LC STANDARD IRS P.863.

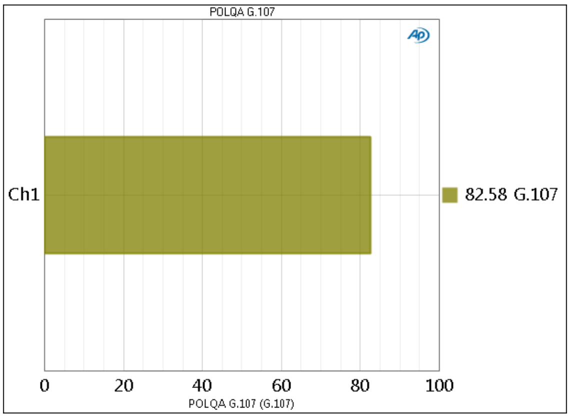
Units

Units for POLQA: MOS (LC STANDARD IRS P.863).

are:

- MOS-LQO

POLQA: POLQA G.107 result



The **POLQA: POLQA G.107** result shows the POLQA score for Wideband and SuperWideband analysis. This result is invalid and not populated with data in Narrowband analysis.

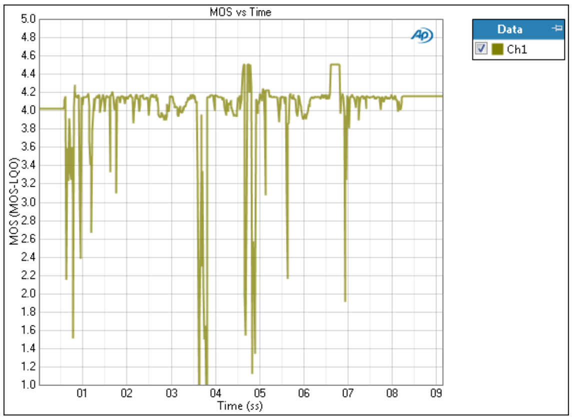
The POLQA G.107 score should be treated as an intermediate result, and the MOS scores reported instead

Units

Units for POLQA: POLQA G.107 results are:

- G.107

POLQA: MOS vs Time result



The POLQA: MOS vs Time is useful for cause analysis. The result shows a graph of the MOS score versus elapsed time.

Units

Units for POLQA: Delay vs Time are:

X-axis

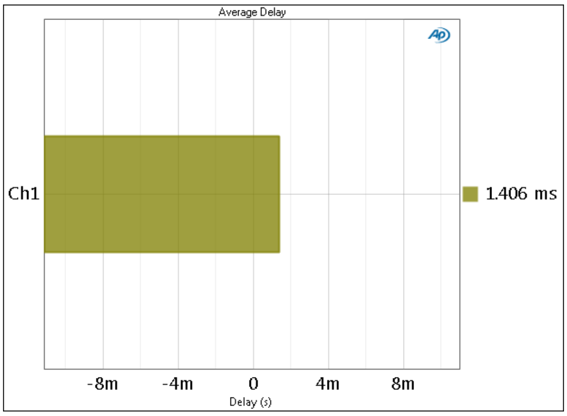
- s (seconds)

Y-axis

- MOS-LQO

Read more about Units in Chapter 98.

POLQA: Average Delay result



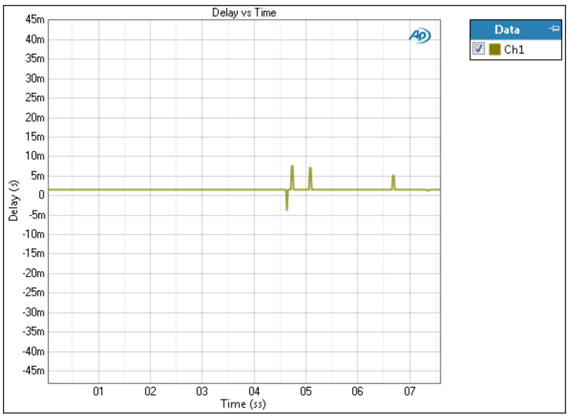
The POLQA: Average Delay result is useful for cause analysis. The result shows the average signal delay through the DUT or system.

Units

Units for POLQA: Average Delay are:

- s (seconds)

POLQA: Delay vs Time result



The POLQA: Delay vs Time result is useful for cause analysis. The result shows signal delay time versus elapsed time.

Units

Units for POLQA: Delay vs Time are:

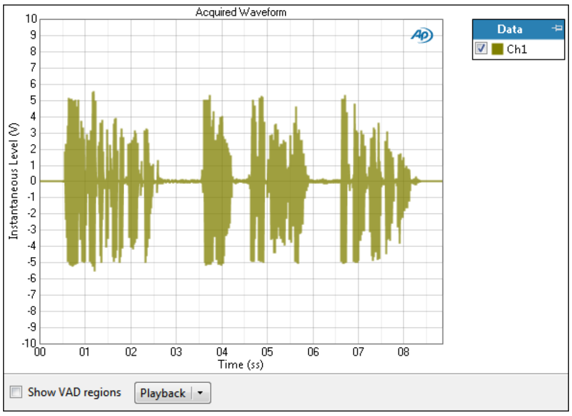
X-axis

- s (seconds)

Y-axis

- s (seconds)

POLQA: Acquired Waveform result



The POLQA: Acquired Waveform result is useful for cause analysis. The result shows a time-domain view of the acquired speech sample waveform.

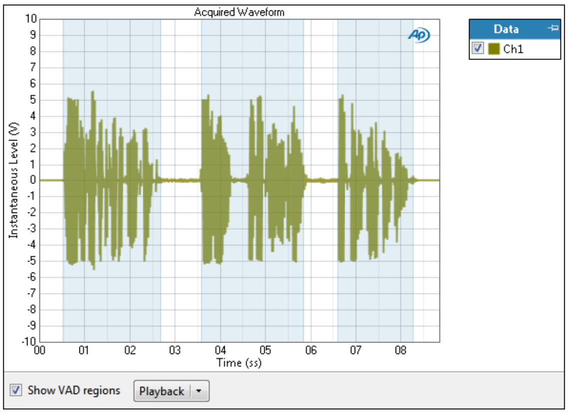
Additional Controls for Acquired Waveform result

Show VAD

If **Show VAD regions** is checked the result will indicate graphically on the acquired waveform where the POLQA algorithm's Voice Activity Detection has found voice sounds and performed its analysis. VAD data is not exported nor retained in the project.

Playback

You can play the POLQA measurement Acquired Waveform through the PC Windows audio. Click the Playback menu to list the available acquired channels, and play the desired channel by selecting it.

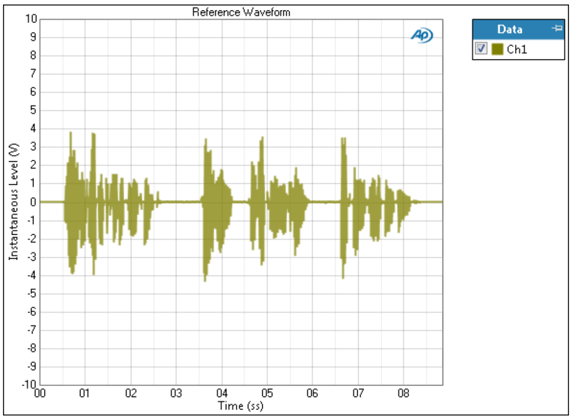


Units

Units available for POLQA: Acquired Waveform are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• s (seconds)	• V	• D • hex

POLQA: Reference Waveform result



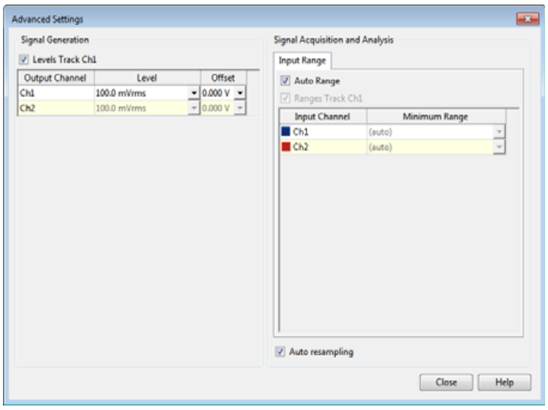
The POLQA: Reference Waveform result is result is useful for cause analysis. The result shows a time-domain view of the reference speech sample waveform.

Units

Units for POLQA: Reference Waveform are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• s (seconds)	• V	• D • hex

Advanced Settings for POLQA



The default settings here are appropriate for meter (single value) measurements under most conditions. You may want to make minor adjustments for special situations.

Signal Generation

If **Track first channel level** is checked (the default), the generator output level value for channel 1 is copied to channel 2, and the level for channel 2 cannot be edited. Any changes made to channel 1 are reproduced in channel 2.

To set individual generator output channel levels, uncheck the "Track first channel level" checkbox and enter values in the output channel Level fields.

Signal Acquisition and Analysis

Input Range tab

Digital inputs do not require ranging, and the Input Range tab is unavailable when inputs are set to a digital format.

If **Auto Range** is checked (the default), each analog channel determines its input range automatically, based on the level of the input signal. If the input signal level changes beyond a ranging threshold, Auto Range will cause the input ranging circuits to move up or down for proper ranging.

Auto Range and POLQA

POLQA and other perceptual audio measurements use voice samples as test signals, and may benefit from the use of fixed ranging. See "When autoranging isn't suitable" on page 552.

If **Auto Range** is unchecked, you can set a fixed range for each analog input channel. If **Track first channel range** is checked (the default), the fixed input range setting for channel 1 is copied to the other channels, to channel 2, and the range setting for channel 2 cannot be edited. Any changes made to channel 1 are reproduced in channel 2.

To set individual analog input channel ranges, **uncheck** the "Track first channel range" checkbox and enter values in the input channel Minimum Range fields.

Auto resampling

If this checkbox is checked, the POLQA algorithm defaults to the lowest sampling rate when it detects a difference between the reference and target signals, and suppresses the error that is thrown. This may result in an improved score, but the result will not be standards compliant.

More about POLQA and perceptual audio testing

Overview

POLQA (Perceptual Objective Listening Quality Assessment). like PEAQ, is a method of grading the quality of an audio communications channel. Perceptual audio evaluation methods were developed to enable evaluation of audio quality in channels of compromised performance, where conventional measurements such as THD+N may provide results that are not useful.

POLQA is the successor to PESQ. POLQA avoids certain weaknesses in PESQ, and is extended towards higher bandwidth audio signals.

Perceptual audio evaluation techniques are based on subjective experiments with listeners, and do not use simple periodic waveforms or sweeps as most APx measurements do; consequently, the perceptual evaluation results cannot be stated as physical values such as volts or hertz. Instead, listeners' opinions of quality (or algorithms designed to respond similarly) are averaged and scored.

PESQ and POLQA are both methods of generating MOS results without actually using a room full of listeners. Instead, pre-recorded speech samples are passed through a DUT or system and are compared with the original speech sample. An algorithm evaluates the difference between the reference audio and the degraded audio, and produces a result.

Channels of compromised audio quality

Many speech communications channels have limited audio capabilities, when compared to full fidelity channels optimized for music transmission. The limited capabilities of speech channels may provide other benefits: lower cost, lower bandwidth, lower power requirements, and so on.

Most of the I/O interfaces offered in an APx analyzer have very high fidelity capabilities, but in some cases the configuration for a low data rate DUT may make POLQA evaluations valuable. Passing a stimulus signal through a telephone network, for example, will likely degrade the signal substantially, as will low data rate codecs. The Bluetooth HSP and HFP profiles, in particular, are of moderate fidelity. Bluetooth manufacturers often use POLQA to characterize Bluetooth devices operating in these profiles.

MOS (Mean Opinion Score)

Perceptual audio evaluations are typically made with a number of listeners, who provide opinions about the quality of audio samples they have listened to. Typically, many audio samples, many listeners and many iterations are used, and the results are averaged into a MOS (Mean Opinion Score). MOS is expressed in a unitless scale from 1 to 5, representing the average opinion of quality, as shown in this table:

MOS	*Quality*	*Impairment*
5	Excellent	Imperceptible
4	Good	Perceptible but not annoying
3	Fair	Slightly annoying
2	Poor	Annoying
1	Bad	Very annoying

MOS results in APx are reported as MOS-LQO (LQO stands for Listening-only Quality, Objective). For POLQA results, the MOS-LQO results reference LC STANDARD IRS P.863.

POLQA ITU-T P.863 (2011)

The POLQA algorithm is defined in ITU-T Recommendation P.863 (2011).

POLQA is the successor of PESQ (ITU-T Rec. P.862). POLQA avoids weaknesses of the current P.862 model and is extended towards handling of higher bandwidth audio signals. Further improvements target the handling of time scaled signals and signals with many delay variations. Similarly to P.862,[2] POLQA supports measurements in the common telephony band (300–3400 Hz), but in addition it has a second operational mode for assessing HD-Voice in wideband and super-wideband speech signals (50–14000 Hz). POLQA also targets the assessment of speech signals recorded acoustically by an artificial head with mouth and ear simulators.

Active Speech Level (ASL) in POLQA

A key difference between POLQA and PESQ is in sensitivity to signal level. PESQ ignores level, but POLQA will produce poor MOS scores if the active speech level (ASL) presented to the algorithm is not matched (within a certain range) to the ASL of the reference file, which itself must be ASL-optimized for POLQA.

APx provides an Auto level alignment checkbox to defeat this level sensitivity. By default, Auto level alignment is ON, which makes good MOS scores easier to obtain. This is useful if your test requirements do not consider signal level, or if temporarily defeating level sensitivity is helpful in diagnosis. However, when Auto level alignment is ON, the APx POLQA measurement is not compliant with the ITU-T P.863 standard.

See Level alignment for POLQA, following, for information on adjusting levels to optimize MOS scores while remaining standards-compliant.

Speech samples

POLQA evaluates speech samples. You can record your own samples, or obtain speech sample files from other sources. The ITU has created a number of samples designed to cover the range of speech sounds uttered by men and women in a range of languages.

These are available at http://www.itu.int/net/itu-t/sigdb/genaudio/Pseries.htm and http://www.itu.int/rec/T-REC-P.501-200912-S/en . See the ITU P501E documentation at the second link for detailed information about creating and using speech samples.

For POLQA, APx500 provides one speech sample at a sample rate of 8 kHz, one at 16 kHz, and one at 48 kHz embedded in the measurement. These default speech samples have been optimized, with ASL set to –26 dB dBov using the ITU-T P.56 meter.

Averaging

Averaging multiple speech samples across multiple iterations provides the most reliable MOS results. APx500 provides POLQA (Averaged) measurements that enable you to average up to 64 speech samples or iterations.

Other results

The primary POLQA results in APx500 are the MOS (LC Standard IRS P.863) result and the MOS Average (LC Standard IRS P.863) result.

Other results are provided that are unsupported by ITU-T P.863. These results may be helpful in diagnostics and cause analysis in the DUT or system.

Level Alignment for POLQA

The POLQA algorithm is more critical than the PESQ algorithm in a number of ways. In particular, POLQA scores are sensitive to signal level where PESQ scores were not. For a good score in POLQA, the Active Speech Level (ASL) of the degraded signal must closely match that of the reference signal, which itself must be ASL-optimized for POLQA.

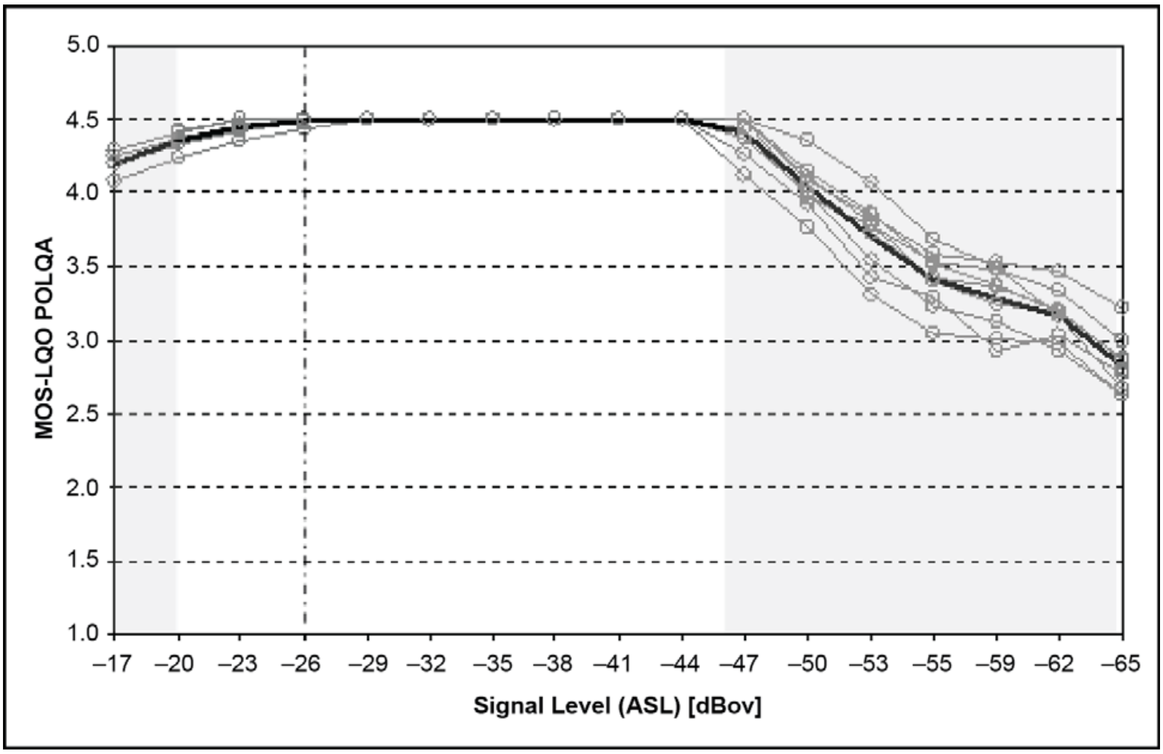
For standards-compliant POLQA MOS results in APx, the Auto level alignment checkbox must be OFF. In that case, you must optimize levels in your signal path so that the speech samples are at –26 dBov within the DUT or system under test, and so that the signal presented to the analyzer inputs is at the correct level to present an ASL of –26 dBov at the POLQA algorithm, after scaling in the analyzer input and converter stages.

–26 dBov ASL

The ideal ASL is –26 dBov. (dBov units are referenced to the level which causes overload in a system). Signals passed to the POLQA algorithm that have an ASL near –26 dBov will be graded with better MOS scores, as shown in the graph on the next page.

Speech files at –26 dBov ASL

The three default speech sample files provided by Audio Precision for POLQA have been optimized, with ASLs set to –26 dB dBov using the ITU-T P.56 meter. In digital loopback, a POLQA measurement using the Default 48 kHz file will score at MOS 4.75 (the maximum for 48 kS/s). In other I/O configurations, generator level settings and DUT or system gain will typically change the level. You can find information about the P.56 meter at http://www.itu.int/rec/T-REC-P.56/ .



Level differences between reference and degraded signals are allowed, but must be within the range of +5 dB above to –20 dB below the –26 dBov level, as shown here.

ASL within the DUT

Within the DUT, the DUT's gain should be set so that the test signal speech level is 26 dB below the DUT's overload level. The DUT's overload level may be known, or may require a series of gain and distortion measurements to discover.

ASL in acoustic paths

For systems that have an acoustic component, the acoustic path should have an ASL of 73 dBSPL (A).

APx analyzer analog input

The APx analyzer analog input sensitivity and the scaling between the input and the POLQA algorithm are such that a speech signal with typical rms values of about

- 30 mVrms to 50 mVrms (for 8 kS/s samples) or
- 50 mVrms to 130 mVrms (for 48 kS/s samples)

will appear at the algorithm at about –26 dBov. Some adjustment of the analog level provided to the analyzer input may be necessary to optimize the MOS.

Bypassing POLQA's level sensitivity

In APx, it is possible to bypass the level sensitivity for testing by setting the Auto level alignment checkbox ON (the default). This enables two simultaneous stages of compensation:

- APx automatically scales the signal so that it is applied to the algorithm at the optimal level.
- The POLQA algorithm implements its internal auto level mode, where both the reference signal and the degraded signal are forced to an ASL of –26 dBov.

Note: when the Auto level alignment checkbox is set to ON, the POLQA measurement is not standards-compliant.

68

POLQA (Averaged)
(Sequence Mode)

This measurement requires a software option key. See page 166 for more information about software options.

POLQA (Perceptual Objective Listening Quality Assessment) is a method of grading the quality of an audio communications channel, resulting in a MOS (Mean Opinion Score). POLQA is the successor to PESQ. See page 413 for more information about POLQA and PESQ.

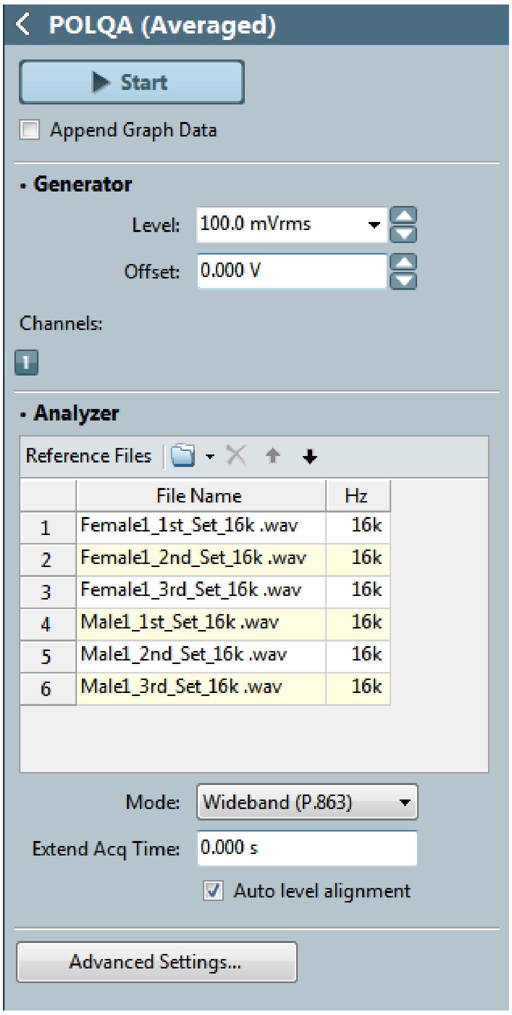
This measurement provides a means to average a number of POLQA measurements, producing a single MOS Average result per channel. Best practice recommends at least two male and two female voice samples, each averaged several times. Some testers will use many different voice samples, repeated and averaged many times.

The POLQA (Averaged) measurement uses a speech sample stimulus signal that is passed through the DUT and compared with the original speech signal serving as a reference. A MOS result and a number of diagnostic cause analysis results are provided.

The POLQA (Averaged) measurement is not available in External Source (open loop) configuration, or when Input Configuration is set to File.

The POLQA (Averaged) measurement results available in APx500 are:

- MOS Average (LC IRS P.863)
- File MOS (LC IRS P.863)



Operation

If you have not yet set up your test, first go to Chapter 6 “Signal Path Setup.”

Start

When Signal Generation and Signal Analysis (below) are set and you have imported all your reference files, click Start. The files will be played through the DUT, one at a time.

Append Graph Data

Normally, the current graph data for all results is deleted each time you start a new measurement. If the **Append** box is checked, the current data is kept in memory as a **Data Set** (see page 165), and the new measurement data is appended as a new data set. You can append many Data Sets.

Generator

Level

Set the generator output level in this field.

The Level control sets the level of the generator output to the value entered in the Level field, for a maximum level in the waveform file. See Chapter 14 for a general discussion of generator level when playing waveform files.

Generator Channels

The generator will output the speech samples to the DUT on the selected generator channels.

Reference Files

For a POLQA measurement, a reference file in the APx is played through the generator, and the output of the DUT or communications channels under test is compared to the reference file. For multiple averaging iterations with the same file or with different files, browse to enter the files in the grid here. These reference files will be played and analyzed in the order shown in the grid. The minimum number of files is 2; the maximum number of files is 64.

Audio Precision provides 48 speech samples for POLQA usage, installed as .WAV waveforms during the installation of APx500. Conforming waveforms are mono WAV files at a sample rate of 8 kHz, 16 kHz or 48 kHz. Content should be speech samples. Length is typically 5 to 10 seconds; maximum length is 12 seconds.

Use the folder icon to open a file browser. You can add files that are already in the project, or you can browse for files on disk. If you choose files of mixed sampling rates, the analysis Mode will automatically be set to the highest resolution mode found in the files. You can force the analysis mode to a lower resolution by making a selection in the Mode menu.

Note: the use a reference file whose sample rate differs from the target file is not standards compliant, and an error will be thrown. To suppress this error, set the Auto Resampling checkbox in Advanced Settings.

Select a file to delete it, or to move it up or down in the list using the arrow buttons.

Analyzer

Filters

No local low pass, high pass or weighting filters are available for this measurement. However, low pass, high pass and weighting filter settings made in Signal Path Setup > Input/Output will affect this measurement. Read more about APx Filters in Chapter 91.

As the generator moves through the Reference Files list, the analyzer acquires the signal and matches the audio received to the reference files, one at a time. The POLQA algorithm assigns a POLQA score, which is then mapped to a MOS score. A running average of the MOS scores is displayed, with the final value being the average of the scores of all the reference files.

Mode

Conforming POLQA speech sample files have an 8 kHz sample rate for testing Narrowband (P.863) channels, a 16 kHz sample rate for testing Wideband (P.863) channels, and a 48 kHz sample rate for testing Super-Wideband (P.863) channels. The mode is selected automatically by the POLQA measurement based on the sample rate contained in the reference file.

Extend Acq. Time:

The acquisition time is set to the length of the speech sample (selected either in Waveform, for closed loop, or Reference Signal for External Source (open loop), plus the time set in this field. If you suspect you are getting poor results because the acquisition length is truncating the input signal, add time here.

Auto level alignment

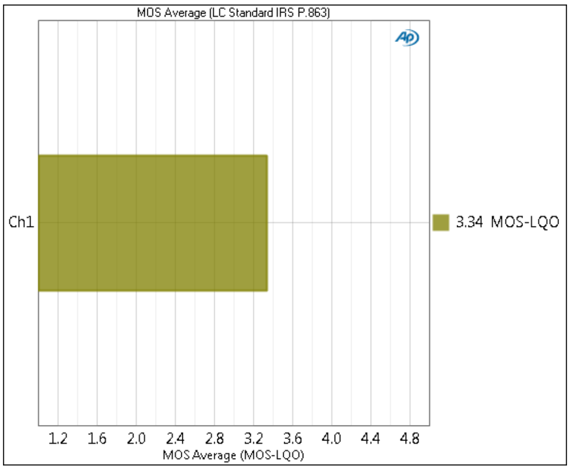
When this checkbox is checked (the default), APx realigns the level of the acquired sample to closely match the level of the signal in the reference file, making MOS results more stable. The APx level alignment does not affect standards compliance.

In rare cases, a user may want the POLQA algorithm to be sensitive to level misalignment, and Auto Level Alignment can be unchecked. When this checkbox is unchecked, no level adjustment will be made and MOS results are more likely to vary.

Advanced Settings

If you'd like to adjust analyzer ranging parameters, click Advanced Settings, where you will also find POLQA-specific settings for Auto Resampling. See Advanced Settings on page 413.

POLQA (Averaged): MOS Average result



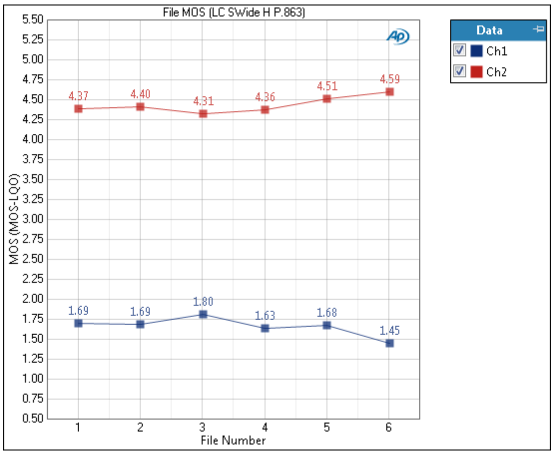
The **POLQA: MOS Average** result is the primary result for the POLQA (Averaged) measurement. The POLQA score is mapped to MOS-LQO using LC Standard IRS P.863. This result provides an average of all the MOS scores of the speech samples in the Reference Files list.

Units

Units for POLQA (Averaged): MOS Average are:

- MOS-LQO

POLQA (Averaged): File MOS result



The POLQA (Averaged): File MOS result is useful for cause analysis. The result shows the MOS score for each of the files in the Reference Files list.

By default, the graph display for this result has Show Markers: Y Labels enabled on the Graph Properties dialog. See page 570.

Units

Units for POLQA (Averaged): File MOS are:

X-Axis

- File number

Y-Axis

- MOS-LQO

69

PSR
(Sequence Mode)

Introduction

This measurement requires the PDM Option. See page 9 and Chapter 17 for more information.

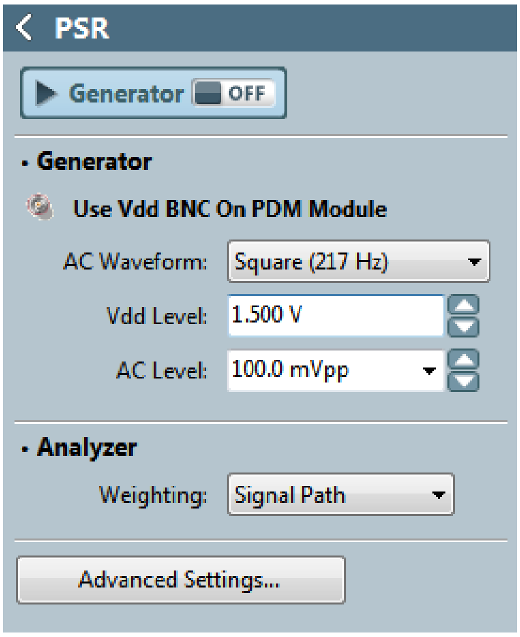
Overview

PSR is a power supply rejection measurement typically used in DUTs that are PDM transmitters. A small AC signal is imposed on the PDM Vdd power supply, and the residual of that signal is measured in the DUT output. PSR (or PSRR) can also be measured in other powered DUTs, such as converter or opamp devices, as long as the Vdd voltage and current limits of the PDM Vdd output are sufficient for the DUT. For DUTs that require more current, a unity gain DC coupled amplifier can be inserted between the APx generator and the DUT.

Unique Characteristics

The PSR and PSR Frequency Sweep measurements have some characteristics unique in APx500:

- Availability
  The PSR measurements are only available if the PDM Option is installed.
- Vdd Supply
  When the **Generator** in a PSR measurement is turned **On**, DC power is applied to the PDM **Vdd Supply** connector, regardless of the current Signal Path settings. When the **Generator** is turned **Off**, DC power at the PDM **Vdd Supply** connector reverts to the Signal Path setting.
- APx Generator
  For a PSR measurement, the output of the APx **Generator** is added to the DC on the **Vdd Supply** BNC power connector, and does not appear at any other output. The analyzer output connector as specified in Signal Path Setup is irrelevant. The generator signal is available to the PSR measurement, even if the **Output Connector** is set to **None**.



There are four results for PSR:

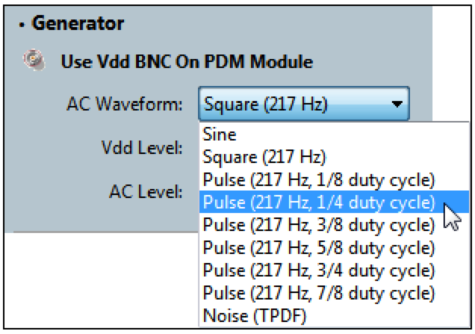
- PSR
- PSR+N
- PSRR (analog inputs only)
- PSRR+N (analog inputs only)

Generator

When this measurement is selected, the PDM **Vdd Supply** is switched to **Off**, if currently **On**.

Click the **Generator** button to toggle the **Generator** to **On**. The signal selected in **AC Waveform** is added to the DC available at the PDM **Vdd Supply** connector, and the **Vdd Supply** is switched to **On**.

AC Waveform



Choose the waveform to be added to the **Vdd Supply** voltage.

- Sine
- Square (217 Hz)
- Pulse (217 Hz, 1/8 duty cycle)
- Pulse (217 Hz, 1/4 duty cycle)
- Pulse (217 Hz, 3/8 duty cycle)
- Pulse (217 Hz, 5/8 duty cycle)
- Pulse (217 Hz, 3/4 duty cycle)
- Pulse (217 Hz, 7/8 duty cycle)
- Noise (TPDF)

The square wave and pulse choices simulate the digitally noisy environment of a GSM mobile phone. 216.667 Hz is approximately the frequency of the GSM frame rate, and the pulses simulate the noise with a variable number of time slots used, from 1 to 8.

The noise signal is a pseudo-random noise waveform having a triangular probability density function and a peak level set by the user.

Vdd Level

Set the (unmodulated) Vdd DC level here. This is the level before the AC signal (above) is added.

AC Level

Set the level of the AC level here.

Frequency

This field is available if **AC Waveform** is set to **Sine**. Set the frequency of the sine wave here.

Analyzer

Weighting Filter

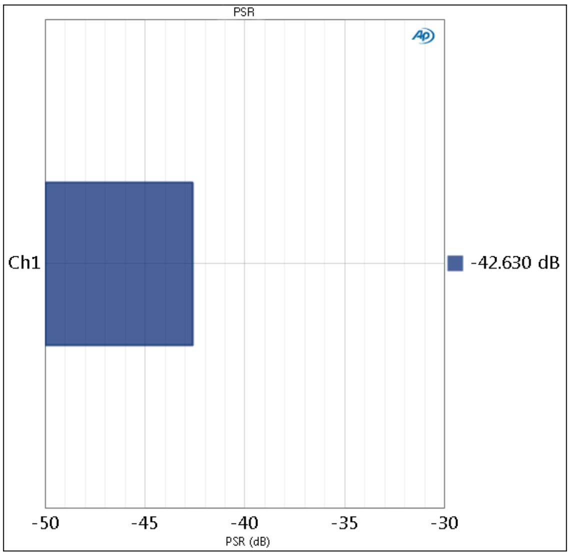
A local weighting filter is available for this measurement. See Measurement Filters on page 547 for detailed information about the filters available locally for some measurements.

Low pass, high pass and weighting Signal Path Filters settings (see page 545) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass or high-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active. A local weighting filter setting does not override any Signal Path filter settings; the local weighting filter is applied in addition to any active Signal Path filters.

Advanced Settings

If your test required special adjustments or settings, click **Advanced Settings**. See page 458. See Chapter 98 for more information about units of measurement.

PSR: PSR Result



When **AC Waveform** is set to **Sine**, the PSR acquisition is filtered first by a weighting filter, if applied, then by a window width bandpass filter at the sine frequency. The PSR is the RMS level inside the bandpass filter.

When **AC Waveform** is set to **Square (217 Hz)**, the PSR acquisition is filtered first by a weighting filter, if applied, then by window width bandpass filters at the fundamental (217 Hz) and at the odd harmonics of 217 Hz, to the 101st harmonic. The PSR is the RMS sum of the levels inside the bandpass filters.

When **AC Waveform** is set to one of the **Pulse** choices, the PSR acquisition is filtered first by a weighting filter, if applied, then by window width bandpass filters at the fundamental (217 Hz) and at the even and odd harmonics of 217 Hz, to the 101st harmonic.

When **AC Waveform** is set to **Noise**, the PSR acquisition is filtered by a weighting filter, if applied. The PSR is the RMS level of the broadband result. For Noise, the PSR and PSR+N results are the same.

Note: the actual square wave frequency is 216.667 Hz. It is displayed nominally as 217 Hz.

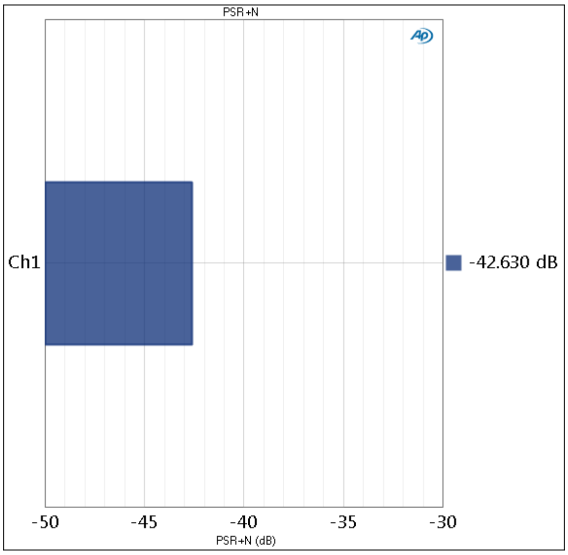
The use of window-width filters provides a low-noise estimate of the PSR, since the power outside the filters is rejected. This enables PSR measurements to be made on microphones which are not in an acoustically quiet environment.

Units

Units available for PSR results are

Analog Signals	Digital Signals
• Vrms	• dBFS
• dBV	• FS
• dBu	• %FS
• dBrA	• dBrA
• dBrB	• dBrB
• dBSPL1	• dBSPL1
• dBSPL2	• dBSPL2
• dBm	
• W (watts)	

PSR: PSR+N Result



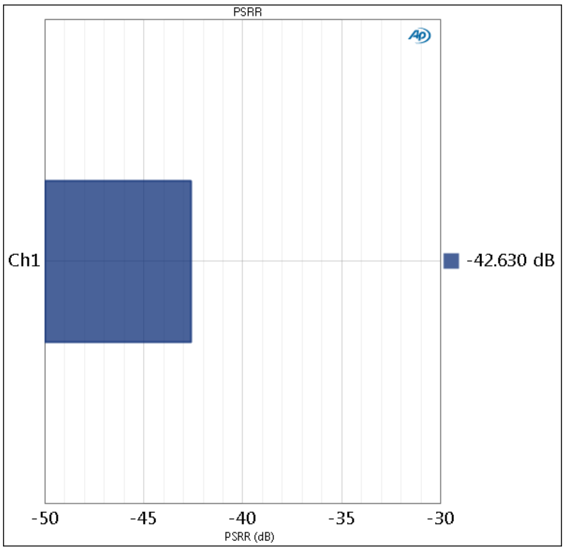
The PSR+N (Power Supply Rejection plus Noise) acquisition is filtered by a weighting filter, if applied. The PSR+N is the RMS level of the broadband result. For the Noise waveform, the PSR and PSR+N results are the same.

Units

Units available for PSR+N results are

Analog Signals	Digital Signals
• Vrms	• dBFS
• dBV	• FS
• dBu	• %FS
• dBrA	• dBrA
• dBrB	• dBrB
• dBSPL1	• dBSPL1
• dBSPL2	• dBSPL2
• dBm	
• W (watts)	

PSR: PSRR Result



PSRR is Power Supply Rejection Ratio. This nomenclature is used to describe the ratio of the level of a residual power supply perturbation found in the output signal, to the perturbation in the power supply. In APx, this result is only available when the analyzer input is analog.

When AC Waveform is set to Sine, the PSRR acquisition is filtered first by a weighting filter, if applied, then by a window width bandpass filter at the sine frequency. The PSRR is the ratio of the RMS level inside the bandpass filter to the RMS level of the AC Waveform.

When AC Waveform is set to Square (217 Hz), the PSRR acquisition is filtered first by a weighting filter, if applied, then by window width bandpass filters at the fundamental (217 Hz) and at the odd harmonics of 217 Hz, to the 101st harmonic. The PSRR is the ratio of the RMS sum of the levels inside the bandpass filters to the RMS level of the AC Waveform.

When AC Waveform is set to one of the Pulse choices, the PSRR acquisition is filtered first by a weighting filter, if applied, then by window width bandpass filters at the fundamental (217 Hz) and at the even and odd harmonics of 217 Hz, to the 101st harmonic. The PSRR is the ratio of the RMS sum of the levels inside the bandpass filters to the RMS level of the AC Waveform.

When **AC Waveform** is set to **Noise**, the PSRR acquisition is filtered by a weighting filter, if applied. The PSRR is the ratio of the RMS level of the broadband result to the RMS level of the AC Waveform. For Noise, the PSRR and PSRR+N results are the same.

Note: the actual square wave frequency is 216.667 Hz. It is displayed nominally as 217 Hz.

The use of window-width filters provides a low-noise estimate of the PSRR, since the power outside the filters is rejected. This enables PSRR measurements to be made on microphones which are not in an acoustically quiet environment.

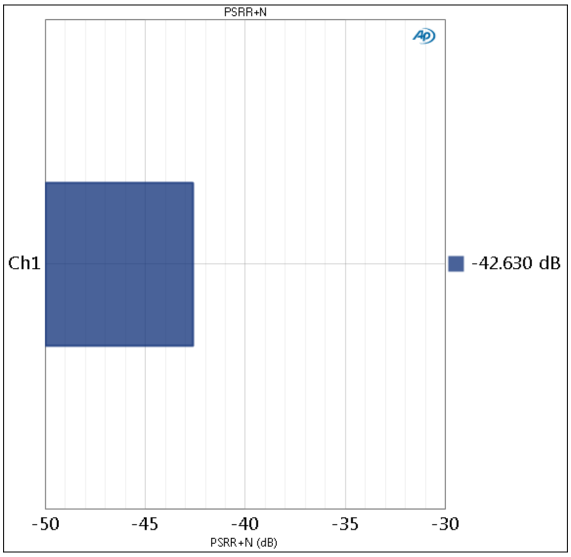
Units

Units available for PSRR results are

- x/y
- %
- ppm
- dB

Read more about Units in Chapter 98.

PSR: PSRR+N Result



PSRR is Power Supply Rejection Ratio plus Noise. This nomenclature is used to describe the ratio of the level of a residual power supply perturbation plus noise found in the output signal, to the AC signal that is the source of perturbation in the power supply. In APx, this result is only available when the analyzer input is analog.

The PSRR+N (Power Supply Rejection Ratio plus Noise) acquisition is filtered by a weighting filter, if applied. The PSRR+N is the ratio of the RMS level of the broadband result to the RMS level of the AC Waveform. For Noise, the PSRR and PSRR+N results are the same.

Units

Units available for PSRR+N results are

- x/y
- %
- ppm
- dB

70

PSR Frequency Sweep
(Sequence Mode)

Introduction

This measurement requires the PDM Option. See page 9 and Chapter 17 for more information.

Overview

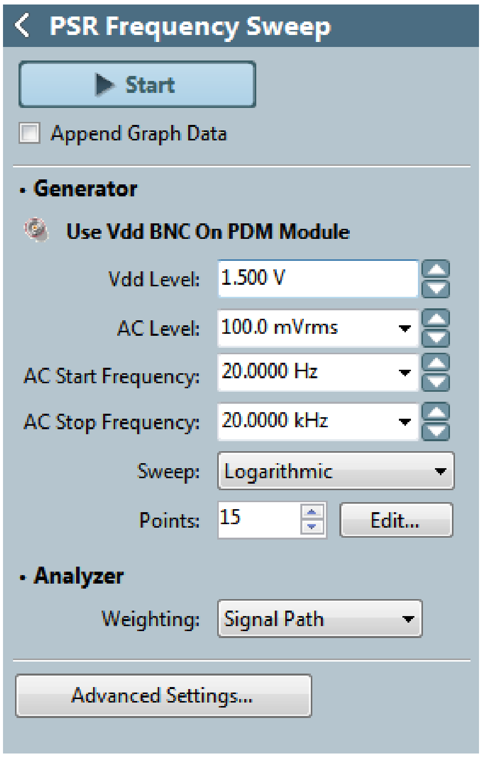
PSR is a power supply rejection measurement typically used in DUTs that are PDM transmitters. A small AC signal is imposed on the PDM Vdd power supply, and the residual of that signal is measured in the DUT output. PSR (or PSRR) can also be measured in other powered DUTs, such as converter or opamp devices, as long as the Vdd voltage and current limits of the PDM Vdd output are sufficient for the DUT.

For PSR Frequency Sweep, the AC signal is a sine wave, swept across a range of frequencies.

Unique Characteristics

The PSR and PSR Frequency Sweep measurements have some characteristics unique in APx500:

- Availability
  The PSR measurements are only available if the PDM Option is installed.
- Vdd Supply
  When the **Generator** in a PSR measurement is turned **On**, DC power is applied to the PDM **Vdd Supply** connector, regardless of the current Signal Path settings. When the **Generator** is turned **Off**, DC power at the PDM **Vdd Supply** connector reverts to the Signal Path setting.
- APx Generator
  For a PSR measurement, the output of the APx **Generator** is added to the DC on the **Vdd Supply** BNC power connector, and does not appear at any other output. The analyzer output connector as specified in Signal Path Setup is irrelevant. The generator signal is available to the PSR measurement, even if the **Output Connector** is set to **None**.



There are four results for PSR Frequency Sweep:

- PSR
- PSR+N
- PSRR (analog inputs only)
- PSRR+N (analog inputs only)

When this measurement is selected, the PDM **Vdd Supply** is switched to **Off**, if currently **On**.

Start

For a default sweep of 20 kHz to 20 Hz in 31 logarithmic steps, simply click **Start**. If you want to specify other parameters for your sweep, review **Signal Gener-**

ation and Signal Acquisition and Analysis settings, discussed below. Click Start when you have approved the settings.

Append Graph Data

Normally, the current graph data for all results is deleted each time you start a new measurement. If the Append box is checked, the current data is kept in memory as a Data Set (see page 165), and the new measurement data is appended as a new data set. You can append many Data Sets.

Generator

When this measurement is selected, the PDM Vdd Supply is switched to Off, if currently On.

Vdd Level

Set the (unmodulated) Vdd DC level here. This is the level before the AC signal (above) is added.

AC Level

Set the level of the AC level here.

AC Start Frequency

Set the sweep start frequency here.

AC Stop Frequency

Set the sweep stop frequency here.

Sweep

Choose

- Logarithmic point spacing
- Linear point spacing
- Custom
  Edit sweep spacing and number of Points to create a Custom sweep.

Points

Set the number of sweep points here.

Points > Edit

Open the Sweep Points dialog to edit, import or export the Sweep Points table.

Step Size

For linear sweeps, enter the Step Size here.

Level

Set the signal level here.

Analyzer

Filters

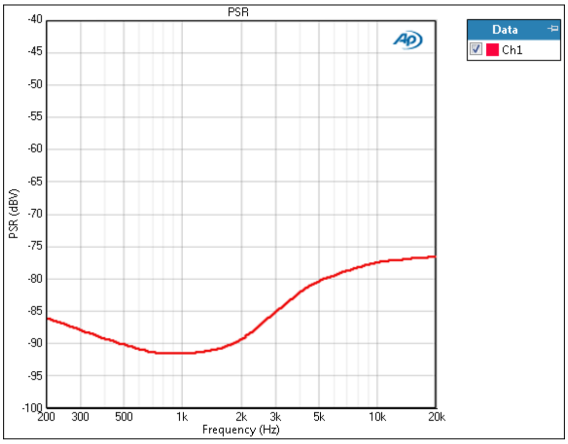
A local weighting filter is available for this measurement. See Measurement Filters on page 547 for detailed information about the filters available locally for some measurements.

Low pass, high pass and weighting Signal Path Filters settings (see page 545) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass or high-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active. A local weighting filter setting does not override any Signal Path filter settings; the local weighting filter is applied in addition to any active Signal Path filters.

Advanced Settings

If your test required special adjustments or settings, click Advanced Settings. Read more about Units in Chapter 98.

PSR Frequency Sweep: PSR Result



The PSR acquisition is filtered first by a weighting filter, if applied, then by a window-width bandpass filter that is swept at the sine frequency. The PSR is the RMS level inside the bandpass filter, plotted against frequency.

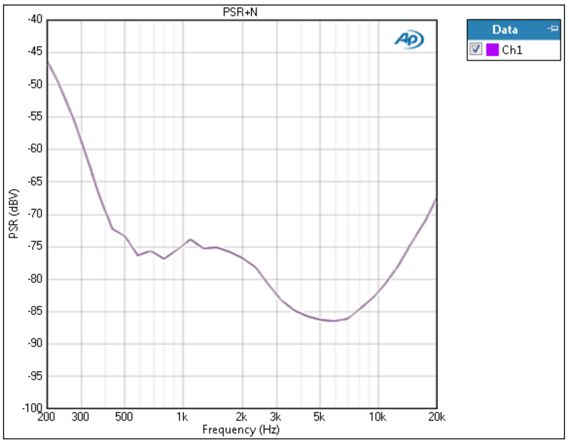
The use of window-width filters provides a low-noise estimate of the PSR, since the power outside the filters is rejected. This enables PSR measurements to be made on microphones which are not in an acoustically quiet environment.

Units

Units available for PSR results are

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz	• Vrms	• dBFS
• dHz	• dBV	• FS
• F/R	• dBu	• %FS
• %Hz	• dBrA	• dBrA
	• dBrB	• dBrB
	• dBSPL1	• dBSPL1
	• dBSPL2	• dBSPL2
	• dBm	
	• W (watts)	

PSR Frequency Sweep: PSR+N Result



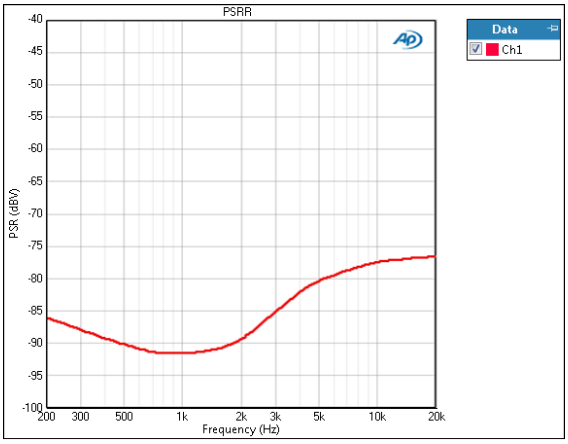
The PSR+N (Power Supply Rejection plus Noise) acquisition is filtered by a weighting filter, if applied. The PSR+N is the RMS level of the broadband result, plotted against frequency.

Units

Units available for PSR results are

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz	• Vrms	• dBFS
• dHz	• dBV	• FS
• F/R	• dBu	• %FS
• %Hz	• dBrA	• dBrA
	• dBrB	• dBrB
	• dBSPL1	• dBSPL1
	• dBSPL2	• dBSPL2
	• dBm	
	• W (watts)	

PSR Frequency Sweep: PSRR Result



PSRR is Power Supply Rejection Ratio. This nomenclature is used to describe the ratio of the level of a residual power supply perturbation found in the output signal, to the perturbation in the power supply. In APx, this result is only available when the analyzer input is analog.

The PSRR acquisition is filtered first by a weighting filter, if applied, then by a window-width bandpass filter that is swept at the sine frequency. The PSRR is the ratio of the RMS level inside the bandpass filter to the RMS level of the perturbation in the power supply, plotted against frequency.

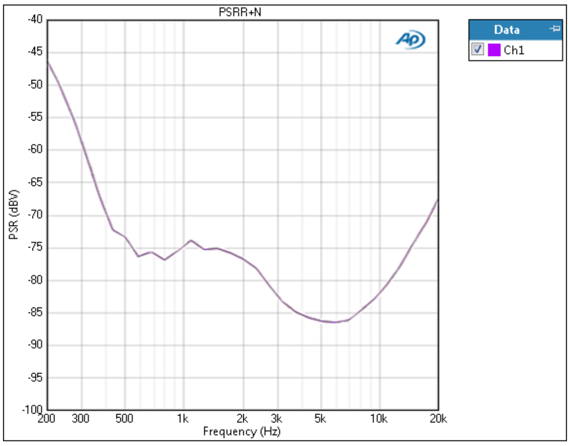
The use of window-width filters provides a low-noise estimate of the PSRR, since the power outside the filters is rejected. This enables PSRR measurements to be made on microphones which are not in an acoustically quiet environment.

Units

Units available for PSRR results are

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

PSR Frequency Sweep: PSRR+N Result



PSRR+N is Power Supply Rejection Ratio plus Noise. This nomenclature is used to describe the ratio of the level of a residual power supply perturbation plus noise found in the output signal, to the AC signal that is the source of perturbation in the power supply. In APx, this result is only available when the analyzer input is analog.

The PSRR+N (Power Supply Rejection Ratio plus Noise) acquisition is filtered by a weighting filter, if applied. The PSRR+N is ratio of the RMS level of the broadband result to the AC signal that is the source of the perturbation in the power supply, plotted against frequency.

Units

Units available for PSRR+N results are

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

71

Regulated Frequency Sweep (Sequence Mode)

Introduction

For the APx515, this measurement requires a software option key. See page 166 for more information about software options.

Regulated Frequency Sweeps use a sine wave stimulus signal that is moved across a range of frequencies in a specified number of points. At each point, the generator level is adjusted to the level that produces the target value in the selected channel. Go to page 433 for more information about regulation.

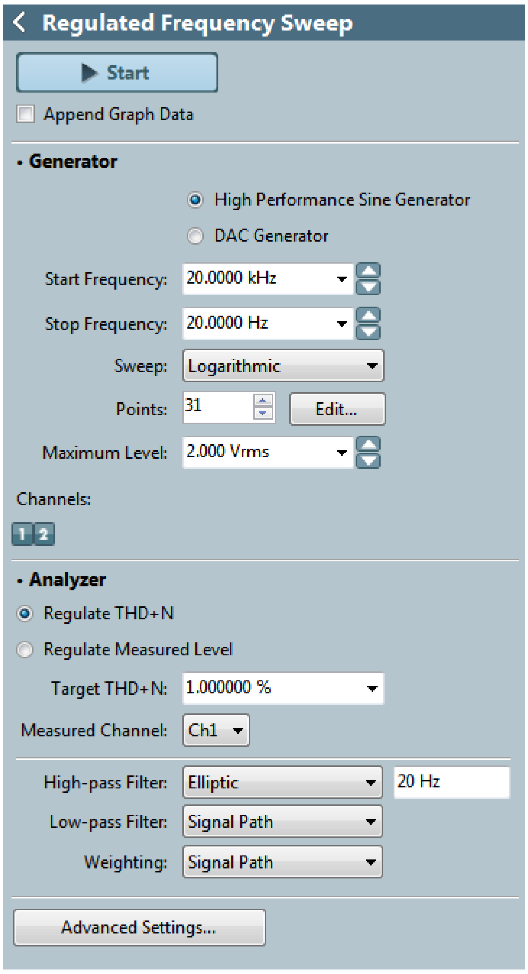
Regulated Frequency Sweeps are not available in External Source configuration.

Regulating THD+N is often employed to plot an amplifier frequency sweep at the amplifier's maximum output level.

Regulated Frequency Sweep results available in APx500 are:

- Level (frequency response)
- THD+N Level
- THD+N Ratio
- Generator Level

All of these results are available from a single measurement.



Operation

If you have not yet set up your test, first go to Signal Path Setup (Chapter 6).

Append Graph Data

Measured data are grouped in a **Data Set**, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 573.

Generator

Regulated Frequency Sweep only supports a sine waveform in a closed-loop configuration.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 49 for a comparison of features.

To make a regulated frequency sweep measurement, first define the sweep by selecting a sweep **Start Frequency**, **Stop Frequency**, the number of sweep points and **Sweep** type. Then set a generator **Maximum Level** to protect your DUT.

Choose **Regulate THD+N** or **Regulate Measured Level**.

For **Regulate THD+N** (non-linear) regulation, select a **Measured Channel** and choose a **Target Distortion** for generator regulation.

For **Regulate Measured Level**, choose a **Target Level**. Each channel will be individually regulated to this value.

Click **Start**. The generator will output a sine wave to the DUT on the selected generator channels at the frequency set in the **Generator** panel. At the first sweep point the software regulates the generator level to the **Target** value. When the result for the first point is acquired, the sweep proceeds to the next points, regulating the generator at each point until the sweep is completed.

Start, Stop and Points for a stepped frequency sweep

The sweep begins at the **Start Frequency** and ends at the **Stop Frequency**. By default, these are set at 20 kHz and 20 Hz, and the sweep moves in logarithmic step points from high frequency to low.

You can enter **Start** and **Stop** frequencies of your choice, and sweep from low to high if you want. The sweep moves in a specified number of points, set in the Points field. The minimum is 2 points; maximum is 65,535. The default setting is 31, which corresponds to 1/3 octave per point in a logarithmic sweep across 20 Hz to 20 kHz. By default, the sweep is logarithmic. You can also select a linear sweep, in which case the **Step Size** filed becomes available.

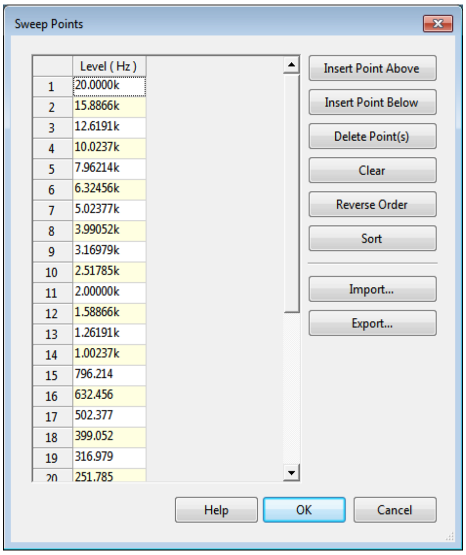
The sweep point spacing is set by selecting one of the following choices in the Sweep field:

- **Logarithmic** (the default); use the **Points** field to set the number of logarithmically spaced points;
- **Linear**, which provides two methods of adjusting spacing: the **Points** field or the **Step Size** field; or
- **Custom**. Click **Edit** to open the **Sweep Points** dialog, where you can set points arbitrarily, or load or save sweep table files.

Viewing or Editing the Sweep Points table

Click Edit to open the Sweep Points table. This table shows each point and its corresponding frequency. You can edit this table to add or delete points or to change the frequency of a point. Points can be sorted or reversed in order using the controls on the right.

A Sweep Points table can be saved as a *.csv file or as a Microsoft Excel *.xls file. A compatible *.csv or *.xls file can be opened and used as a Sweep Points table.



Analyzer

Regulation method

- Regulate THD+N
  For **Regulate THD+N**, choose a **Target THD+N** value and a **Measured Channel**.
- Regulate Measured Level
  For **Regulate Measured Level**, choose a **Target Level** value and a **Measured Channel** (or **All Channels**).

See More About Regulation on page 433 for a detailed discussion of these methods.

Filters

Local high-pass, low-pass and weighting filters are available for this measurement. The low-pass and high-pass filters, if selected, are applied to the entire measured signal and affect all the results. The weighting filter, if selected, affects only the distortion and distortion plus noise results. See Measurement Fil-

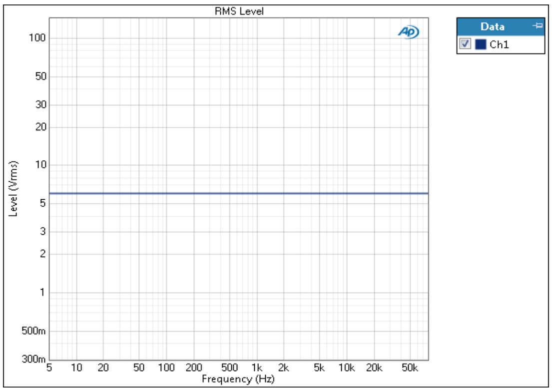
ters on page 547 for detailed information about the filters available locally for some measurements.

Low pass, high pass and weighting Signal Path Filters settings (see page 545) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass or high-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active. A local weighting filter setting does not override any Signal Path filter settings; the local weighting filter is applied in addition to any active Signal Path filters.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, see Advanced Settings for Stepped Sweeps on page 458. See Chapter 98 for more information about units of measurement.

Regulated Frequency Sweep: Level



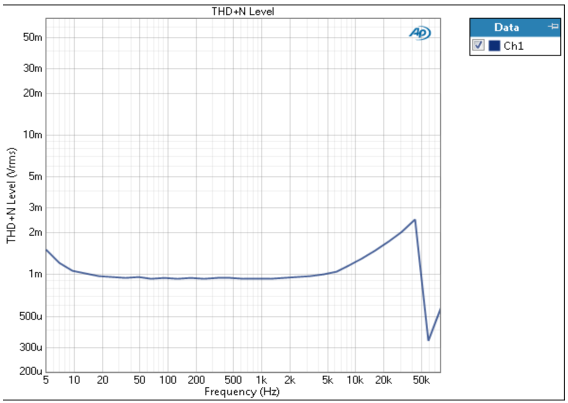
The Level result plots the DUT measured output level on the Y axis, against the generator frequency on the X axis.

Units

Units available for Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz	• Vrms	• dBFS
• dHz	• dBV	• FS
• F/R	• dBu	• %FS
• %Hz	• dBrA	• dBrA
	• dBrB	• dBrB
	• dBSPL1	• dBSPL1
	• dBSPL2	• dBSPL2
	• dBm	
	• W (watts)	

Regulated Frequency Sweep: THD+N Level



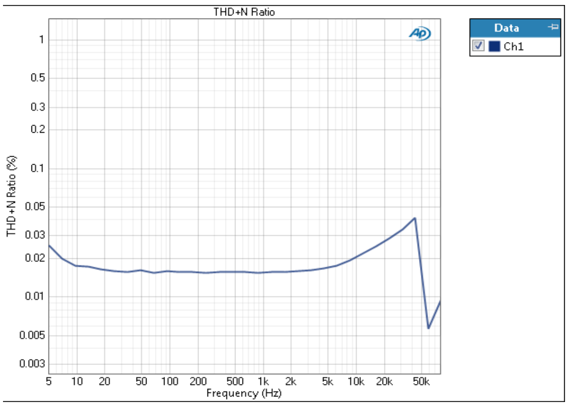
The THD+N Level result plots the DUT measured THD+N output level on the Y axis, against the generator frequency on the X axis. This method is often employed to plot an amplifier frequency sweep at the amplifier's maximum output level.

Units

Units available for THD+N Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz	• Vrms	• dBFS
• dHz	• dBV	• FS
• F/R	• dBu	• %FS
• %Hz	• dBrA	• dBrA
	• dBrB	• dBrB
	• dBSPL1	• dBSPL1
	• dBSPL2	• dBSPL2
	• dBm	
	• W (watts)	

Regulated Frequency Sweep: THD+N Ratio



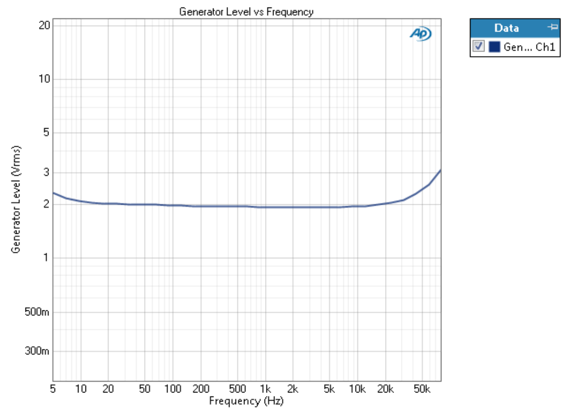
The THD+N Ratio result plots the DUT measured THD+N output Ratio on the Y axis, against the generator frequency on the X axis. This method is often employed to plot an amplifier frequency sweep at the amplifier's maximum output level.

Units

Units available for THD+N Ratio are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Regulated Frequency Sweep: Generator Level



The Level result plots the generator output level on the Y axis, against the generator frequency on the X axis.

Units

Units available for Generator Level results are:

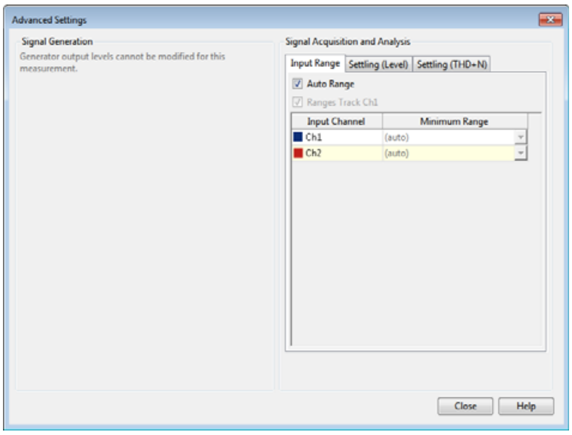
X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz	• Vrms	• FS
• dHz	• Vp	• %FS
• F/R	• Vpp	• dBFS
• %Hz	• dBV	• dBrG
	• dBu	
	• dBrG	
	• dBm	
	• W (watts)	

Y-axis (acoustic output configuration)

- Pa
- dBSPL

Advanced Settings for Regulated Measurements

The default settings here are appropriate for most measurements. You may want to make minor adjustments for special situations.



Signal Generation

There are no advanced settings for signal generation for regulated sweep measurements.

Signal Acquisition and Analysis

Input Range tab

Digital inputs do not require ranging, and the Input Range tab is unavailable when inputs are set to a digital format.

If Auto Range is checked (the default), each analog channel determines its input range automatically, based on the level of the input signal. If the input signal level changes beyond a ranging threshold, Auto Range will cause the input ranging circuits to move up or down for proper ranging.

See Chapter 92 for more information about ranging.

If Auto Range is unchecked, you can set a fixed range for each analog input channel. If Track first channel range is checked (the default), the fixed input range setting for channel 1 is copied to the other channels, and range settings for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels.

To set individual analog input channel ranges, uncheck the “Track first channel range” checkbox and enter values in the input channel Range fields.

Settling tab

Two Settling tabs are available for regulated measurements, Settling (Level) and Settling (THD+N).

See Chapter 92 for more information about Settling for regulated measurements.

More about regulation and maximum operating level

Regulation in Audio Precision analyzers refers to automatic control of one parameter by another. APx500 uses regulation to automatically adjust the generator level so that a specified target value is produced at the DUT output. Target values can be the measured DUT output THD+N ratio (non-linear regulation) or the measured DUT output level (linear regulation).

Measurements using Regulation in APx500

The Maximum Output and Maximum Output (CEA-2006) measurement results use regulation to a target THD+N ratio.

The Auto Gen Level feature in Reference Levels and the Regulated Frequency Sweep measurement use regulation to either a target THD+N ratio, or to a target level.

Regulating to a target THD+N ratio (non-linear regulation)

For audio testing, it is often desirable to perform certain tests just below the clipping point of the DUT. As the level rises, a harmonic distortion value exceeding 1.0 % is commonly agreed to indicate the onset of soft clipping, establishing a device’s maximum operating level, or MOL.

The amount of distortion in the output signal does not follow the generator level linearly, requiring more iterations to satisfy the regulation algorithm. For practical considerations, the regulation algorithm operates on one selected Measured Channel.

APx500 non-linear regulation algorithm

The generator is turned on at a certain level (see Generator Starting Level, below). A measurement is taken. If the initial result is below the target, the generator level is increased by a factor Δ_{gen}, and another measurement is taken; if the initial result is above the target, the generator level is decreased by Δ_{gen} for the next measurement. When the measured result crosses the target value, Δ_{gen} is halved and its sign is changed, so that for the next measurement the generator level moves in the opposite direction by a smaller amount. When Δ_{gen} becomes small enough that the change in the generator level would be smaller than the value set in the Generator Tolerance field, regulation ends.

Generator starting level for non-linear regulation

The starting level of the generator depends on the context:

In the Auto Gen Level function in Reference Levels, the generator starts at its current level, displayed in the Level field.

In the other measurement results, the generator starts at half the value set in the Maximum Level field.

The first point in a Regulated Frequency Sweep follows rule #2, starting at half the value set of Maximum Level. For each subsequent point, the starting generator level is set to the final regulated generator level for the previous point.

Regulating to a target level (linear regulation)

It is often desirable to make an audio measurement at a specific DUT output level (1 dBu, for example), regardless of the gain or loss of the DUT channel. Within the operating range of the DUT, the output level follows the generator level linearly, requiring only one or two iterations of to satisfy the regulation algorithm. Linear regulation can operates on all channels or on a selected channel.

APx500 linear regulation algorithm

1. The generator is turned on at the level specified in the Level field in the Reference Levels measurement. A measurement is taken. The gain of the DUT is inferred from the measured level.

2. The generator level is set as the Target Level divided by the DUT gain. If this level is greater than the Maximum Level, the process aborts with an error.

3. If the resulting measured level is not within 1% of the specified target level, steps 1 and 2 are performed again using the current generator level.

4. If the resulting measured level is still not within 1% of the specified target level, the process aborts with an error.

Maximum Level setting

A maximum level is set to protect the DUT. If the generator reaches the value set in Maximum Level, or the minimum level (fixed internally), regulation ends and a failure message is displayed.

Troubleshooting Regulation Issues

Regulation can fail under three common conditions:

1. If the Maximum Level setting (set to protect the DUT from overload) is set too low for a particular DUT, the target value will not be achieved and regulation will fail.

   Solution: set the Maximum Level higher.

2. If cables between instrument and DUT are accidentally crossed, the regulation will fail. This is particularly interesting for the “Regulate Measured Level” setting in the Regulated Frequency Sweep, where crossed cables can cause regulation to simultaneously drive one channel up and another channel down.

   Solution: uncross the cables.

3. For regulation to measured THD+N Ratio, if the initial generator level is set too low, the regulation algorithm may direct the generator to a lower level rather than a higher level, and regulation will fail. Explanation: If the measured distortion is below the target distortion, the algorithm responds by increasing the generator level. If the distortion is above the target, the algorithm decreases the level.
   For example, a target distortion of 1% is equivalent to
   –40 dB. If the initial generator level is at least 40 dB above the noise floor, the measured THD+N will be less than 1% and regulation will correctly proceed by increasing the generator level. If the initial generator level is less than 40 dB above the noise floor, the measured THD+N will be more than the target (again, using 1% as an example) and regulation will attempt to reduce the distortion by reducing the generator level. This moves the measurement closer to the noise floor, and the THD+N ratio will increase. This process repeats, ultimately resulting in a failure of regulation.

   Solution: set the initial generator level higher. As explained above, in Auto Gen Level initial level is directly set; in the other regulation features initial level is one-half of the Maximum Level value.

4. Regulation my not be possible if the DUT is characterized by high distortion, high noise, or level instability.

72

Resistance (DCX)
(Sequence Mode)

DCX measurements require that a DCX-127 Multifunction Module be connected to the APx analyzer system, typically using an Audio Precision USB/APIB adapter connected to the PC.

The DCX-127 brings additional measurement and control features to an Audio Precision analyzer. The APx Resistance (DCX) measurement uses the DCX DMM (Digital Multimeter) mode to implement a precise ohmmeter. Download the DCX-127 Multifunction Module User's Guide from the Audio Precision Web site at ap.com for complete operational information for the DCX-127.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 6 for information about Signal Path Setup.

The DCX-127 must be powered on and connected to the measurement PC using an Audio Precision USB/APIB adapter.

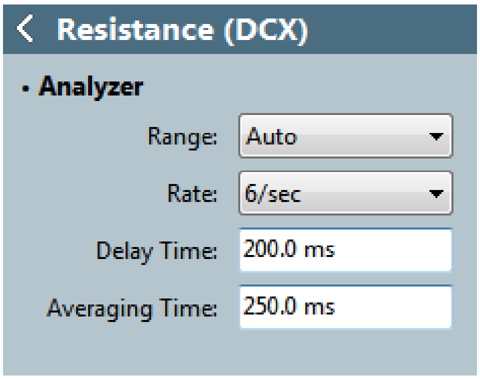
Selecting the Resistance (DCX) measurement places the DCX in "DMM" mode. Note that you can configure the DCX DC Outputs or Auxiliary outputs in Signal Path Setup > DCX, independent of any settings or operation in the Resistance (DCX) measurement.

Generator

No Generator functions are provided for the Resistance (DCX) measurement.

Analyzer

The DCX has a digital multimeter (DMM) that is configured as an ohmmeter for this measurement. The APx analyzer is not used; all the data comes from the DCX.



Range

Select a fixed range for the Resistance measurement, or select Auto.

- Auto
- 2 MOhm
- 200 kOhm
- 20 kOhm
- 2 kOhm
- 200 Ohm

Rate

The Rate control allows you to select one of two meter reading rates.

- 6/sec
  Approximately 6 meter readings are taken per second.
- 25/sec
  Approximately 25 meter readings are taken per second.

At 6/sec, DCX meter resolution is a full 4 1/2 digits. At 25/sec, the resolution is reduced. The number of digits displayed is the same but the least significant digit is always 5 or 0.

Delay Time

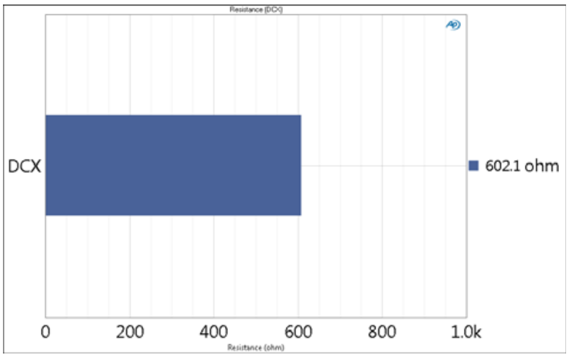
A Delay Time interval is started when the Resistance (DCX) measurement is started within a sequence. APx ignores DCX data until the time set in Delay Time has passed. Minimum Delay Time is 200 ms, maximum is 10 s, 200 ms is the default.

Averaging Time

APx performs a rolling average of the DCX readings across a specified time before display. Minimum Averaging Time is 200 ms, maximum is 10 s, 250 ms is the default.

See Chapter 98 for more information about units of measurement.

Resistance (DCX)



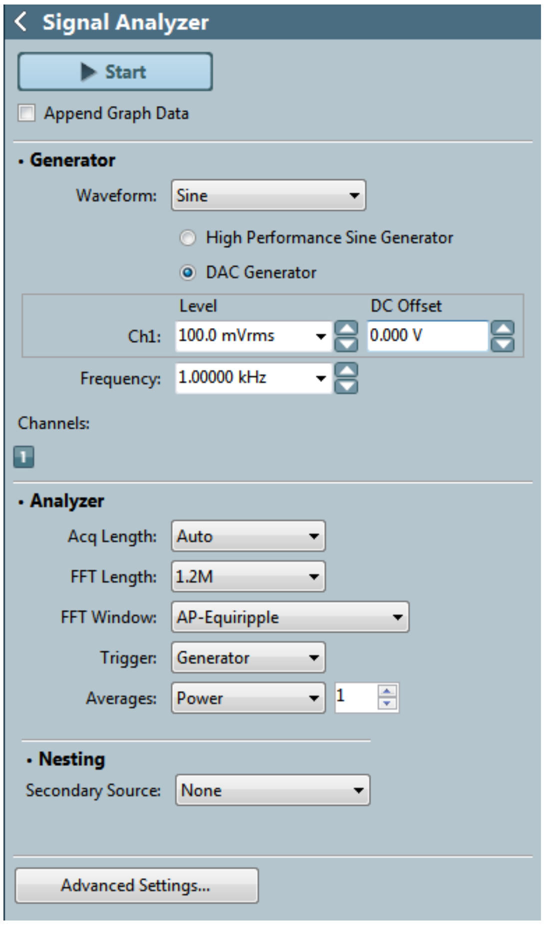
Units

Units available for Resistance (DCX) are

- ohm

73

Signal Analyzer
(Sequence Mode)



Introduction

For the APx515, this measurement requires a software option key. See page 166 for more information about software options.

The Signal Analyzer is a general-purpose diagnostic tool that provides a time-domain Scope result view, an FFT Spectrum result view and noise density views.

Signal Analyzer results available in APx500 are:

- Scope
- FFT Spectrum
- Amplitude Spectral Density
- Power Spectral Density

Operation

If you have not yet set up your test, first go to Signal Path Setup (Chapter 6). If you are configured for External Source, see the topic below.

Append Graph Data

Measured data are grouped in a **Data Set**, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 573.

Generator

The typical configuration for this measurement is the closed-loop configuration, using the APx generator as a stimulus. Alternatively, you can make open-loop measurements, using either the External Source or File Input configuration.

See Chapter 5 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 49 for a comparison of features.

Running the measurement

Click Start. The generator will output the test signal to the DUT on the selected generator channels. In a short period of time, the signal will be acquired and processed, and the two small graphs in the Selector will be populated with results.

If you would like to view the DUT output using no stimulus signal (for a view of DUT noise, for example), set the Level in Signal Generation to zero. Alternatively, you can use the Signal Analyzer in External Source configuration.

Analyzer

No local low pass, high pass or weighting filters are available for this measurement. However, low pass, high pass and weighting filter settings made in Signal Path Setup > Input/Output will affect this measurement. Read more about APx Filters in Chapter 91.

Acq Length

Set the acquisition length here, in seconds or in samples. Auto, the default, sets the acquisition length to the FFT length.

FFT Length

Set the FFT record length here. Options from 256 points to 1.2M points are available, depending upon the number of input channels in use; default is 16K points.

See “More About FFTs” on page 442.

FFT Window

FFT acquisitions must either be synchronous or have one of a number of amplitude windows applied to provide useful data for interpretation. Each window function brings advantages and disadvantages. The default selection, AP-Equiripple, is a proprietary Audio Precision FFT window that is an excellent choice for most FFT results.

The APx500 Signal Analyzer FFT window choices are:

- None
- Hann
- Blackman-Harris 3-term
- Blackman-Harris 4-term
- Flat Top
- AP-Equiripple

Detailed descriptions of the FFT window functions begin on page 443.

Trigger

- Free Run (the default)
  When Trigger is set to Free Run, the acquisition begins when the Start button is clicked (for a generator waveform, at the beginning of the waveform).
- +0 Crossing
  When Trigger is set to +0 Crossing, the acquisition begins at the waveform’s first positive-going zero crossing, for the channel selected.
- Generator
  Generator is only available if Signal Generation: Waveform is configured to play a file. When Trigger is set to Generator, the acquisition begins at the file’s first sample. A new acquisition is triggered each time the generator waveform loops back to its first sample.
- External
  External is only available for analyzers equipped with the Advanced Master Clock. The acquisition begins when a trigger pulse is received at the rear-panel Trigger In BNC connector. Set the Trigger Input in Signal Path Setup > External Trigger to the desired voltage threshold and pulse edge.

Averages

For non-periodic waveforms such as noise, averaging multiple acquisitions can provide a more useful view. By setting Averages to more than 1, you can specify how many acquisitions to average. Default is 1; maximum is 1000.

- Power
  The scope result does not use power averaging. Power averaging affects only the frequency domain results. See “More About Averaging” on page 444 for a description of power averaging.
- Synchronous
  When Trigger is set to Generator, Synchronous Averaging is an option. Synchronous averaging affects both the time domain and the frequency domain results. See “More About Averaging” on page 444 for a description of synchronous averaging.

Max Hold and Min Hold are not averaging functions, but use the multiple acquisitions available to the Averages feature to find maximum or minimum values for each FFT bin.

- Max Hold
  Max Hold displays the highest value for each bin found in all the acquisitions set in Averages. As the acquisitions progress, the value in each bin can

only go up. The scope result does not use **Max Hold**. This selection only affects the frequency domain results.

- **Min Hold**
  Min Hold displays the lowest value for each bin found in all the acquisitions set in **Averages**. As the acquisitions progress, the value in each bin can only go down. The scope result does not use **Min Hold**. This selection only affects the frequency domain results.

Signal

The Signal field is only available when Input Connector in Signal Path Setup is set to PDM.

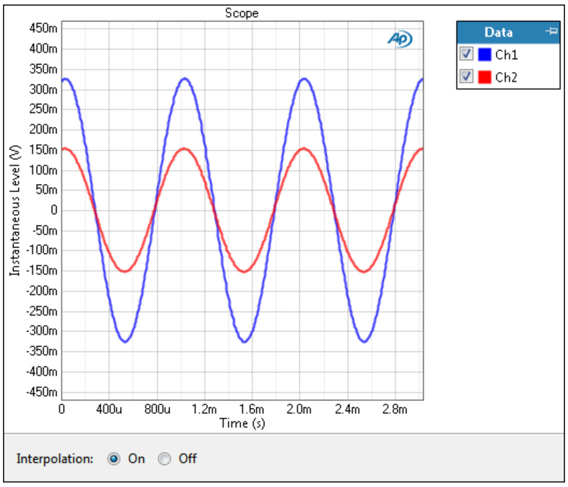
- **Decimated Audio**
  The decimated PDM audio (at the baseband sample rate and bandwidth) is the input signal.
- **PDM Bitstream**
  The PDM bitstream (at the PDM bit clock rate and bandwidth) is the input signal. This is useful to view the out-of-band noise and other components of the PDM bitstream.

Note: When the PDM input decimation is set to x1, (see "Settings for PDM Input" on page 138) the PDM signal is not decimated. In that case, either of the Signal selections in the Signal Analyzer will present the undecimated PDM bitstream for the analyzer acquisition.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or FFT windowing, click **Advanced Settings**. See Advanced Settings for the Signal Analyzer on page 441.

Signal Analyzer: Scope



The Signal Analyzer is a general purpose diagnostic tool that uses a high-resolution FFT technique to provide several different display results. Scope is a time domain result, displaying the signal amplitude versus time, analogous to an oscilloscope view.

Interpolation

The interpolation setting is only available when viewing Scope results.

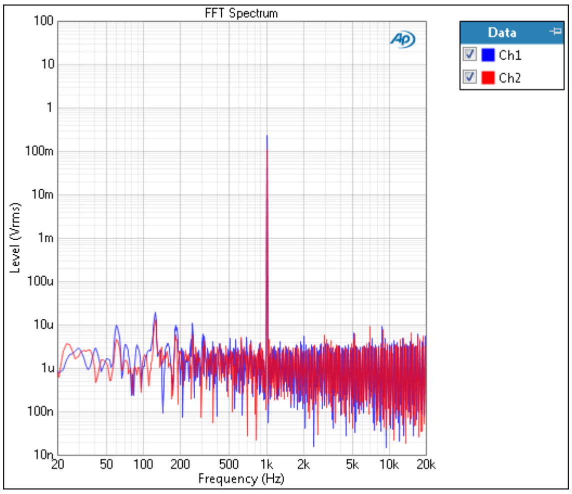
For time-domain displays, the APx500 graphs are normally plotted using sinc function interpolation. Interpolation adds points to the displayed trace that do not exist in the measured data, to make data trends more easily visualized. The default in APx is to have Interpolation **On**. However, digital domain signals are sometimes best understood when viewing actual samples, with interpolation switched **Off**. Also see "Interpolation and limit failure markers" on page 581.

Units

Units available for Signal Analyzer: Scope are

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• s	• V	• D • hex

Signal Analyzer: FFT Spectrum



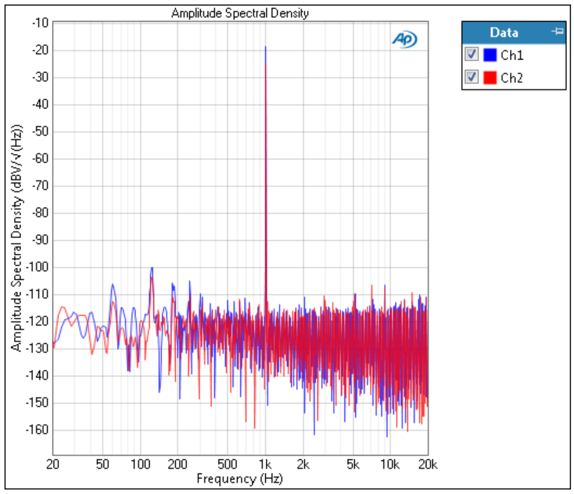
The Signal Analyzer is a general purpose diagnostic tool that uses a high-resolution FFT technique to provide several different display results. FFT Spectrum is a frequency domain result, displaying the signal amplitude versus frequency.

Units

Units available for Signal Analyzer: FFT Spectrum are

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz	• Vrms	• dBFS
• dHz	• dBV	• FS
• F/R	• dBu	• %FS
• %Hz	• dBrA	• dBrA
	• dBrB	• dBrB
	• dBSPL1	• dBSPL1
	• dBSPL2	• dBSPL2
	• dBm	
	• W (watts)	

Signal Analyzer: Amplitude Spectral Density



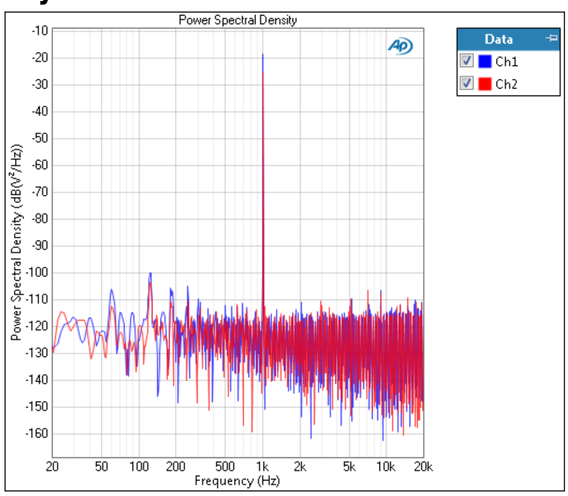
The Signal Analyzer is a general purpose diagnostic tool that uses a high-resolution FFT technique to provide several different display results. Amplitude Spectral Density is a frequency domain result, displaying the signal amplitude per root Hertz, versus frequency.

Units

Units available for Signal Analyzer: Amplitude Spectral Density are

X-axis	*Y-axis (analog)*	*Y-axis (digital)*
• Hz	• Vrms/√(Hz)	• FS/√(Hz)
• dHz	• dBV/√(Hz)	• dBFS/√(Hz)
• F/R	• dBu/√(Hz)	
• %Hz		

Signal Analyzer: Power Spectral Density



The Signal Analyzer is a general purpose diagnostic tool that uses a high-resolution FFT technique to provide several different display results. Power Spectral Density is a frequency domain result, displaying the signal amplitude squared per Hertz, versus frequency.

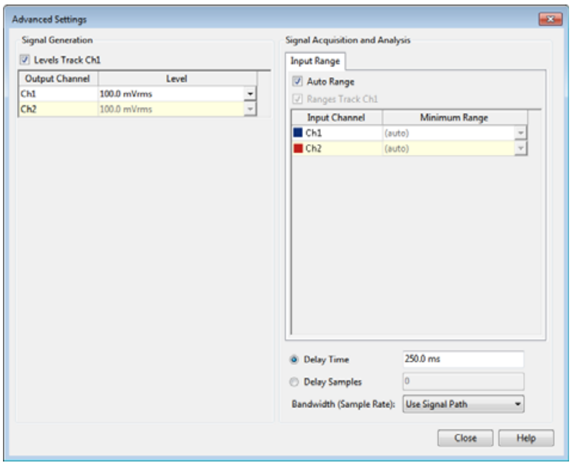
Units

Units available for Signal Analyzer: Power Spectral Density are

X-axis	***Y-axis (analog)***	***Y-axis (digital)***
• Hz	• dB(V^2/Hz)	• dB(FS^2/Hz)
• dHz	• V^2/Hz	• FS^2/Hz
• F/R		
• %Hz		

Advanced Settings for Signal Analyzer

The default settings here are appropriate for most measurements. You may want to make minor adjustments for special situations.



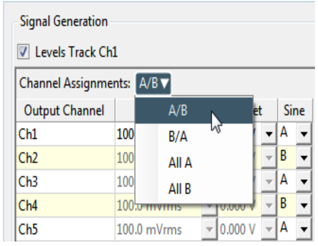
Signal Generation

If **Track first channel** is checked (the default), the generator output level values for channel 1 are copied to the other channels, and output levels for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels.

To set individual generator output channel levels, uncheck the “Track first channel level” checkbox and enter values in the output channel Level fields.

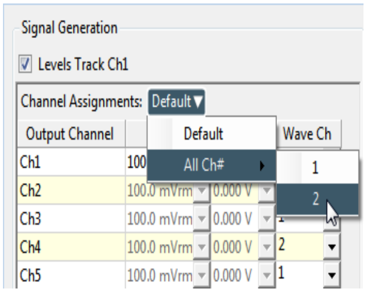
Set Channel Assignments for special waveforms

For **Split Sine** special waveform generation, you can set channel mapping. By default, channel A audio is mapped to odd numbered output channels, channel B audio to even numbered output channels. These assignments can be remapped by changing individual settings in the **Sine** column, or by selection one of several presets from the **Set Channel Assignments** menu.



Set Channel Assignments for generator waveforms

For stereo or multichannel generator waveform files, you can set channel mapping. By default, channel 1 audio is mapped to channel 1 output, channel 2 to channel 2, and so on. If the number of channels in the waveform file is less than the number of output channels, the waveform channels resume numbering at 1 and wrap to the next available output channel. These assignments can be remapped by changing individual settings in the **Wave Ch** column, or by selection one of several presets from the **Set Channel Assignments menu.**



To set individual generator output channel levels, **uncheck** the **Track first channel** checkbox and enter values in the output channel **Level** fields.

Signal Acquisition and Analysis

Input Range

Digital inputs do not require ranging, and Input Range settings are unavailable when inputs are set to a digital format.

The default operation of the Signal Analyzer measurement result views sets the channel 1 analyzer input minimum range to an initial value of 320.0 mVrms. The other channels follow the setting in Channel 1. The Advanced Settings dialog allows you to set independent minimum ranges for each analyzer input channel.

To set individual analyzer input channel minimum ranges, uncheck the **Track first channel** checkbox and enter values in the input channel Minimum Range fields.

If your Signal Analyzer measurement is a step in an automated sequence that you would like to run as fast as possible, you can optimize the speed of the ranging process by setting the range floor to the correct range for the measurement.

Delay Time/Delay Samples

In normal operation, both the generator and the analysis for the Signal Analyzer begin at the moment the Start button is clicked. The Delay Time/Delay Samples setting allows you to delay the analysis by a specific period of time the generator start.

This is sometimes necessary to avoid analysis of spurious signals created when using a coded generator file with a DUT that requires some time to match the coded format, or to allow the filter to settle when AC coupled.

Enter the delay in Seconds. Minimum time is 0.0 s; maximum is 60.0 s; the default it 250 ms.

If Signal Generation is using a Generator Waveform and Trigger is set to Generator, you can choose to enter the delay in Samples rather than seconds. Minimum sample count is 0; maximum is 12,960,000.

Bandwidth (Sample Rate)

This control is only available when the analyzer input configuration is set to Analog.

The Bandwidth (Sample Rate) control allows you to override the analog input bandwidth setting in Signal Path Setup. The input bandwidth set here operates only on the Signal Analyzer.

If your system analog bandwidth is high, you can reduce it here for faster measurements. If your system bandwidth is low, you can expand it here for more detailed Signal Analyzer results.

The analog input bandwidth (sample rate) settings, whether made here or in Signal Path Setup, result in channel-count constraints at high sample rates.

The bandwidth choices, followed by their corresponding sample rates, are:

- 2.75 kHz (6 kS/s)
- 3.5 kHz (8 kS/s)
- 5.5 kHz (12 kS/s)
- 7 kHz (16 kS/s)
- 11 kHz (24 kS/s)
- 20 kHz (44.1 kS/s)
- 22 kHz (48 kS/s)
- 40 kHz (88.2 kS/s)
- 45 kHz (96 kS/s)
- 80 kHz (176.4 kS/s)
- 90 kHz (192 kS/s)
- 250 kHz (624 kS/s) (requires the BW52 Option)
- 500 kHz (1.248 MS/s) (requires the BW52 Option)
- 1 MHz (2.496 MS/s) (requires the BW52 Option, set to 1 channel)
- Use Signal Path (the default)

Note that recording files at bandwidths above 90 kHz (sample rates above 192 kS/s) requires a very fast PC and hard disk, due to the high data rates. Slower systems can suffer recording interruptions and failure.

More about FFTs

The Fast Fourier Transform, or FFT, is a mathematical technique that deconstructs a complex waveform into its component sine waves. Presented with a segment of data which represents the amplitude vs. time relationships of a complex waveform, an FFT will derive two sets of results: the amplitude spectrum (amplitude vs. frequency) and the phase spectrum (phase vs. frequency) of the signal. This information is sufficient to describe the relationships of the sine waves which make up the waveform, and with further computation the FFT results can be interpreted for several different types of display.

Much of the signal analysis in APx500 is performed using FFT techniques. These results are typically displayed in the frequency domain, as level vs. frequency or phase vs. frequency spectrum graphs. Some results (such as Impulse Response and Scope) are displayed in the time domain, as level vs. time.

FFT length

The length (in samples) of the audio data being transformed is called the FFT length. Since this length consists of samples, its length in time varies with the sample rate. Many of the nominal FFT lengths shown in the APx menu are convenient approximations. The actual lengths are shown here:

Nominal (menu)	Actual Length in samples
256	256
512	512
1K	1,024
2K	2,048
4K	4,096
8K	8,192
12K	12,000
16K	16,384
24K	24,000
32K	32,768
48K	48,000

64K	65,536
96K	96,000
128K	131,072
192K	192,000
256K	262,144
300K	312,000
512K	524,288
600K	624,000
1M	1,048,576
1.2M	1,248,000

FFT bandwidth

The bandwidth of the FFT is from DC to 1/2 the sample rate.

FFT bins and bin width

The bandwidth of the FFT is divided into bins, the number of which is 1/2 the FFT length. The bin width (also called line spacing) defines the frequency resolution of the FFT. The FFT provides amplitude and phase values for each bin.

The bin width is stated in hertz. The bin width can be calculated by dividing the sample rate by the FFT length; or by dividing the bandwidth by the number of bins (which is equal to 1/2 the FFT length).

Examples

As an example, at a sample rate of 48 kHz and an FFT length of 16,384 (the 16K setting), the bandwidth would be DC to 24 kHz, the number of bins would be 8,192, and the bin width would be 24 kHz / 8,192 = 2.93 Hz.

With care, you can select sample rates and FFT lengths that will result in a specific bin width; for example, an FFT length of 65,536 samples (APx menu setting of 64K) and a sample rate of 65,536 Hz (we're assuming a digital input, with the signal at that sample rate) provides a bin width of exactly 1 Hz.

The FFT lengths of 312000, 624000 and 1248000 are included for use with the BW52, to provide integer bin widths at the sample rates used for BW52 bandwidths.

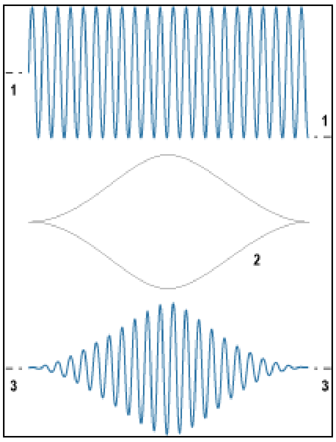
FFT windows

The Fourier transform has the effect of connecting the end of the FFT record to its beginning, forming an endless loop of data in the calculation.

In almost all cases, there is a discontinuity at the "splice," and just like a bad audio edit, the discontinuity is an impulse that spills energy across the spectrum. Unmodified, this energy swamps the useful data. In APx500 we use FFT windowing to eliminate the discontinuity.

An FFT window is an amplitude envelope that is applied to the FFT record before the transform. FFT windows change the data, and therefore affect the results. However, a number of windows have been developed that provide useful results while eliminating the looping discontinuity.

The example below shows the Hann window in the time domain, with the FFT record at the top, the window function in the center and the modified FFT record at the bottom. Other window functions have different shapes to optimize the resultant FFT for different purposes. What they all have in common is a tapering to zero at both ends, to eliminate the discontinuity.



The windows provided with APx500 are detailed here:

	Minimum sidelobe atten. (dB)	*Mainlobe width (bins)*	*Maximum level error (dB)*
Hann	----	4	–1.43
Blackman-Harris 3-term	58	6	–1.10
Blackman-Harris 4-term	92	8	–0.83
Flat Top	94	10	–0.02
AP-Equiripple	179	12	–0.50
Dolph-Chebyshev 150	147	12	–0.60
Dolph-Chebyshev 200	195	16	–0.46
Dolph-Chebyshev 250	240	20	–0.37

Hann

The Hann window is a raised cosine window that provides good selectivity near the top of the main lobe (about –6 dB at one bin away from center and about –30 dB at two bins away), with no side lobes. Its skirts more than 3 bins off center are not as steep as the Blackman-Harris window. The Hann window causes approximately –1.5 dB maximum amplitude error due to window attenuation, if the signal is at the extreme edge of the bin.

Blackman-Harris 3-term

The Blackman-Harris 3-term window is about –5 dB in the bins adjacent to the center, about –20 dB two bins off, and about –160 dB three bins off. Response fall-off is monotonic. Scallop loss is 1.1 dB maximum.

Blackman-Harris 4-term

The Blackman-Harris window is a 4-term minimum side lobe window, as used in AP2700 software. When compared to the Hann window, it is not quite as selective across the central several bins (about –3 dB in the adjacent bins and about –14 dB at two bins off), but has steeper skirts beyond that point. The Blackman-Harris window has side lobes below –92 dB (response fall-off is not monotonic). It has a reasonably flat top with a maximum amplitude error of about –0.8 dB if the signal is at the extreme edge of the bin.

Flat Top

The Flat-Top window is designed for the greatest amplitude measurement accuracy. It provides a maximum amplitude error due to window attenuation of less than –0.02 dB. However, its selectivity is poorer than the other windows. The Flat-Top window is the appropriate window for accurate amplitude measurements (such as when measuring individual harmonics) except when signals are so closely spaced that its selectivity becomes a problem.

AP-Equiripple

The Equiripple window, developed at Audio Precision, is an approximation to the Dolph-Chebyshev window that has the narrowest main lobe width for a given maximum side lobe depth. The main lobe is approximately 12 bins wide; that is, the first null is about six bins from the main lobe center. The first side lobe, which is also the highest, is –147 dB from the main lobe. The maximum amplitude error with a signal at the bin boundary is about 0.5 dB.

Dolph-Chebyshev

The Dolph-Chebyshev windows are true equiripple windows. For a given minimum sidelobe attenuation, they provide the minimum possible mainlobe width. The Dolph-Chebyshev 250 window is capable of revealing details in very low noise floors that other windows obscure.

More About Averaging

The Acoustic Response and Signal Analyzer measurements use averaging to reduce noise in measurement results. This technique averages the signal over a specified number of acquisitions by summing the signals from all acquisitions and dividing the result by the number of acquisitions.

Synchronous averaging

Synchronous averaging operates on a time domain acquisition. Synchronous averaging is useful to examine coherent time domain waveforms by reducing the average level (but not the variance) of noise and other non-coherent signals. Since the frequency domain results in a measurement are derived from the time domain acquisition, synchronous averaging affects spectrum results as well.

Power averaging

Power averaging (spectrum averaging) operates only on a frequency domain result. The power spectra for all averaged acquisitions are summed and then divided by the number of acquisitions. Power averaging helps reveal coherent frequency components by reducing the variance (but not the average level) of noise and other non-coherent signals.

Application in APx500

Synchronous averaging is available to all the results in the Signal Analyzer measurements, when using a Generator Waveform file as the stimulus signal, and triggering from the generator. The generator waveform is looped end-to-end for playback in the generator, and each repetition is synchronously acquired and added to the averaging process.

Synchronous averaging is also available to all the results in the Acoustic Response measurement.

Power averaging is available to the Signal Analyzer FFT Spectrum, Amplitude Spectral Density and Power Spectral Density results, and to the FFT Monitor.

Averaging acquisitions from 2 to 1000 are available. A selection of 1 disables averaging.

74

Signal to Noise Ratio
(Sequence Mode)

The Signal to Noise Ratio measurement provides a single-value result that shows the signal to noise ratio (SNR) of the output signal from each DUT channel. To use the SNR measurement to produce a dynamic range result, set the initial generator level to produce the maximum level in the DUT.

For the AES17 Dynamic Range measurement for digital converters, go to Chapter 41.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 6 for information about Signal Path Setup.

Running the measurement

To measure SNR, click **Start**. The generator is turned on briefly, then off. The results are compared and displayed as meter bars.

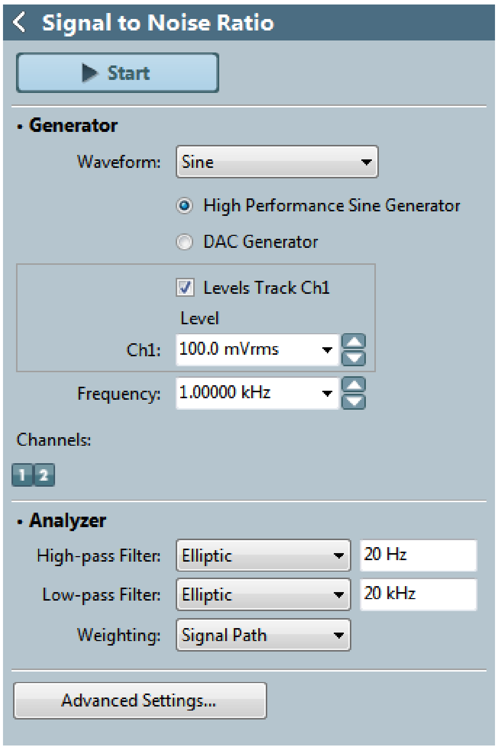
Generator

The typical configuration for this measurement is the closed-loop configuration, using the APx generator as a stimulus. Alternatively, you can make open-loop measurements, using either the External Source or File Input configuration.

See Chapter 5 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 49 for a comparison of features.



Level

For signal to noise ratio, you must choose a generator level that represents the maximum level for the device under test (DUT).

Analyzer

Filters

Local high-pass, low-pass and weighting filters are available for the "noise" (generator **OFF** or low level) portion of this measurement. See Measurement Filters on page 547 for detailed information about the filters available locally for some measurements.

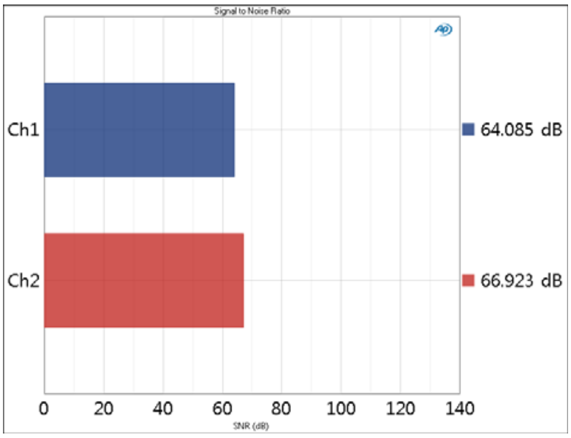
Low pass, high pass and weighting Signal Path Filters settings (see page 545) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass or high-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active. A local weighting filter setting does not override any Signal Path filter settings; the local weighting filter is applied in addition to any active Signal Path filters.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, click **Advanced Settings**. See Advanced Settings for single value measurement on page 317.

See Chapter 98 for more information about units of measurement.

Signal to Noise Ratio



The Signal to Noise Ratio measurement provides a single-value result showing the SNR for each DUT channel.

Units

Units available for SNR are

- x/y
- dB

More about SNR

Overview: SNR

Signal-to-noise ratio (SNR) was first used to evaluate the intelligibility of analog radio voice communications, and expressed the difference between the nominal signal level and the noise at the radio receiver. In excellent conditions, the voice might be 40 dB above the noise; in the worst conditions, the signal would be “lost in the noise” and unintelligible.

Strictly speaking, then, the “signal” level is arbitrary, although it is usually taken to be the “nominal” program level. SNR is actually two measurements: first, of the signal level; and second, of the noise level with the signal turned off. These two measurements are expressed as a ratio, almost always in decibels.

Like most noise measurements, the SNR results are typically bandpass limited using high and low pass filters, or a weighting filter. Filter use should be stated in the results.

The APx500 Signal-to-Noise Ratio measurement makes the two measurements and computes the ratio in one operation.

Overview: Dynamic Range

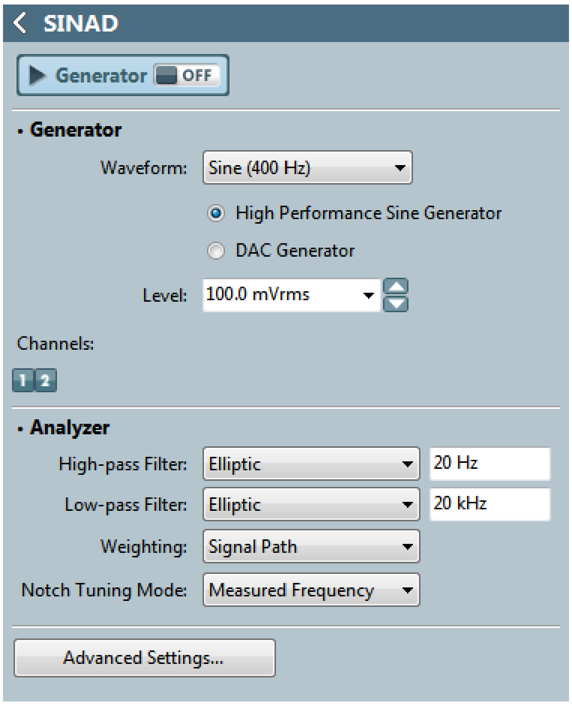
Dynamic range is an expression of the ratio of the largest signal a device can pass to the device's noise floor. “Largest signal” usually refers to a signal at a specified degree of distortion, often 1%. Signal-to-noise ratio and dynamic range are essentially the same measurement, except that the signal in SNR is arbitrary (and should be stated in the results), and the signal in dynamic range is at the maximum (details of which should also be stated in the results).

For the AES17 Dynamic Range measurement for digital converters, see Chapter 41.

75

SINAD
(Sequence Mode)

SINAD provides a single-value measurement of SINAD (Signal to Noise and Distortion) in the output signal from each DUT channel. ENOB is the mathematical equivalent of SINAD, and ENOB results are provided as well.



More Information

See More about SINAD and More about ENOB on page 448.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 6 for information about Signal Path Setup.

Running the measurement

To measure SINAD, click the Generator button to **On**. The generator will output the test signal to the DUT on the selected generator channels. Read the SINAD ratio for each channel from the meter bar display. Click the Generator button to **Off**.

Generator

The typical configuration for this measurement is the closed-loop configuration, using the APx generator as a stimulus. Alternatively, you can make open-loop measurements, using either the External Source or File Input configuration.

See Chapter 5 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

Selecting the generator waveform

By default, the SINAD measurement uses a sine wave at 400 Hz at the level set in the Signal Generation panel as the test signal. Alternatively, you can select a 1 kHz sine wave.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 49 for a comparison of features.

Analyzer

Filters

Local high-pass, low-pass and weighting filters are available for this measurement. See Measurement Fil-

ters on page 547 for detailed information about the filters available locally for some measurements.

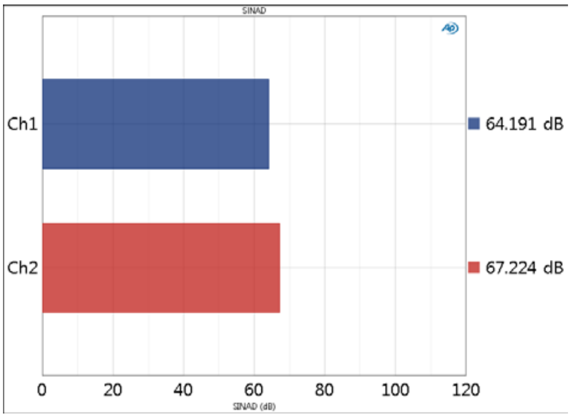
Low pass, high pass and weighting Signal Path Filters settings (see page 545) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass or high-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active. A local weighting filter setting does not override any Signal Path filter settings; the local weighting filter is applied in addition to any active Signal Path filters.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, click **Advanced Settings**. See Advanced Settings for single value measurement on page 317.

See Chapter 98 for more information about units of measurement.

SINAD



The SINAD measurement provides a single-value result showing the SINAD for each DUT channel.

Units

Units available for SINAD are

- x/y
- dB

More about SINAD

SINAD is an abbreviation for Signal-to-Noise and Distortion. It is a measure of audio quality used in low signal-to-noise ratio situations such as radio communications systems. SINAD is usually performed with a stimulus signal of 400 Hz or 1 kHz.

SINAD is defined as the ratio of the total signal (desired signal plus the sum of all distortion and noise components) to the sum of all distortion and noise components. SINAD can never be less than 1, and is always positive when expressed in dB. SINAD is the reciprocal of THD+N.

More about ENOB

ENOB (Effective Number Of Bits) is a measure of quality of a signal, expressed in terms that relate to digital acquisition processes. ENOB is the mathematical equivalent of SINAD (and, by extension, THD+N under similar conditions), by the conversion shown here:

$$\mathrm{ENOB} = \frac{\mathrm{SINAD} - 1.76}{6.02}$$

76

Stepped Frequency Sweep (Sequence Mode)

The **Stepped Frequency Sweep** measurement uses a sine wave stimulus signal that is moved across a range of frequencies in a series of steps or points. The DUT output is acquired by the analyzer and processed for display. X axis is the sweep frequency; Y axis is DUT output.

Stepped Frequency Sweep measurement results available in APx500 are:

- RMS Level (freq. response)
- THD Ratio
- Distortion Product Level
- Gain
- THD Level
- SINAD
- Relative Level
- THD+N Ratio
- Peak Level
- Deviation
- THD+N Level
- Average Jitter Level
- Phase
- Distortion Product Ratio

All of these results are available from a single measurement.

Operation

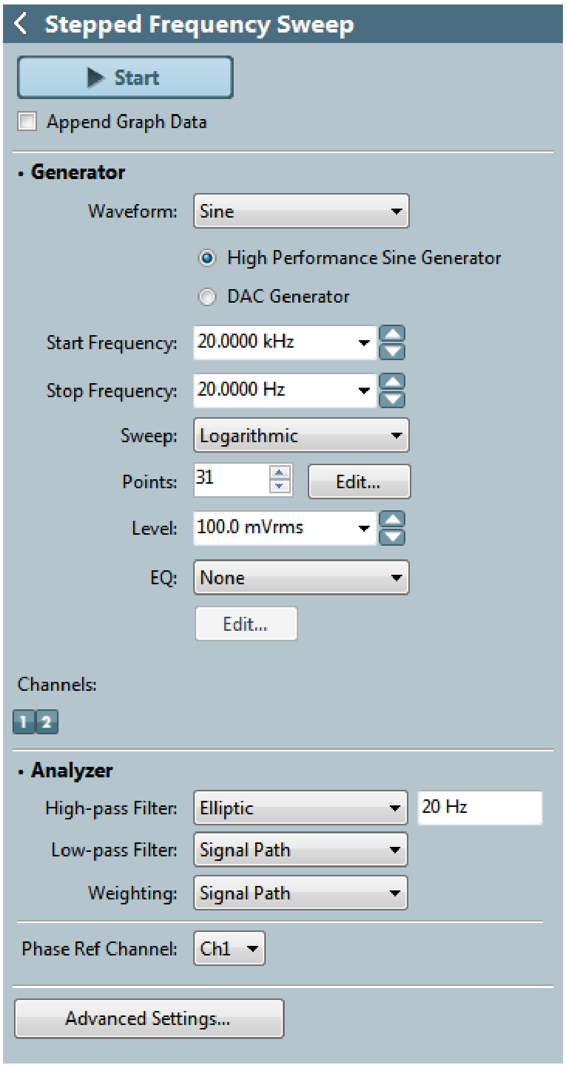
If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 6 for information about Signal Path Setup.

Running the Measurement

For a default sweep of 20 kHz to 20 Hz in 31 logarithmic steps, simply click **Start**. If you want to specify other parameters for your sweep, review **Generator** and **Analyzer** settings, discussed below. Click **Start** when you have approved the settings.

Append Graph Data

Measured data are grouped in a **Data Set**, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement,



and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 573.

Generator

The typical configuration for this measurement is the closed-loop configuration, using the APx generator as a stimulus. Alternatively, you can make open-loop measurements, using either the External Source or File Input configuration.

See Chapter 5 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 49 for a comparison of features.

Start Frequency

Set the sweep start frequency here.

Stop Frequency

Set the sweep stop frequency here.

Sweep

Choose

- **Logarithmic** point spacing
- **Linear** point spacing
- **Custom**
  Edit sweep spacing and number of **Points** to create a Custom sweep.

Points

Set the number of sweep points here.

Points / Edit

Open the **Sweep Points** dialog to edit import or export the generator **Sweep Points** table. See page 457.

Step Size

For linear sweeps, enter the **Step Size** here.

Level

Set the signal level here. This level may be changed if generator equalization is used, and can be adjusted on a per-channel basis in **Advanced Settings** (see page 458).

Offset

Set any DC offset to the generator signal here.

EQ

You can optionally modify the sweep with by applying a equalization curve to the generator. See "Generator Equalization" on page 170 for a general discussion of this feature. Choose

- **None**, which applies no equalization to the generator.
- **Relative**, which applies the gain or loss specified in the EQ table to the current generator level for each channel.
- **Absolute**, which sets the generator level for all channels to the levels specified in the EQ table. Note: when **Absolute** is selected, current channel level settings are lost.

Click the **Edit** button to create or change the EQ curve. See the **Edit EQ Table** dialog discussion on page 170.

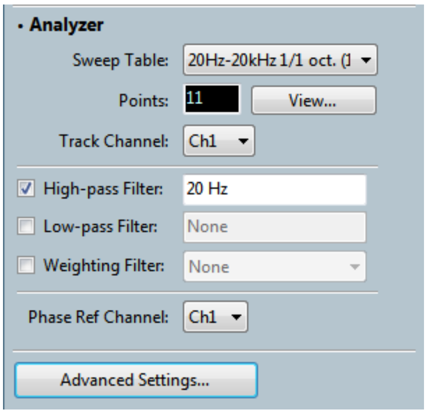
Ch

Toggle specific output channels on or off by clicking on the Channel number button.

Analyzer

This measurement can be configured to measure jitter in Signal Path Setup > Input > Measure. Read detailed information about jitter generation and measurement beginning on page 60.

Sweep Table



The analyzer Sweep Table menu, and the Points, View/Edit and Track Channel controls discussed below, are not shown for a closed loop sweep that uses the internal generator. In that case, the stimulus sweep definition is directly available to the analyzer and these controls are unnecessary.

When the measurement is configured so that the stimulus sweep definition is not available to the analyzer (in **External Source, Generator Waveform** or **File Input** configurations), you must select an analyzer

Sweep Table (see page 457) that matches the definition of the sweep you are using.

The analyzer **Sweep Table** menu lists the embedded sweep tables and any new tables that have been added to the project. Select a table from the analyzer **Sweep Table** menu, or choose **Create New** or **Browse** for a sweep table file.

Points

This field displays the number of points in the analyzer **Sweep Points** table.

View / Edit

For embedded analyzer sweep tables, click **View** (see page 457) to view or export the table, or to create a WAV file from the table. For other analyzer sweep tables, click **Edit** to edit, view or export the table, or to create a WAV file from the table (see page 458).

Filters

Local high-pass, low-pass and weighting filters are available for this measurement. The low-pass and high-pass filters, if selected, are applied to the entire measured signal and affect all the results. The weighting filter, if selected, affects only the distortion and distortion plus noise results. See Measurement Filters on page 547 for detailed information about the filters available locally for some measurements.

Low pass, high pass and weighting Signal Path Filters settings (see page 545) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass or high-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active. A local weighting filter setting does not override any Signal Path filter settings; the local weighting filter is applied in addition to any active Signal Path filters.

Phase Ref Channel

Interchannel Phase measurements must be referenced to one channel. Set the phase reference channel here.

Track Channel

When using a sweep table (in **External Source, Generator Waveform** or **File Input** configurations), the analyzer must determine the relation of the incoming signal to the sweep table. When **Auto** is selected, the analyzer continually polls all the channel inputs throughout the sweep, switching between channels as necessary to track the input with the highest signal level.

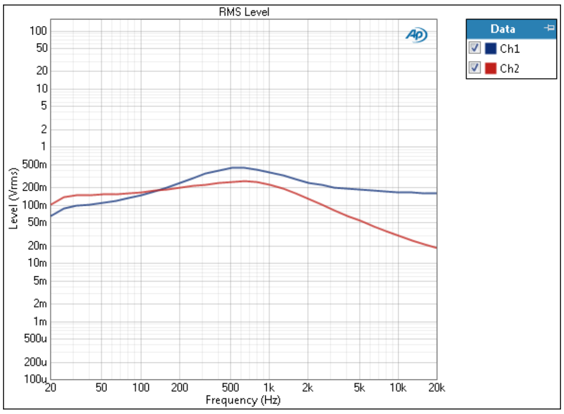
Alternatively, you can select a specific input channel for the analyzer to use for sweep tracking.

Advanced Settings

If your test required special adjustments or settings, click **Advanced Settings**. See page 458.

See Chapter 98 for more information about units of measurement.

RMS Level results



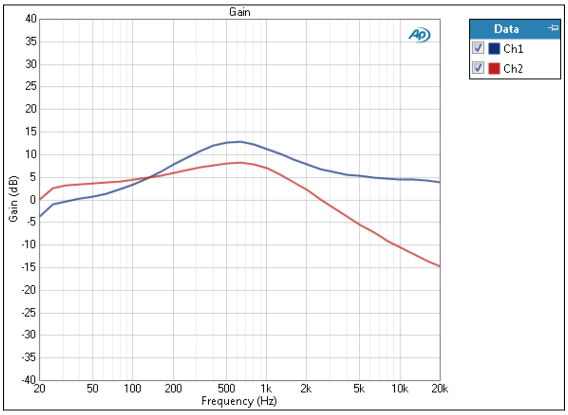
Stepped Frequency Sweep: RMS Level results uses a sine wave stimulus signal that is moved across a range of frequencies in a specified number of points. This is the conventional frequency response sweep.

Units

Units available for Stepped Frequency Sweep: RMS Level results are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*	*Jitter signals*
• Hz	• Vrms	• dBFS	• UI
• dHz	• dBV	• FS	• dBUI
• F/R	• dBu	• %FS	• s
• %Hz	• dBrA	• dBrA	
	• dBrB	• dBrB	
	• dBSPL1	• dBSPL1	
	• dBSPL2	• dBSPL2	
	• dBm		
	• W (watts)		

Gain



Gain is a stepped frequency sweep result that provides a graphical display of the voltage gain in each channel. In this result the DUT gain is plotted against frequency. The stepped frequency sweep Gain measurement is not available in External Source configuration.

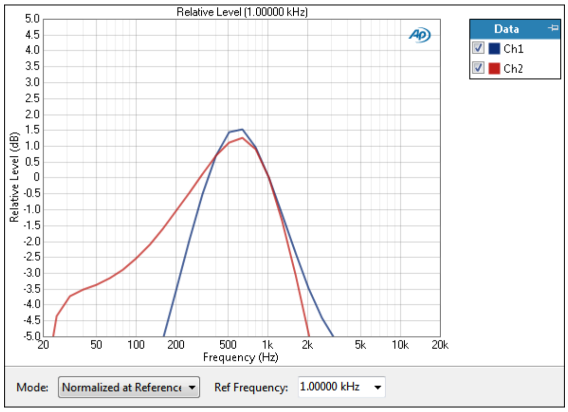
Gain results are not available when External Source is selected as the output configuration, when File is selected as the input configuration or when a generator waveform file is selected as the test signal.

Units

Units available for Stepped Frequency Sweep: Gain are:

X-axis	*Y-axis same-domain*	*Y-axis cross-domain*
• Hz	• x/y	• FS/Vrms
• dHz	• %	• dB(FS/Vrms)
• F/R	• ppm	—or—
• %Hz	• dB	• Vrms/FS
		• dB(Vrms/FS)

Stepped Frequency Sweep: Relative Level



The Stepped Frequency Sweep: Relative Level result uses a sine wave stimulus signal that is moved across a range of frequencies in a specified number of points, providing a conventional frequency response sweep. Additionally, in this result the response curve is plotted in relation to the measured level at a selected frequency. This enables you to specify the frequency where the relative response will be unity and to view the response in relation to that frequency. A midband frequency such as 1 kHz is often set as the reference frequency, and is the default here.

Mode

Select one of two result display modes.

Normalized at Reference

Enter a reference frequency for normalization in the Ref Frequency field. The measured level at the frequency you choose is set as 0 dB on the graph, and the DUT response is plotted in relation to this reference. You can change the Ref Frequency at any time (except after appending) and the graph will immediately be redrawn to reflect the new setting.

Auto-Centered in Limits

This control is only usable if upper and lower limits are set on this graph. In some applications, the general conformance of a trace to the shape of a set of limits is of importance, with absolute level less important or irrelevant. Auto-Centered in Limits centers the data trace relative to the upper and lower limits. To exclude possible exceptional data points at frequency extremes, the frequency range that is considered for the centering operation is constrained by the values in the Min and Max fields.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Ref Frequency

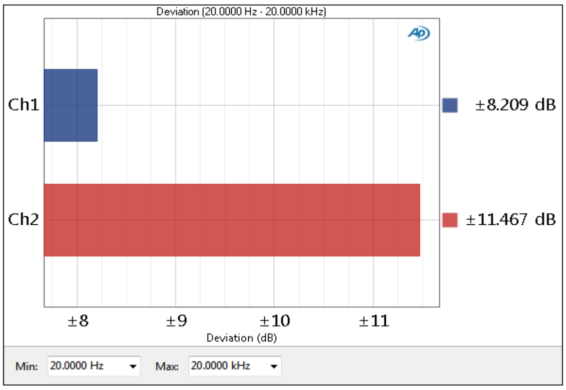
Choose a Reference Frequency here.

Units

Units available for Stepped Frequency Sweep: Relative Level are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Deviation



The Stepped Frequency Sweep: Deviation result uses a sine wave stimulus signal that is moved across a range of frequencies in a specified number of points. This result is a single value measurement computed from the stepped sweep that shows the frequency deviation (the total range of frequency variation) of each channel as a meter bar. You can specify a minimum and maximum frequency to define the range of the deviation measurement.

Result Settings

These settings are made in the Result Settings bar, beneath the graph display.

Min

Set the minimum frequency of the range of interest here.

Max

Set the maximum frequency of the range of interest here.

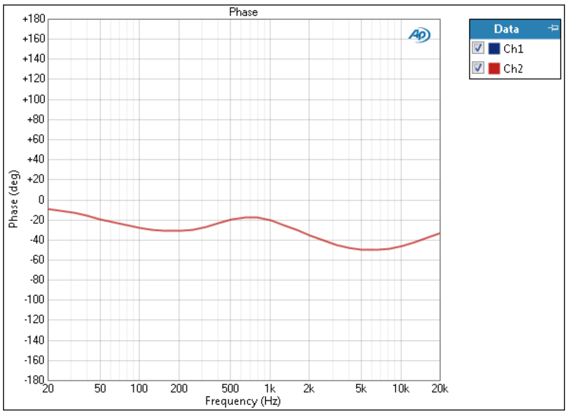
Units

Units available for Stepped Frequency Sweep: Deviation are

- dB

See Chapter 98 for more information about units of measurement.

Phase



The Phase result is a stepped frequency sweep measurement that provides a graphical display of the Phase response of each channel. One channel is chosen as the phase reference channel, and the remaining channels are measured against it.

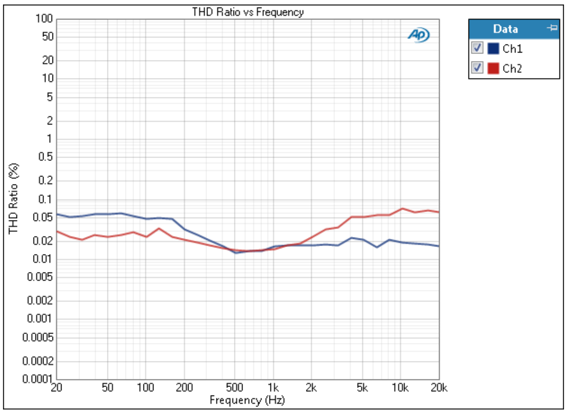
Select a phase reference channel in **Phase: Ref Channel.** The remaining channels are plotted against the selected channel.

Units

Units available for Stepped Frequency Sweep: Phase are:

X-axis	*Y-axis*
• Hz	• deg
• dHz	• rad
• F/R	
• %Hz	

THD Ratio



The THD Ratio result is a stepped frequency sweep measurement that provides a graphical display of the THD response of each channel. In this result the ratio of the level of the THD (total harmonic distortion) to the total signal in the DUT output is plotted against frequency.

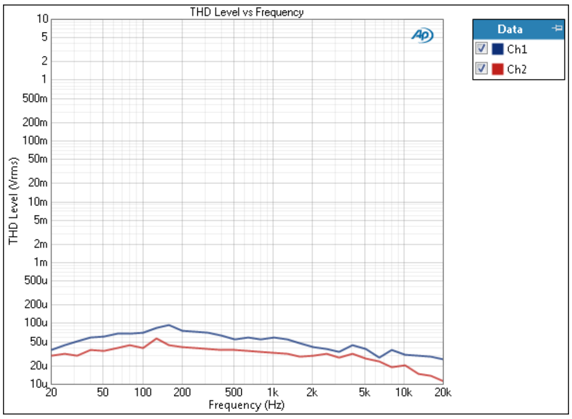
See page 475 for more information about THD.

Units

Units available for Stepped Frequency Sweep: THD Ratio are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

THD Level



The THD Level result is a stepped frequency sweep measurement that provides a graphical display of the THD response of each channel. In this result the level of the THD (total harmonic distortion) in the DUT output is plotted against frequency.

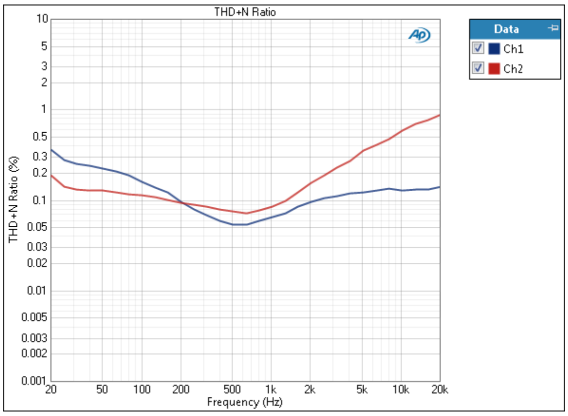
See page 475 for more information about THD.

Units

Units available for Stepped Frequency Sweep: THD Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*	*Jitter signals*
• Hz	• Vrms	• dBFS	• UI
• dHz	• dBV	• FS	• dBUI
• F/R	• dBu	• %FS	• s
• %Hz	• dBrA	• dBrA	
	• dBrB	• dBrB	
	• dBSPL1	• dBSPL1	
	• dBSPL2	• dBSPL2	
	• dBm		
	• W (watts)		

THD+N Ratio



The THD+N Ratio result is a stepped frequency sweep measurement that provides a graphical display of the THD+N response of each channel. In this result the ratio of the level of the THD+N (total harmonic distortion plus noise) to the total signal in the DUT output is plotted against frequency.

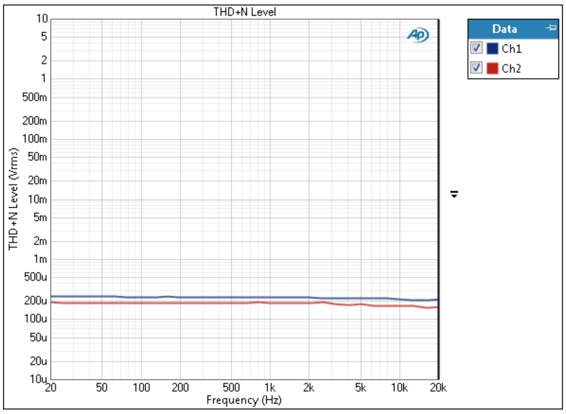
See page 475 for more information about THD+N.

Units

Units available for Stepped Frequency Sweep: THD+N Ratio are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

THD+N Level



The THD+N Level result is a stepped frequency sweep measurement that provides a graphical display of the THD+N response of each channel. In this result the level of the THD+N (total harmonic distortion plus noise) in the DUT output is plotted against frequency.

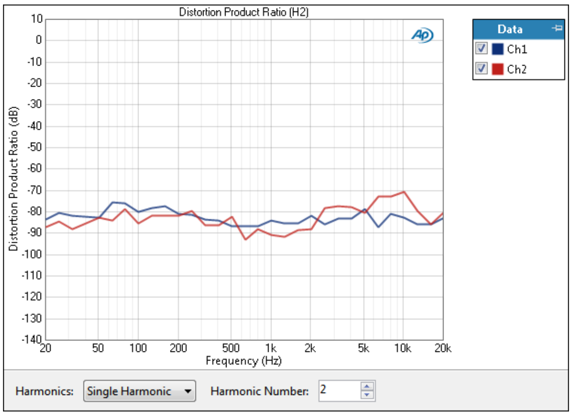
See page 475 for more information about THD+N.

Units

Units available for Stepped Frequency Sweep: THD+N Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*	*Jitter signals*
• Hz	• Vrms	• dBFS	• UI
• dHz	• dBV	• FS	• dBUI
• F/R	• dBu	• %FS	• s
• %Hz	• dBrA	• dBrA	
	• dBrB	• dBrB	
	• dBSPL1	• dBSPL1	
	• dBSPL2	• dBSPL2	
	• dBm		
	• W (watts)		

Distortion Product Ratio



The Stepped Frequency Sweep Distortion Product Ratio result provides a graphical display of the selected harmonic distortion products present in each channel. In this result the ratio of the level of the selected harmonic distortion product to the fundamental in the DUT output is plotted against frequency.

Select the harmonic distortion product(s) you wish to view using the **Harmonics** controls, described below.

Additional Controls for Distortion Product Ratio

Harmonics

For a graph of the ratio of one specific harmonic product to the signal level, select **Single Harmonic** and set the **Harmonic Number** to your choice, from 2nd harmonic to 20th harmonic.

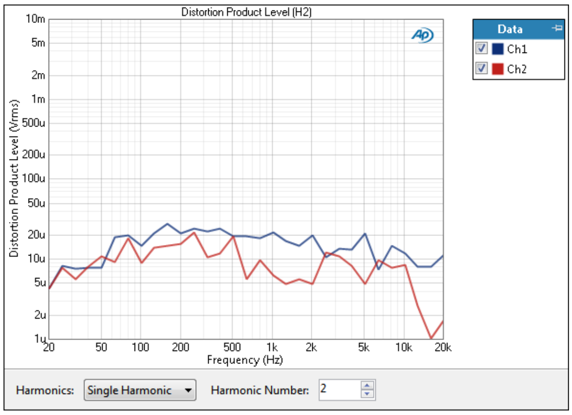
For a graph of the sum of the levels of any combination of harmonic products (from **H2** through **H20**), divided by the signal level, select **Sum of Harmonics**. Then click **Select Harmonics...** to open a result definition dialog.

Units

Units available for Stepped Frequency Sweep Distortion Product Ratio are:

X-axis	*Y-axis*
• Hz	• x/y
• dHz	• %
• F/R	• ppm
• %Hz	• dB

Distortion Product Level



The Stepped Frequency Sweep Distortion Product Level result provides a graphical display of the selected harmonic distortion products present in each channel. In this result the level of the selected harmonic distortion product in the DUT output is plotted against frequency.

Select the harmonic distortion product(s) you wish to view using the **Harmonics** controls, described below.

Additional Controls for Distortion Product Level

Harmonics

For a graph of the level of one specific harmonic product, select **Single Harmonic** and set the **Harmonic Number** to your choice, from 2nd harmonic to 20th harmonic.

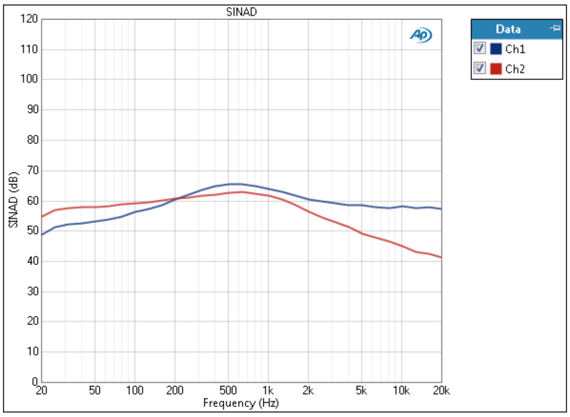
For a graph of the level of the sum of several or all harmonic products (from **H2** through **H20**), select **Sum of Harmonics**, then click **Select Harmonics...** to open a result definition dialog.

Units

Units available for Stepped Frequency Sweep Distortion Product Level are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*	*Jitter signals*
• Hz	• Vrms	• dBFS	• UI
• dHz	• dBV	• FS	• dBUI
• F/R	• dBu	• %FS	• s
• %Hz	• dBrA	• dBrA	
	• dBrB	• dBrB	
	• dBSPL1	• dBSPL1	
	• dBSPL2	• dBSPL2	
	• dBm		
	• W (watts)		

SINAD



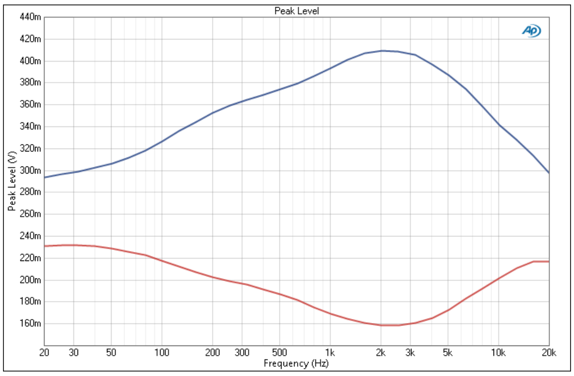
The Stepped Level Sweep: SINAD result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is generator level; Y axis is DUT output SINAD ratio.

Units

Units available for SINAD are:

X-axis (analog)	*X-axis (digital)*	*Y-axis*
• Vrms	• dBFS	• x/y
• Vp	• FS	• dB
• Vpp	• %FS	
• dBV	• dBrG	
• dBu		
• dBrG		
• dBm		
• W (watts)		

Peak Level



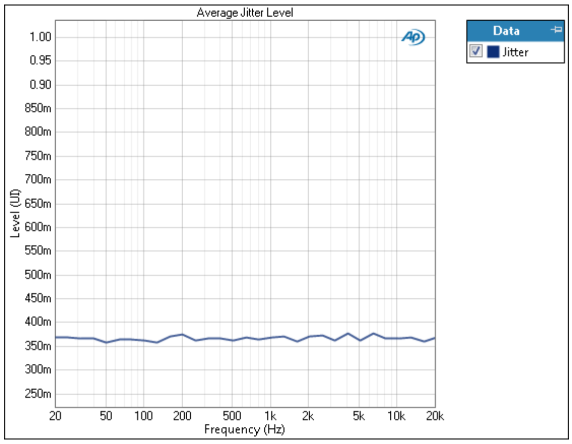
Stepped Frequency Sweep: Peak Level results uses a sine wave stimulus signal that is moved across a range of frequencies in a specified number of points. This is a frequency response sweep, with peak-scaled results.

Units

Units available for Stepped Frequency Sweep Peak Level results are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*	*Jitter*
• Hz	• V	• D	• UI
• dHz		• hex	• dBUI
• F/R			• s
• %Hz			

Average Jitter Level



Stepped Frequency Sweep: Average Jitter Level results uses a sine wave stimulus signal that is moved across a range of frequencies in a specified number of points.

Units

Units available for Stepped Frequency Sweep Level results are:

X-axis	*Y-axis (analog)*	*Y-axis (digital)*	*Jitter*
• Hz	• Vrms	• dBFS	• UI
• dHz	• dBV	• FS	• dBUI
• F/R	• dBu	• %FS	• s
• %Hz	• dBrA	• dBrA	
	• dBrB	• dBrB	
	• dBSPL1	• dBSPL1	
	• dBSPL2	• dBSPL2	
	• dBm		
	• W (watts)		

Using Sweep Tables with Stepped Frequency Sweep Measurements

A stepped frequency sweep depends upon the analyzer correctly tracking the stimulus sweep as it moves across its range.

When the source of the signal is the APx generator in an internal (closed loop) sweep, the tracking information is shared from generator to analyzer in real time. Additionally, the analyzer informs the generator when a tone has been successfully acquired, so that the generator can move to the next tone.

When a generator waveform or external source is used as the stimulus, or when a file is used as the analyzer input, a table that defines the sweep points must be provided to the analyzer. The analyzer then tracks the sweep (as defined in the **Sweep Table**) by monitoring one or more input channels. This method works well, unless the input signal is severely degraded.

When measuring from an external source, instead of using the generator, you will play an audio signal from the DUT.

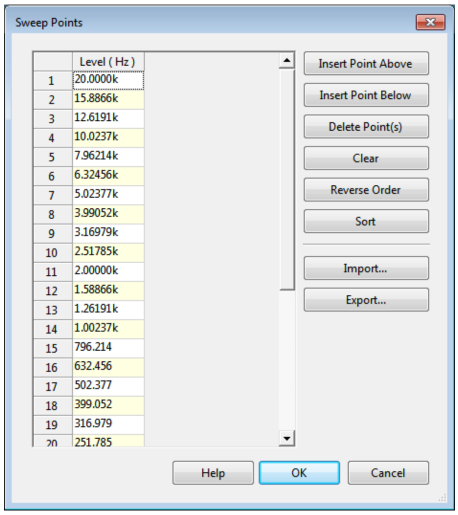
For a stepped frequency sweep, you must select one of the AP stepped frequency sweep test signals listed on the External Source panel, which are provided on-disc in several different formats. See “AP Test Signals on Disc” on page 150 and “Stepped Frequency Sweep Signals” on page 459.

When you have the test signal ready on the DUT and have selected it in APx500, click **Start** and play the test signal from the DUT. The analyzer will wait for the pilot tone, and then wait for each frequency point in the sweep. The signal is acquired and analyzed at each point, and the results are graphed.

If the wait for the pilot tone or any sweep point exceeds 60 seconds, the sweep will time out and end.

Viewing or Editing the Sweep Points tables

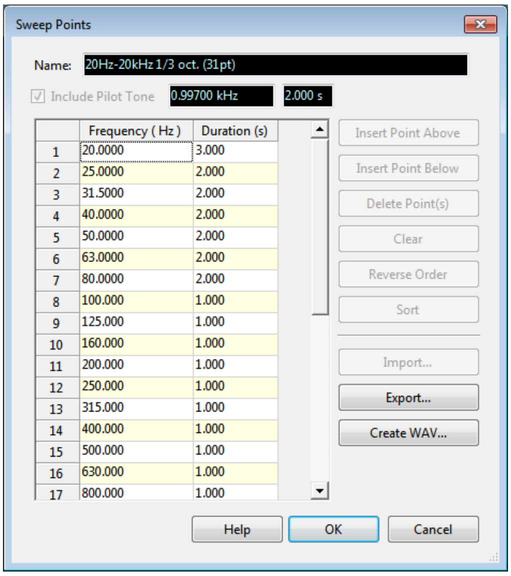
Stepped sweeps are defined by the points in a sweep table.



Generator Sweep Points Table (internal sweep).

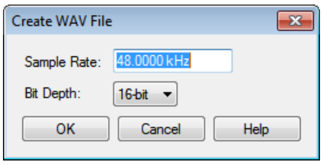
For internal sweeps (generated in real time), the dialog above allows you to specify generator frequencies and to add, delete and order the sweep points. The generator **Sweep Points** table can be exported to an XLS or CSV file; compatible files can be imported into the current sweep.

For external sweeps (from generator waveforms, file input or external source), an analyzer **Sweep Table** is necessary for the analyzer to track the sweep. The dialog below has the features of the internal generator **Sweep Points** dialog, with the addition of pilot tone settings and tone duration settings. You also have the option of creating a WAV file from the **Sweep Table**, as discussed next.



Analyzer Sweep Points Table (external sweep)

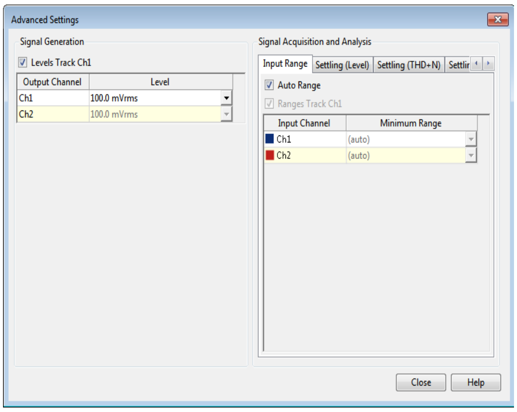
Create a WAV file from a Sweep Table



When you click **Create WAV File...** in the Sweep Points dialog, this dialog opens. Enter the **Sample Rate** (any sample rate from 22 kS/s to 192 kS/s) and **Bit Depth** (either 16-bit or a 24-bit) you require for your sweep WAV file. Click **OK** to open a file browser to save the file.

Advanced Settings for Stepped Sweep Measurements

The default settings here are appropriate for most measurements. You may want to make minor adjustments for special situations.



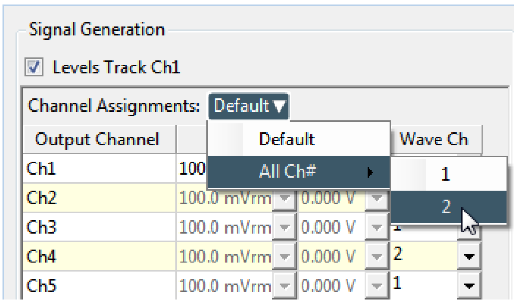
Signal Generation

If **Track First Channel** is checked (the default), the generator output level values for channel 1 are copied to the other channels, and output levels for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels.

To set individual generator output channel levels, **uncheck** the **Track First Channel** checkbox and enter values in the output channel Level fields.

Set Channel Assignments for generator waveforms

For stereo or multichannel generator waveform files, you can set channel mapping. By default, channel 1 audio is mapped to channel 1 output, channel 2 to channel 2, and so on. If the number of channels in the waveform file is less than the number of output channels, the waveform channels resume numbering at 1 and wrap to the next available output channel. These assignments can be remapped by changing individual settings in the **Wave Ch** column, or by selection one of several presets from the **Set Channel Assignments menu.**



File Playback

For generator waveform file playback, you can view the length of the file, and you can adjust the playback start position. Select Seconds or Samples for the Length and Start Position units. Enter a new Start Position if desired.

Signal Acquisition and Analysis

Input Range tab

Digital inputs do not require ranging, and the Input Range tab is unavailable when inputs are set to a digital format.

If Auto Range is checked (the default), each analog channel determines its input range automatically, based on the level of the input signal. If the input signal level changes beyond a ranging threshold, Auto Range will cause the input ranging circuits to move up or down for proper ranging.

See page 551 for more information about ranging and autoranging.

If Auto Range is unchecked, you can set a fixed maximum range for each analog input channel. If Track First Channel is checked (the default), the fixed input range setting for channel 1 is copied to the other channels, and range settings for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels.

To set individual analog input channel ranges, uncheck the Track First Channel checkbox and enter values in the input channel Maximum Range fields.

Settling tabs

Two or three Settling tabs are available for Single Value and Stepped Sweep measurement results. These tabs may have different parameters and default values depending upon the See page 552 for more information about settling.

Stepped Frequency Sweep Signals

Most of these sweeps are provided on playable optical disc media available from Audio Precision. These sweeps are available at several sample rates, several signal levels, several channels counts, and as linear and a choice of coded formats.

- APx-DVD1 has a range of sample rates and formats.
- APx-CD1 has only 16-bit linear files at 44.1 kHz sample rate.

A subset of somewhat shorter versions of the linear files are also available for generator waveform use. These are available on the AP Web site at ap.com.

Coded waveforms (such as Dolby or DTS) must be decoded in the DUT before analysis.

The six sweeps at the end of the list are Dolby or DTS proprietary stimulus signals, available on discs provided to licensees of Dolby or DTS technologies.

The Audio Precision sweeps begins with a short (2 second to 5 second) 997 Hz pilot tone at –20 dBFS. This sets the analyzer to its ready condition. The rest of the sweep is at 0 dBFS, –1 dBFS or –20 dBFS, as selected.

Sweep definitions for stimulus signals on Audio Precision media.

20 Hz–20 kHz 1 octave (11 point)

Media: APx-CD1, APx-DVD1

Description: Pilot tone, followed by 11 pt sweep (approximate 1-octave centers) 20 Hz–20 kHz @ 0 dBFS.

Sweep duration: 00:15.

20 Hz–20 kHz 1 octave (11 point) (slow)

Media: APx–CD1, APx-DVD1

Description: Pilot tone, followed by 11 pt sweep (approximate 1-octave centers) 20 Hz–20 kHz @ 0 dBFS. Low-frequency tones are long to allow stabilization of DUT and analyzer.

Sweep duration: 00:33.

20 Hz–20 kHz 1/3 octave (31 point)

Media: APx-CD1, APx-DVD1

Description: Pilot tone, followed by 31 pt sweep (approximate 1/3-octave centers) 20 Hz–20 kHz @ 0 dBFS.

Sweep duration: 00:39.

20 Hz–20 kHz 1/3 octave (31 point) (slow)

Media: APx-CD1, APx-DVD1

Description: Pilot tone, followed by 31 pt sweep (approximate 1/3-octave centers) 20 Hz–20 kHz @ 0 dBFS. Low-frequency tones are long to allow stabilization of DUT and analyzer.

Sweep duration: 01:33.

20 Hz–20 kHz 1/6 octave (61 point)

Media: APx-CD1, APx-DVD1

Description: Pilot tone, followed by 61 pt sweep (approximate 1/6-octave centers) 20 Hz–20 kHz @ 0 dBFS.

Sweep duration: 1:15.

20 Hz–20 kHz 1/6 octave (61 point) (slow)

Media: APx-CD1, APx-DVD1

Description: Pilot tone, followed by 61 pt sweep (approximate 1/6-octave centers) 20 Hz–20 kHz @ 0 dBFS. Low-frequency tones are long to allow stabilization of DUT and analyzer.

Sweep duration: 03:03.

20 Hz–22 kHz 1/6 octave (62 point)

Description: Pilot tone, followed by 62 pt sweep (approximate 1/6-octave centers) 20 Hz–22.4 kHz @ 0 dBFS.

Proprietary sweep definitions from Dolby.

You must be a licensee and obtain stimulus files or media from Dolby Labs.

Dolby 1/6 Oct. sweep (AutoTest 2.0) 20 Hz–20 kHz

Proprietary sweep available from Dolby for licensees of Dolby technology.

Dolby 1/6 Oct. sweep 20 kHz–20 Hz (60pt)

Proprietary sweep available from Dolby for licensees of Dolby technology.

Dolby TrueHD 20 Hz–24 kHz (60pt)

Proprietary sweep available from Dolby for licensees of Dolby technology.

Dolby TrueHD 24 kHz–20 Hz (60pt)

Proprietary sweep available from Dolby for licensees of Dolby technology.

Dolby TrueHD 20 Hz–48 kHz (60pt)

Proprietary sweep available from Dolby for licensees of Dolby technology.

Dolby TrueHD 48 kHz–20 Hz (60pt)

Proprietary sweep available from Dolby for licensees of Dolby technology.

Dolby TrueHD 20 Hz–96 kHz (60pt)

Proprietary sweep available from Dolby for licensees of Dolby technology.

Dolby TrueHD 96 kHz–20 Hz (60pt)

Proprietary sweep available from Dolby for licensees of Dolby technology.

Dolby 1/6 Oct. sweep (Test DVD 1.5)

Proprietary sweep available from Dolby for licensees of Dolby technology.

Dolby 1/6 Oct. LFE (20–120 Hz) sweep (Test DVD 1.5)

Proprietary sweep available from Dolby for licensees of Dolby technology.

Proprietary sweep definitions from DTS.

You must be a licensee and obtain stimulus files or media from DTS.

DTS 20 Hz–20 kHz @ –20 dBFS (30 point)

Proprietary sweep available from DTS for licensees of DTS technology.

DTS 20 Hz–300 Hz @ –20 dBFS (15 point)

Proprietary sweep available from DTS for licensees of DTS technology.

DTS 20 Hz–40 kHz (31 point)

Proprietary sweep available from DTS for licensees of DTS technology.

77

Stepped Level Sweep
(Sequence Mode)

The **Stepped Level Sweep** measurement provides a sine wave stimulus signal that is moved across a range of levels in a series of points. The DUT output is acquired by the analyzer and processed for display. **Stepped Level Sweep** measurements are not supported in the **External Source** configuration.

Stepped Level Sweep results available in APx500 are:

• Level	• THD Ratio	• THD Level vs. Measured Level
• Gain	• THD Level	• SINAD
• Linearity	• THD+N Ratio vs. Measured Level	• Peak Level
• THD+N Ratio	• THD+N Level vs. Measured Level	• Average Jitter Level
• THD+N Level	• THD Ratio vs. Measured Level	

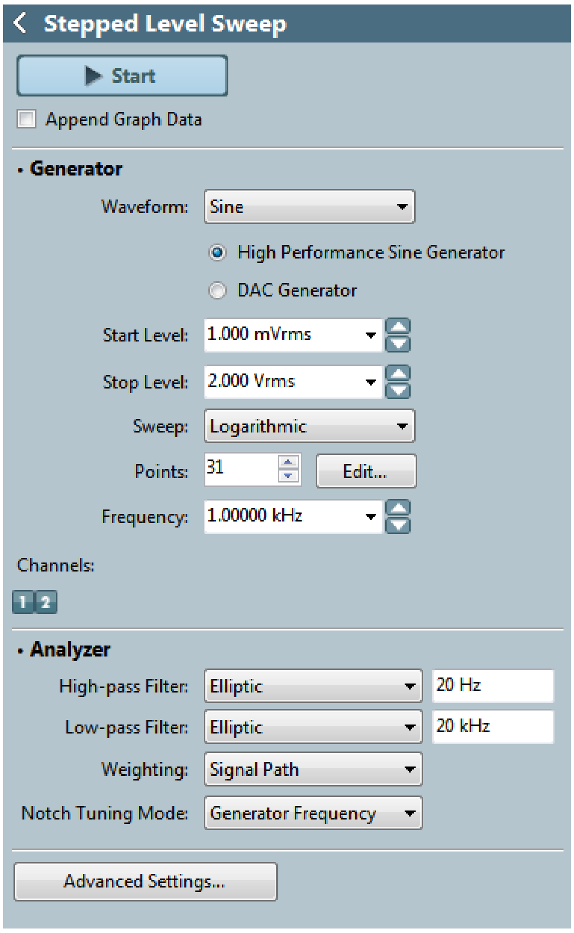
All of these results are available from a single measurement.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 6 for information about Signal Path Setup.

Running the Measurement

For a default sweep of 1 mVrms to 2 Vrms in 31 logarithmic steps, simply click **Start**. If you want to specify other parameters for your sweep, review **Generator** and **Analyzer** settings, discussed below. Click **Start** when you have approved the settings.



Append Graph Data

Measured data are grouped in a **Data Set**, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 573.

Generator

The configuration for this measurement must be the closed-loop configuration, using the APx generator as a stimulus.

See Chapter 5 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings.

High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 49 for a comparison of features.

Note: if Output Configuration is set to Bluetooth (A2DP Source profile), the Bluetooth hardware imposes a –54 dBFS cutoff. Signals below –54 dBFS are not transmitted. Please consider this when defining the range of levels for your Stepped Level sweep.

Start Level, Stop Level and Sweep Points

The sweep moves between two levels, set in the **Start Level** and **Stop Level** fields. The default levels are 1.000 mVrms and 2.000 Vrms.

The sweep moves in a specified number of step points, set in the **Points** field. The minimum is 2 points; maximum is 65,535. The default setting is 31.

The sweep point spacing is set by selecting one of the following choices in the Sweep field:

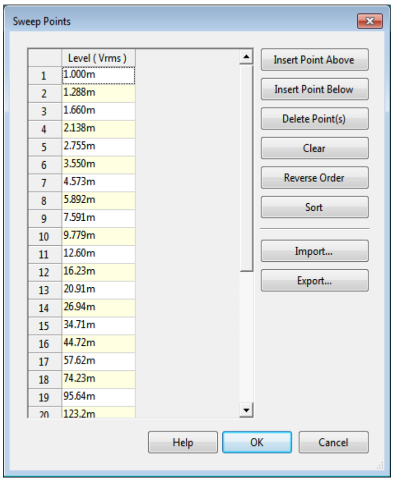
- **Logarithmic** (the default); use the **Points** field to set the number of logarithmically spaced points;
- **Linear**, which provides two methods of adjusting spacing: the **Points** field or the **Step Size** field; or
- **Custom**. Click **Edit** to open the **Sweep Points** dialog, where you can set points arbitrarily, or load or save sweep table files.

Viewing or Editing the Sweep Points table

You can view or edit the current generator sweep points at any time.

Click **Edit** to open the generator **Sweep Points** table. The table shows each sweep point and its corresponding level. You can edit this table to add or delete points, or to change the level of a point. Points can be sorted or reversed in order using the controls on the right.

A **Sweep Points** table can be saved as a *.csv file or as a Microsoft Excel *.xls file. A compatible *.csv or *.xls file can be opened and used as a **Sweep Points** table.



Analyzer

Filters

Local high-pass, low-pass and weighting filters are available for this measurement. The low-pass and high-pass filters, if selected, are applied to the entire measured signal and affect all the results. The weighting filter, if selected, affects only the distortion and distortion plus noise results. See Measurement Filters on page 547 for detailed information about the filters available locally for some measurements.

Low pass, high pass and weighting Signal Path Filters settings (see page 545) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass or high-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active. A local weighting filter setting does not override any Signal Path filter settings; the local weighting filter is applied in addition to any active Signal Path filters.

Notch Tuning Mode

For the harmonic distortion results, a tunable notch filter is used to remove the fundamental from the measurement. The filter can be tuned to:

- Generator Frequency
  The current APx audio generator frequency. When the generator channels are outputting different frequencies (Split Frequency generation), the notch filter center is set to Frequency A. This mode is not available when using a generator waveform file.
- Jitter Generator Frequency
  The current APx jitter generator frequency, when

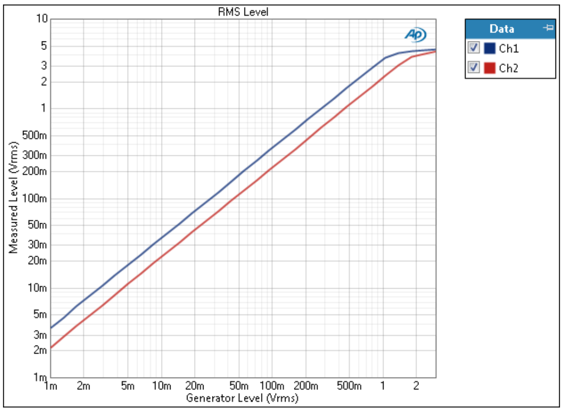
jitter generation is available and enabled. Read more about jitter beginning on page 60.

- Measured Frequency
  The current measured frequency. When the analyzer channels are receiving different frequencies, the notch filter for each channel is centered on the frequency in that channel.
- Fixed Frequency
  A fixed frequency selected by the user. When **Fixed Frequency** is selected, a **Filter Frequency** entry field becomes available beneath the **Notch Tuning Mode** control.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, click **Advanced Settings**. See Advanced Settings for Stepped Sweeps on page 458. See Chapter 98 for more information about units of measurement.

Stepped Level Sweep: Level



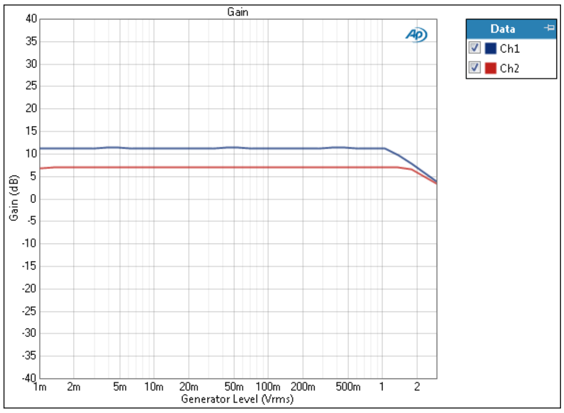
The Stepped Level Sweep: Level result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is generator level; Y axis is DUT output level.

Units

Units available for Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis (analog)*	*Y-axis (digital)*
• Vrms	• dBFS	• Vrms	• dBFS
• Vp	• FS	• dBV	• FS
• Vpp	• %FS	• dBu	• %FS
• dBV	• dBrG	• dBrA	• dBrA
• dBu		• dBrB	• dBrB
• dBrG		• dBSPL1	• dBSPL1
• dBm		• dBSPL2	• dBSPL2
• W (watts)		• dBm	
		• W (watts)	

Stepped Level Sweep: Gain



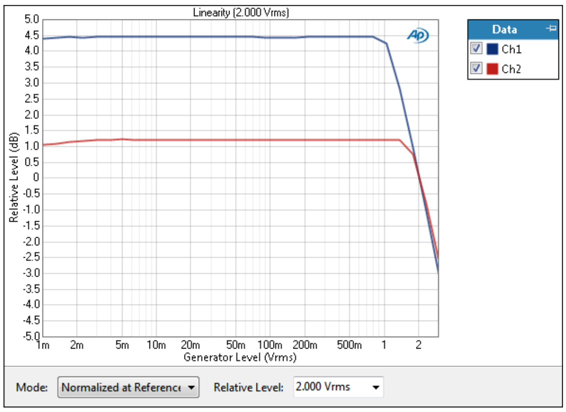
The Stepped Level Sweep: Gain result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is generator level; Y axis is gain in the DUT. Gain is a ratio, calculated by dividing the DUT output level by the generator level.

Units

Y-axis units available for Gain are:

X-axis (analog)	*X-axis (digital)*	*Y-axis same-domain*	*Y-axis cross-domain*
• Vrms	• dBFS	• x/y	• FS/Vrms
• Vp	• FS	• %	• dB(FS/Vrms)
• Vpp	• %FS	• ppm	—or—
• dBV	• dBrG	• dB	• Vrms/FS
• dBu			• dB(Vrms/FS)
• dBrG			
• dBm			
• W (watts)			

Stepped Level Sweep: Linearity



The Stepped Level Sweep: Linearity result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is generator level; Y axis is linearity in the DUT. Linearity is a ratio, calculated by dividing the DUT output level by the generator level, normalized for a chosen relative level.

Mode

Normalized at Reference

Enter a reference level for normalization in the **Relative Level** field. To calculate linearity, each channel is normalized so that a linearity of unity (0 dB) is at the same measured DUT output level. Set this reference level in the **Relative Level** field. Default **Relative Level** is 1 Vrms (analog), 1 FS (digital). You can change the **Relative Level** at any time (except after appending) and the graph will immediately be redrawn to reflect the new setting.

Auto-Centered in Limits

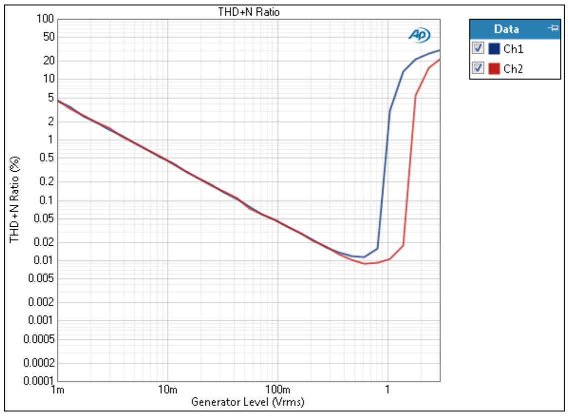
This control is only usable if upper and lower limits are set on this graph. In some applications, the general conformance of a trace to the shape of a set of limits is of importance, with absolute level less important or irrelevant. **Auto-Centered in Limits** centers the data trace relative to the upper and lower limits. To exclude possible exceptional data points at level extremes, the amplitude range that is considered for the centering operation is constrained by the values in the **Min** and **Max** fields.

Units

Units available for Linearity are:

X-axis (analog)	*X-axis (digital)*	*Y-axis*
• Vrms	• dBFS	• x/y
• Vp	• FS	• %
• Vpp	• %FS	• ppm
• dBV	• dBrG	• dB
• dBu		
• dBrG		
• dBm		
• W (watts)		

Stepped Level Sweep: THD+N Ratio



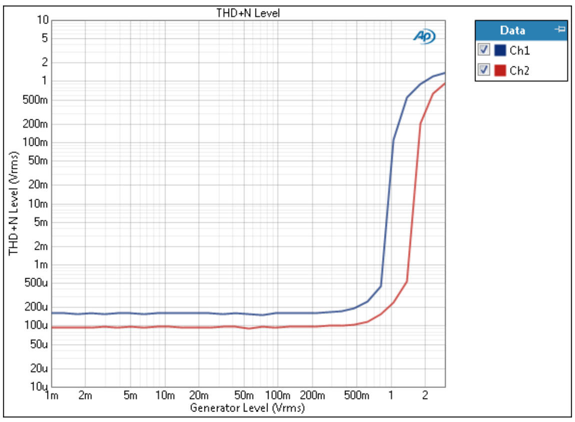
The Stepped Level Sweep: THD+N Ratio result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is generator level; Y axis is DUT output THD+N ratio.

Units

Units available for THD+N Ratio are:

X-axis (analog)	*X-axis (digital)*	*Y-axis*
• Vrms	• dBFS	• x/y
• Vp	• FS	• %
• Vpp	• %FS	• ppm
• dBV	• dBrG	• dB
• dBu		
• dBrG		
• dBm		
• W (watts)		

Stepped Level Sweep: THD+N Level



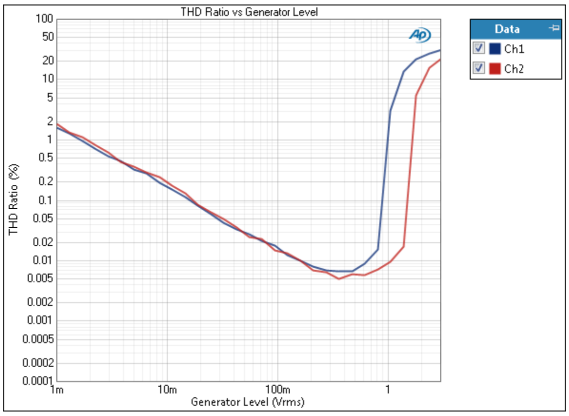
The Stepped Level Sweep: THD+N Level result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is generator level; Y axis is DUT output THD+N level.

Units

Y-axis units available for THD+N Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis (analog)*	*Y-axis (digital)*
• Vrms	• dBFS	• Vrms	• dBFS
• Vp	• FS	• dBV	• FS
• Vpp	• %FS	• dBu	• %FS
• dBV	• dBrG	• dBrA	• dBrA
• dBu		• dBrB	• dBrB
• dBrG		• dBSPL1	• dBSPL1
• dBm		• dBSPL2	• dBSPL2
• W (watts)		• dBm	
		• W (watts)	

Stepped Level Sweep: THD Ratio



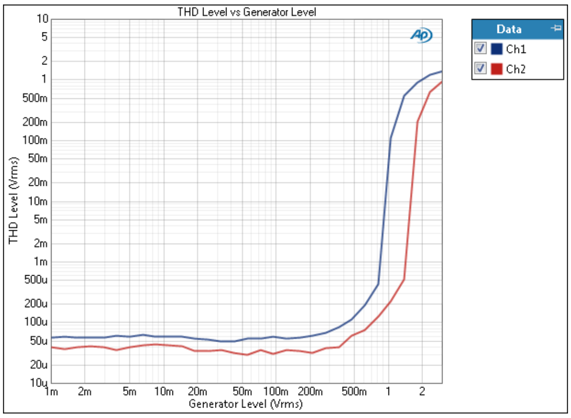
The Stepped Level Sweep: THD Ratio result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is generator level; Y axis is DUT output THD ratio.

Units

Units available for THD Ratio are:

X-axis (analog)	*X-axis (digital)*	*Y-axis*
• Vrms	• dBFS	• x/y
• Vp	• FS	• %
• Vpp	• %FS	• ppm
• dBV	• dBrG	• dB
• dBu		
• dBrG		
• dBm		
• W (watts)		

Stepped Level Sweep: THD Level



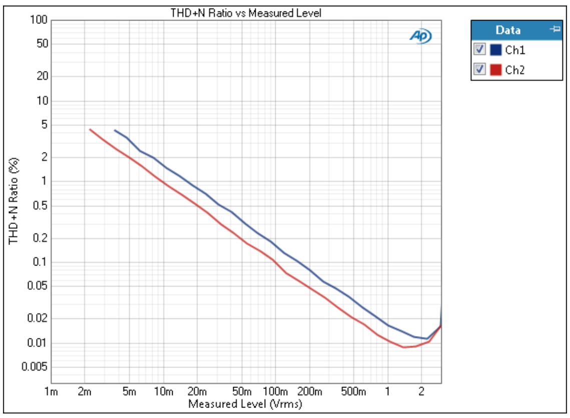
The Stepped Level Sweep: THD Level result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is generator level; Y axis is DUT output THD level.

Units

Y-axis units available for THD Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis (analog)*	*Y-axis (digital)*
• Vrms	• dBFS	• Vrms	• dBFS
• Vp	• FS	• dBV	• FS
• Vpp	• %FS	• dBu	• %FS
• dBV	• dBrG	• dBrA	• dBrA
• dBu		• dBrB	• dBrB
• dBrG		• dBSPL1	• dBSPL1
• dBm		• dBSPL2	• dBSPL2
• W (watts)		• dBm	
		• W (watts)	

Stepped Level Sweep: THD+N Ratio vs. Measured Level



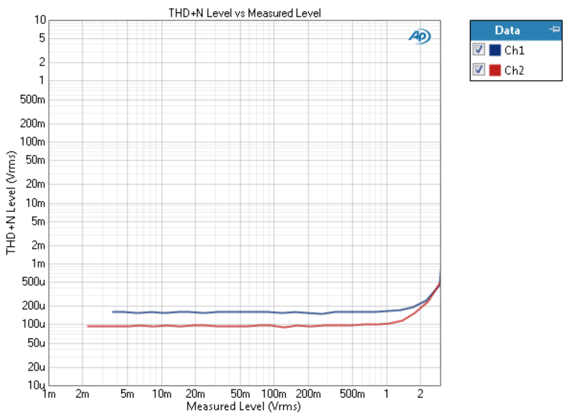
The Stepped Level Sweep: THD+N Ratio vs. Measured Level result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is measured DUT output level; Y axis is DUT output THD+N ratio.

Units

Units available for THD+N Ratio vs. Measured Level are:

	X-axis (digital)	*Y-axis*
• Vrms		
• dBV	• dBFS	• x/y
• dBu	• FS	• %
• dBrA	• %FS	• ppm
• dBrB	• dBrA	• dB
• dBSPL1	• dBrB	
• dBSPL2	• dBSPL1	
• dBm	• dBSPL2	
• W (watts)		

Stepped Level Sweep: THD+N Level vs. Measured Level



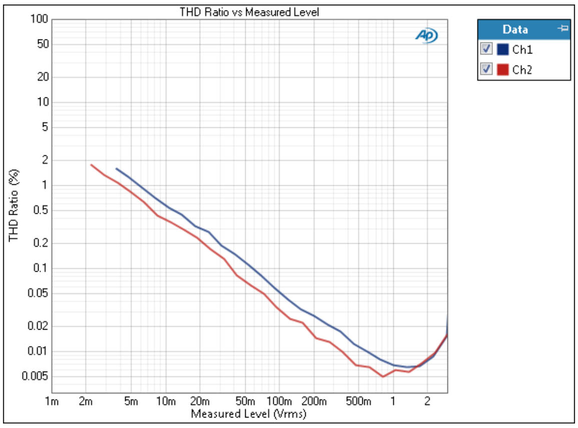
The Stepped Level Sweep: THD+N Level vs. Measured Level result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is measured DUT output level; Y axis is DUT output THD+N level.

Units

Y-axis units available for THD+N Level vs. Measured Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis (analog)*	*Y-axis (digital)*
• Vrms	• dBFS	• Vrms	• dBFS
• dBV	• FS	• dBV	• FS
• dBu	• %FS	• dBu	• %FS
• dBrA	• dBrA	• dBrA	• dBrA
• dBrB	• dBrB	• dBrB	• dBrB
• dBSPL1	• dBSPL1	• dBSPL1	• dBSPL1
• dBSPL2	• dBSPL2	• dBSPL2	• dBSPL2
• dBm		• dBm	
• W (watts)		• W (watts)	

Stepped Level Sweep: THD Ratio vs. Measured Level



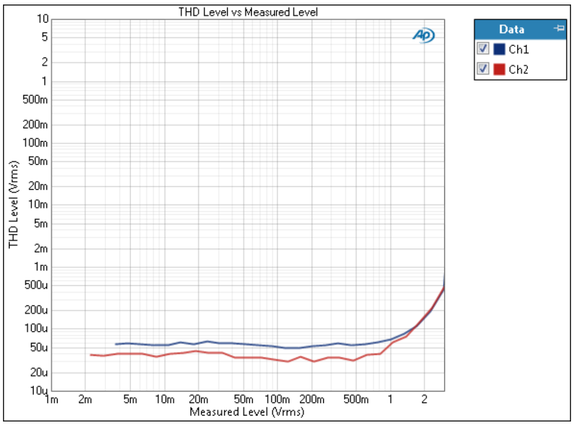
The Stepped Level Sweep: THD Ratio vs. Measured Level result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is measured DUT output level; Y axis is DUT output THD ratio.

Units

Units available for THD Ratio vs. Measured Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis*
• Vrms	• dBFS	• x/y
• dBV	• FS	• %
• dBu	• %FS	• ppm
• dBrA	• dBrA	• dB
• dBrB	• dBrB	
• dBSPL1	• dBSPL1	
• dBSPL2	• dBSPL2	
• dBm		
• W (watts)		

Stepped Level Sweep: THD Level vs. Measured Level



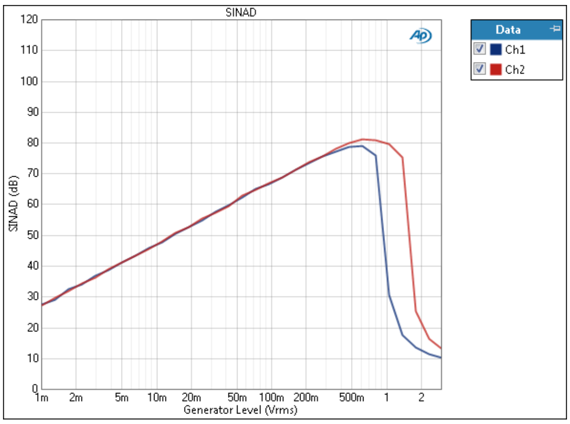
The Stepped Level Sweep: THD Level vs. Measured Level result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is measured DUT output level; Y axis is DUT output THD level.

Units

Y-axis units available for THD Level vs. Measured Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis (analog)*	*Y-axis (digital)*
• Vrms	• dBFS	• Vrms	• dBFS
• dBV	• FS	• dBV	• FS
• dBu	• %FS	• dBu	• %FS
• dBrA	• dBrA	• dBrA	• dBrA
• dBrB	• dBrB	• dBrB	• dBrB
• dBSPL1	• dBSPL1	• dBSPL1	• dBSPL1
• dBSPL2	• dBSPL2	• dBSPL2	• dBSPL2
• dBm		• dBm	
• W (watts)		• W (watts)	

Stepped Level Sweep: SINAD



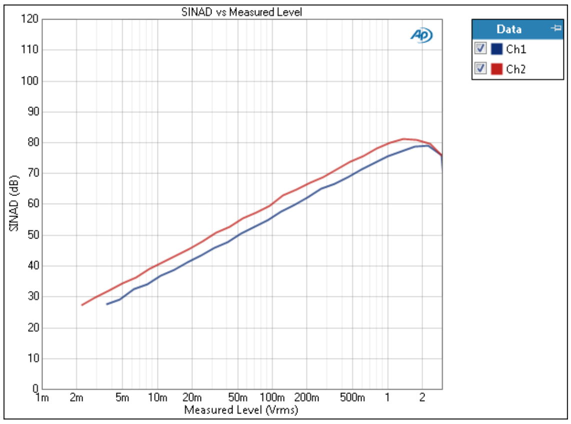
The Stepped Level Sweep: SINAD result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is generator level; Y axis is DUT output SINAD ratio.

Units

Units available for SINAD are:

X-axis (analog)	*X-axis (digital)*	*Y-axis*
• Vrms	• dBFS	• x/y
• Vp	• FS	• dB
• Vpp	• %FS	
• dBV	• dBrG	
• dBu		
• dBrG		
• dBm		
• W (watts)		

Stepped Level Sweep: SINAD vs. Measured Level



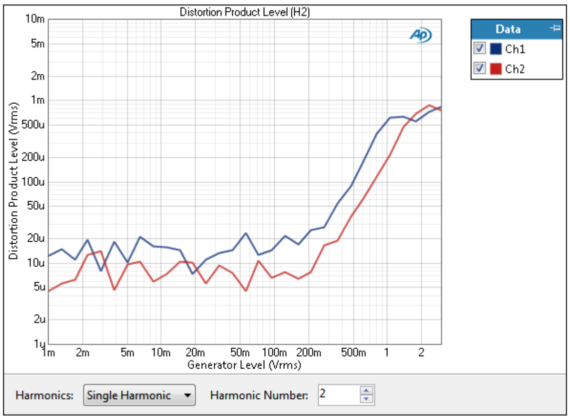
The Stepped Level Sweep: SINAD vs. Measured Level result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is measured level; Y axis is DUT output SINAD ratio.

Units

Units available for SINAD vs. Measured Level are

X-axis (analog)	*X-axis (digital)*	*Y-axis*
• Vrms	• dBFS	• x/y
• dBV	• FS	• dB
• dBu	• %FS	
• dBrA	• dBrA	
• dBrB	• dBrB	
• dBSPL1	• dBSPL1	
• dBSPL2	• dBSPL2	
• dBm		
• W (watts)		

Stepped Level Sweep: Distortion Product Level



The Stepped Level Sweep Distortion Product Level result provides a graphical display of the selected harmonic distortion products present in each channel. In this result the level of the selected harmonic distortion product in the DUT output is plotted against generator level.

Select the harmonic distortion product(s) you wish to view using the **Harmonics** controls, described below.

Harmonics

For a graph of the level of one specific harmonic product, select **Single Harmonic** and set the **Harmonic Number** to your choice, from 2nd harmonic to 20th harmonic.

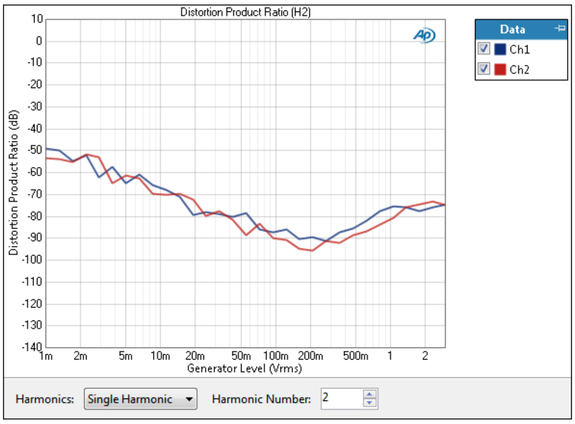
For a graph of the level of the sum of several or all harmonic products (from **H2** through **H20**), select **Sum of Harmonics**, then click **Select Harmonics...** to open a result definition dialog.

Units

Units available for Stepped Level Sweep Distortion Product Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis (analog)*	*Y-axis (digital)*
• Vrms	• dBFS	• Vrms	• dBFS
• Vp	• FS	• dBV	• FS
• Vpp	• %FS	• dBu	• %FS
• dBV	• dBrG	• dBrA	• dBrA
• dBu		• dBrB	• dBrB
• dBrG		• dBSPL1	• dBSPL1
• dBm		• dBSPL2	• dBSPL2
• W (watts)		• dBm	
		• W (watts)	

Stepped Level Sweep: Distortion Product Ratio



The Stepped Level Sweep Distortion Product Ratio result provides a graphical display of the selected harmonic distortion products present in each channel. In this result the ratio of the level of the selected harmonic distortion product to the fundamental in the DUT output is plotted against generator level.

Select the harmonic distortion product(s) you wish to view using the **Harmonics** controls, described below.

Harmonics

For a graph of the ratio of one specific harmonic product to the signal level, select **Single Harmonic** and set the **Harmonic Number** to your choice, from 2nd harmonic to 20th harmonic.

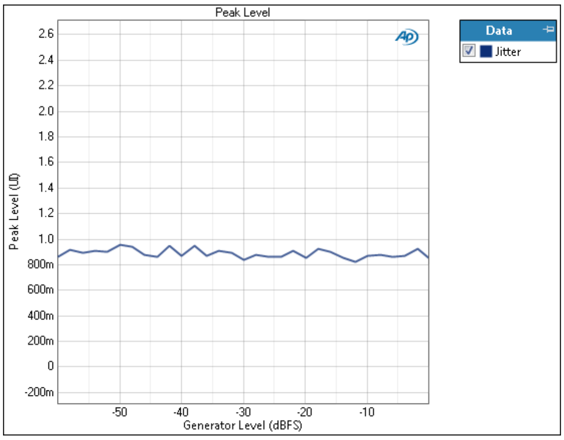
For a graph of the sum of the levels of any combination of harmonic products (from **H2** through **H20**), divided by the signal level, select **Sum of Harmonics**. Then click **Select Harmonics...** to open a result definition dialog.

Units

Units available for Stepped Level Sweep Distortion Product Ratio are:

X-axis (analog)	*X-axis (digital)*	*Y-axis*
• Vrms	• dBFS	• x/y
• Vp	• FS	• %
• Vpp	• %FS	• ppm
• dBV	• dBrG	• dB
• dBu		
• dBrG		
• dBm		
• W (watts)		

Peak Level



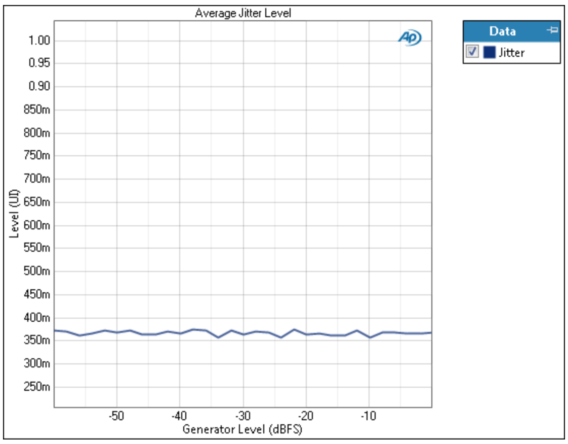
The **Stepped Level Sweep: Peak Level** result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is generator level; Y axis is the bandpass filtered DUT output level, scaled in peak units.

Units

Units available for Peak Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis (analog)*	*Y-axis (digital)*
• Vrms	• dBFS	• V	• D
• Vp	• FS		
• Vpp	• %FS		
• dBV	• dBrG		
• dBu		*Y-axis (jitter)*	
• dBrG		• UI	
• dBm		• dBUI	
• W (watts)		• s	

Average Jitter Level



The **Stepped Level Sweep: Average Jitter Level** result uses a sine wave stimulus signal that is moved across a range of levels in a series of points. X axis is generator level; Y axis is the bandpass filtered DUT output jitter level, scaled in average units.

Units

Units available for Average Jitter Level are:

X-axis (analog)	*X-axis (digital)*	*Y-axis (jitter)*
• Vrms	• dBFS	• UI
• Vp	• FS	• dBUI
• Vpp	• %FS	• s
• dBV	• dBrG	
• dBu		
• dBrG		
• dBm		
• W (watts)		

78

THD+N (Sequence Mode)

The THD+N measurement provides single-value results that show the THD+N (total distortion plus noise) in the output signal from each DUT channel, as measured at each analyzer input. The THD and Noise values are also reported in separate results. For more information about THD+N go to page 475.

THD+N results available in APx500 are:

- THD+N Ratio
- THD+N Level
- THD Ratio
- THD Level
- Noise Ratio
- Noise Level
- Distortion Product Level
- Distortion Product Ratio

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. See Chapter 6 for information about Signal Path Setup.

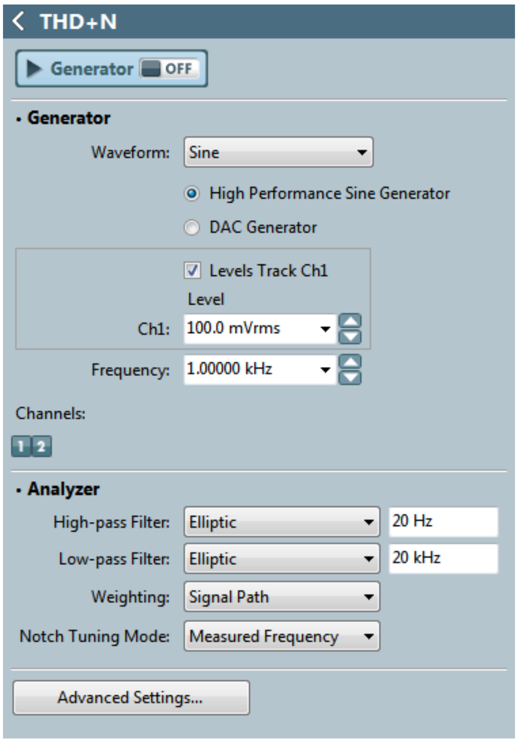
Running the measurement

To measure THD+N Ratio, click the Generator button to On. The generator will output the test signal to the DUT on the selected generator channels. Read the THD+N Ratio of each channel from the meter bar display.

Generator

The typical configuration for this measurement is the closed-loop configuration, using the APx generator as a stimulus. Alternatively, you can make open-loop measurements, using either the External Source or File Input configuration.

See Chapter 5 for detailed information about the using the APx Generator and making Waveform, Level and Frequency settings, or for information about using External Source.



High Performance Sine Generator / DAC Generator

For an APx555, you have the option of using the High Performance Sine Generator (very low distortion sine waves and wide bandwidth) or the DAC generator, which provides special sine waveforms. See High Performance Sine Generator starting on page 49 for a comparison of features.

Analyzer

Filters

Local high-pass, low-pass and weighting filters are available for this measurement. See Measurement Fil-

ters on page 547 for detailed information about the filters available locally for some measurements.

Low pass, high pass and weighting Signal Path Filters settings (see page 545) made in Signal Path Setup > Input/Output will also affect this measurement. When set, the local low-pass or high-pass filter setting overrides the corresponding Signal Path filter setting while the measurement is active. A local weighting filter setting does not override any Signal Path filter settings; the local weighting filter is applied in addition to any active Signal Path filters.

Notch Tuning Mode

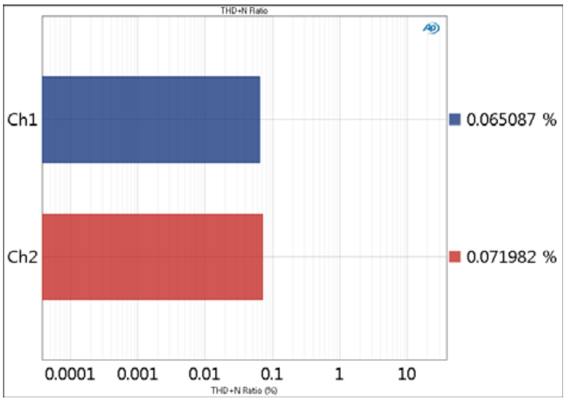
For the harmonic distortion results, a tunable notch filter is used to remove the fundamental from the measurement. The filter can be tuned to:

- Generator Frequency
  The current APx audio generator frequency. When the generator channels are outputting different frequencies (Split Frequency generation), the notch filter center is set to Frequency A. This mode is not available when using a generator waveform file.
- Jitter Generator Frequency
  The current APx jitter generator frequency, when jitter generation is available and enabled. Read more about jitter beginning on page 60.
- Measured Frequency
  The current measured frequency. When the analyzer channels are receiving different frequencies, the notch filter for each channel is centered on the frequency in that channel.
- Fixed Frequency
  A fixed frequency selected by the user. When **Fixed Frequency** is selected, a **Filter Frequency** entry field becomes available beneath the **Notch Tuning Mode** control.

Advanced Settings

If your test requires individual generator settings for each channel or special analyzer ranging or settling, click **Advanced Settings**. See Advanced Settings for single value measurement on page 317.

THD+N Ratio



The THD+N Ratio result provides a single-value measurement of the THD+N (total harmonic distortion plus noise) ratio in the output signal from each DUT channel.

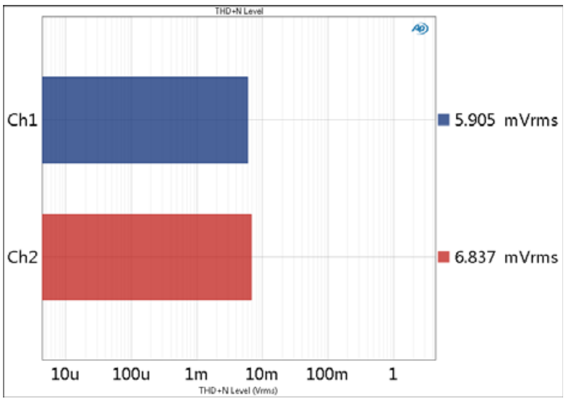
Units

Units available for THD+N Ratio are

- x/y
- %
- ppm
- dB

See Chapter 98 for more information about units of measurement.

THD+N Level



The THD+N Level result provides a single-value measurement of the THD+N (total harmonic distortion plus noise) level in the output signal from each DUT channel.

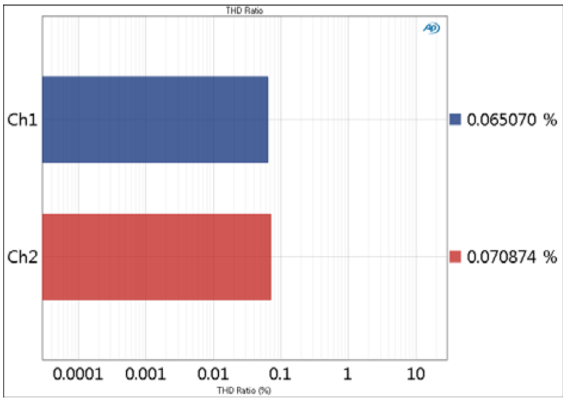
Units

Units available for THD+N Level are

Analog Signals	*Digital Signals*
• Vrms	• dBFS
• dBV	• FS
• dBu	• %FS
• dBrA	• dBrA
• dBrB	• dBrB
• dBSPL1	• dBSPL1
• dBSPL2	• dBSPL2
• dBm	
• W (watts)	

See Chapter 98 for more information about units of measurement.

THD Ratio



The THD Ratio result provides a single-value measurement of the THD (total harmonic distortion) ratio in the output signal from each DUT channel.

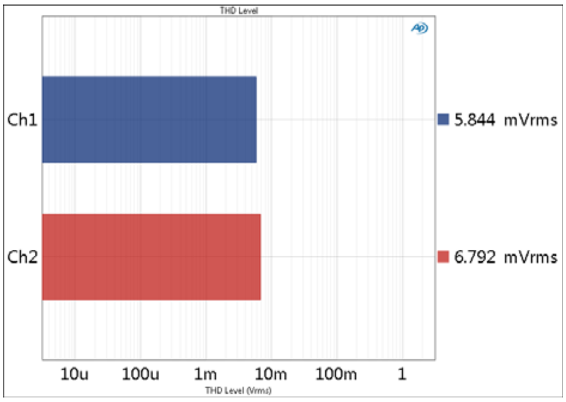
Units

Units available for THD Ratio are

- x/y
- %
- ppm
- dB

See Chapter 98 for more information about units of measurement.

THD Level



The THD Level result provides a single-value measurement of the THD (total harmonic distortion) level in the output signal from each DUT channel.

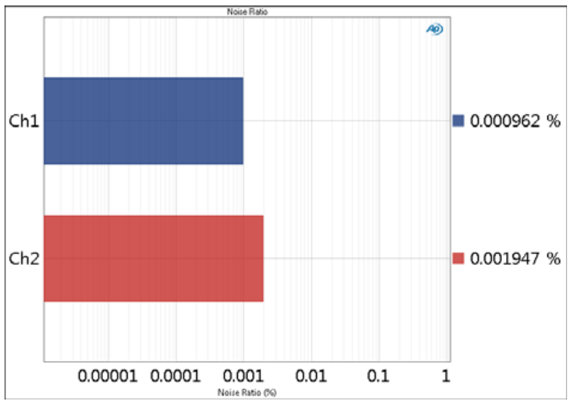
Units

Units available for THD Level are

Analog Signals	*Digital Signals*
• Vrms	• dBFS
• dBV	• FS
• dBu	• %FS
• dBrA	• dBrA
• dBrB	• dBrB
• dBSPL1	• dBSPL1
• dBSPL2	• dBSPL2
• dBm	
• W (watts)	

See Chapter 98 for more information about units of measurement.

Noise Ratio



The Noise Ratio result provides a single-value measurement of the Noise ratio in the output signal from each DUT channel.

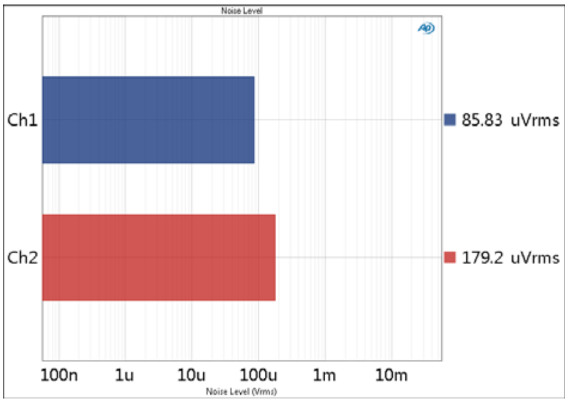
Units

Units available for Noise Ratio are

- x/y
- %
- ppm
- dB

See Chapter 98 for more information about units of measurement.

Noise Level



The Noise Level result provides a single-value measurement of the Noise level in the output signal from each DUT channel.

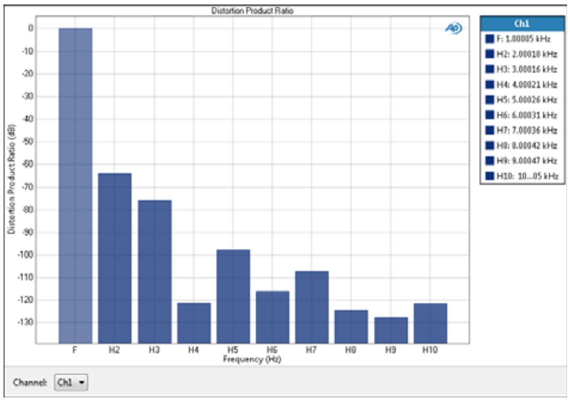
Units

Units available for Noise Level are

Analog Signals	*Digital Signals*
• Vrms	• dBFS
• dBV	• FS
• dBu	• %FS
• dBrA	• dBrA
• dBrB	• dBrB
• dBSPL1	• dBSPL1
• dBSPL2	• dBSPL2
• dBm	
• W (watts)	

See Chapter 98 for more information about units of measurement.

THD+N Distortion Product Ratio



The THD+N Distortion Product Ratio result displays a single-value measurement of the ratio of a distortion product (single harmonic) to the total signal for a selected channel, for the harmonics 2–10. The level displayed in each bar is the rms level of that harmonic tone divided by the rms level of the total signal.

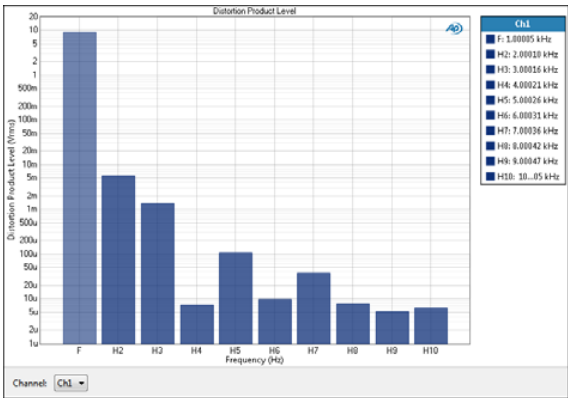
Units

Units available for THD+N Distortion Product Ratio are

- x/y
- %
- ppm
- dB

See Chapter 98 for more information about units of measurement.

THD+N Distortion Product Level



The THD+N Distortion Product Level result displays a single-value measurement of the rms level of distortion products (harmonics) for the harmonics 2–10. The level displayed in each bar is the rms level of that harmonic tone.

Units

Units available for THD+N Distortion Product Level are

Analog Signals	*Digital Signals*
• Vrms	• dBFS
• dBV	• FS
• dBu	• %FS
• dBrA	• dBrA
• dBrB	• dBrB
• dBSPL1	• dBSPL1
• dBSPL2	• dBSPL2
• dBm	
• W (watts)	

See Chapter 98 for more information about units of measurement.

More about THD+N

THD+N

THD+N is an abbreviation for Total Harmonic Distortion plus Noise. Total Harmonic Distortion is the sum of all the harmonic distortion products in a measurement, as opposed to selective harmonic distortion, which measures discrete harmonic products separately.

THD+N is the most common distortion measurement for audio signals. In the APx500 single value THD+N result, the measurement is performed in the conventional manner: a single sine wave stimulus tone is applied to the DUT input(s). In analysis, the bandwidth is usually controlled using filters; the stimulus tone is removed, and the signals that remain are reported as one measurement. These remainders are the total harmonic distortion (all harmonics within the bandwidth), noise from the DUT, and any other signals such as hum, spurious signals and other distortion products.

Because of its lack of discrimination, THD+N is not always the most useful measurement for troubleshooting; however, it is an excellent and trusted quick check on several aspects of a system's overall performance.

For THD+N Level, the measured distortion plus noise (the signal with the stimulus tone removed) is expressed as an rms level.

For THD+N Ratio, the rms level of the measured distortion plus noise (the signal with the stimulus tone removed) is divided by the rms level of the total signal. The result displayed on a bar meter. THD+N ratio is most often stated in as a percentage or as a decibel value, where 0 dB represents the total signal.

Also see More about THD (Total Harmonic Distortion, without noise results) on page 216.

THD+N Distortion Product Meter bar results

The harmonic distortion product results offer much more detail into the makeup of the harmonic distortion. For each channel, the rms level of stimulus tone is displayed as f, the fundamental, followed by the rms level of the harmonics 2f through 10f.

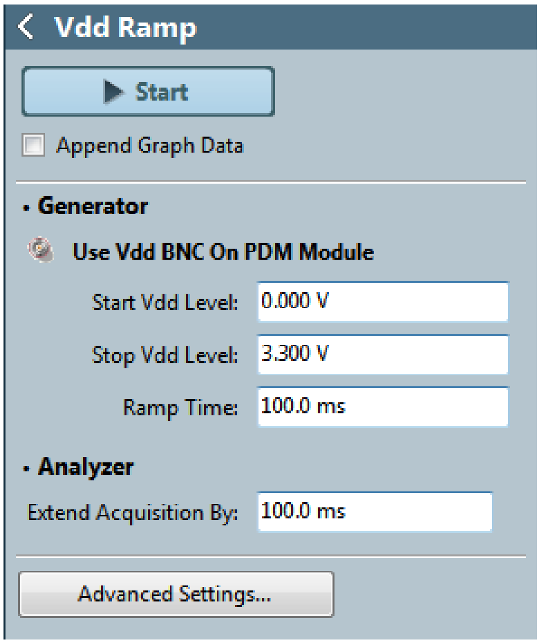
In the Distortion Product Level result, the bars show rms level. In the Distortion Product Ratio result, the bars represent the ratio of the rms level of the distortion product to the rms level of the entire signal.

79

Vdd Ramp
(Sequence Mode)

This measurement requires the PDM Option (see page 9).

Note: The Vdd Ramp measurement can sweep the Vdd voltage to 0 VDC. However, early PDM Option modules do not support Vdd voltages below 0.5 Vdc. If an early PDM Module is fitted in your instrument, the Vdd Ramp measurement will not be available. Contact your Audio Precision representative to upgrade your PDM Option module.



Overview

Vdd Ramp is measurement that characterizes the audio output of a device while its operating power is swept from one value to another. The range of voltage and the current capability are both modest, designed to provide swept operating voltage to a small integrated circuit. It is typically used with DUTs that are PDM transmitters.

Other powered DUTs, such as converter or opamp devices, can be tested with Vdd Ramp, as long as the Vdd voltage and current limits of the PDM Vdd output are sufficient for the DUT. For DUTs that require more current, a unity gain DC coupled amplifier can be inserted between the APx generator and the DUT.

Unique Characteristics

The Vdd Ramp measurement has some features notable in APx500:

- Availability
  The Vdd Ramp measurement is only available if the PDM Option is installed. See the Note above regarding early PDM Option modules.
- Vdd Supply
  When the Vdd measurement is started, DC power is applied to the PDM Vdd Supply connector, regardless of the current Signal Path settings.
  When the measurement is completed, DC power at the PDM Vdd Supply connector reverts to the Signal Path setting.
- APx Generator
  For a Vdd Ramp measurement, the output of the APx Generator does not appear at any analog or digital audio output. The analyzer output connector as specified in Signal Path Setup is irrelevant.

Append Graph Data

Measured data are grouped in a **Data Set**, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 573.

Generator

When this measurement is selected, the PDM Vdd Supply is switched to **Off**, if currently **On**. Click the **Start** button to begin the measurement. The Vdd Supply is switched to **On**, and the dc voltage on the Vdd connector is swept from the **Start Value** to the **Stop** Value, in a linear sweep which has the duration set in Ramp Time.

Start Vdd Level

Set the Vdd start voltage here. Start Value can be lower or higher than Stop Value.

Stop Vdd Level

Set the Vdd stop voltage here. The Vdd voltage remains at this level until the end of the time set in Extend Acquisition By. Stop Value can be lower or higher than Start Value.

Ramp Time

Set the duration of the linear sweep here.

Analyzer

Extend Acquisition By

Extend the acquisition by time set in this field. The enables viewing the stability of the DUT output for a period after the Vdd operating voltage has reached its maximum (Stop Vdd Level). The measurement ends at after the time set here.

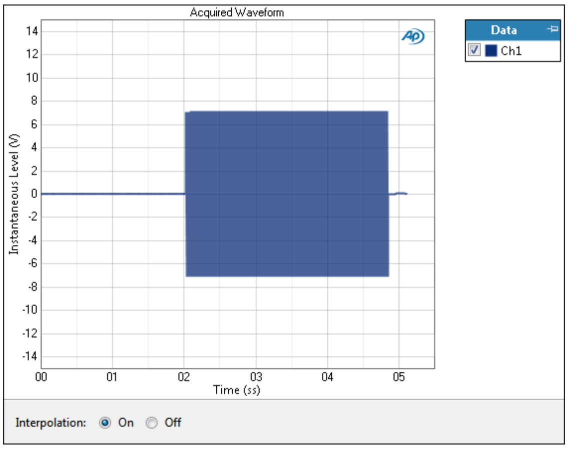
Result

There is one result for Vdd Ramp, the time-domain acquisition from the input selected in Signal Path Setup.

Advanced Settings

If your test requires special analyzer ranging, see Advanced Settings, following. See Chapter 98 for more information about units of measurement.

Acquired Waveform Result



Units

Units available for Vdd Ramp are

X-axis	***Y-axis (analog)***	***Y-axis (digital)***
• s	• V	• D • hex

Advanced Settings for Vdd Ramp

Signal Acquisition and Analysis

Input Range tab

Digital inputs do not require ranging, and the Input Range tab is unavailable when inputs are set to a digital format.

If **Auto Range** is checked (the default), each analog channel determines its input range automatically, based on the level of the input signal. If the input signal level changes beyond a ranging threshold, Auto Range will cause the input ranging circuits to move up or down for proper ranging.

Read more about ranging in Chapter 92.

If **Auto Range** is unchecked, you can set a fixed range for each analog input channel. If **Track first channel range** is checked (the default), the fixed input range setting for channel 1 is copied to the other channels, and range settings for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels.

To set individual analog input channel ranges, **uncheck** the “Track first channel range” checkbox and enter values in the input channel **Minimum Range** fields.

80

Automation

Introduction

Using the built-in Sequencer in Sequence Mode, APx500 measurements can be run in a programmed sequence to automate repeated tasks, to ensure consistent methods and to evaluate a series of devices against pass / fail limits.

You can also control APx500 using the application programming interface (API). The APx500 API is discussed briefly beginning on page 495. Detailed API documentation and API examples are provided on the Audio Precision Web site at ap.com.

The Sequencer

The Sequencer is a flexible and powerful automation interface that allows you to choose any combination of signal paths and measurements to be included in an automated sequence. You can determine the order in which the measurements are run, require satisfaction of pass / fail limits, add steps for prompts, user input, and external program execution and create a detailed report of settings and settled results.

See Chapter 94 for more information about pass / fail limits, and Chapter 81 for more information about reports.

The Sequencer is controlled from the Navigator panel, using the tree interface to select signal paths and measurements, display pass/fail markers and access each measurement's prompts and sequence properties.

Starting a sequence

Select the measurements to be included, and click Start Sequence or select Project > Start Sequence. A Running Sequence dialog will open to show the progress of the sequence. Click Cancel to quit the sequence.

Settling in a sequence

For those measurements that can be settled, the results are settled for reporting as the sequence is run. When the sequence is completed, a report is generated. Go to Chapter 92 for detailed information about settling.

At least one measurement must be selected to run a sequence.

Starting a sequence at a selected measurement

Instead of running an entire sequence, you can select a measurement in the sequence and start the sequence from that point.

Report data replacement

If there is existing report data from a previous sequence, the new data will replace the old data if the selected measurements exist in the previous sequence. If the new measurements are not in the previous sequence, their results will be added to the report. Report data is organized in the same order as measurements in the current signal path.

Right-click the measurement you would like to select as a starting point, and select Start Sequence From Here; or, highlight the measurement and from the Main menu select Project > Start Sequence from Selected Measurement.

Starting only a selected measurement

You can select one measurement, and run it as if it were in a sequence. Results will be settled and a report will be created.

Report data replacement

If there is existing report data from a previous sequence, the new data will replace the old data if the selected measurement exists in the previous sequence. If the new measurement is not in the previous sequence, its results will be added to the report. Report data is organized in the same order as measurements in the current signal path.

Right-click the measurement you would like to start, and select **Start Selected Measurement**; or, highlight the measurement and from the Main menu select **Project > Start Selected Measurement**.

Report Data

All the settings and results of a sequence, called the report data, are stored in temporary memory. If an entire sequence is run again, the memory is flushed and all the report data is replaced by data from the new sequence. If a partial sequence is run (see above), only the data for the selected measurements is replaced.

You can clear the report data manually. Right-click on the Report node at the bottom of the Navigator tree, and select **Clear Report Data**.

Data Output

The Data Output feature sends certain sequence results to a CSV file as tabular data. Each time the sequence is run, the target file is opened, and the new data is appended to the table.

This allows you to view and organize sequence results in a spreadsheet or database program.

Enable **Data Output** by setting a checkmark in the **Data Output** node at the bottom of the Navigator tree. You can specify the CSV file name and location by right-clicking on the **Data Output** node to open the **Data Output Properties** dialog.

Data Output reports date, start time, device ID, limits, sequence pass or fail and meter values. XY graph values are not reported.

Opened in a spreadsheet, a typical series of sequences might follow this pattern.

	A	B	C	D	E	F	G	H	I
1	Data	Date	Start Time	Device ID	Sequence Result	M1 Name	M2 Name	M3 Name	M4 Name
2	Upper Limit					x	x	x	x
3	Lower Limit					x	x	x	x
4	Sequence Run	11/14/2011	13:50:01		PASS	y	y	y	y
5	Sequence Run	11/14/2011	13:51:01		PASS	y	y	y	y
6	Sequence Run	11/14/2011	13:52:01		PASS	y	y	y	y
7	Sequence Run	11/14/2011	13:53:01		PASS	y	y	y	y
8	Sequence Run	11/14/2011	13:54:01		PASS	y	y	y	y
9									

Device ID

To have the operator enter a device ID (typically, the device serial number) for a sequence, enable a **Device ID** prompt at the start of the sequence. See **Project/Sequence Properties: Start** tab for more information. If the **Device ID** prompt is enabled and **Automatically Generate File Name** is set in the **Report Properties Auto Save** tab, the **Device ID** will be used in the report filename.

Note: in this case, avoid the characters \ / : * ? " < > | in the **Device ID**. These are illegal characters in a Windows file name, and APx will replace these with underscores "_" before saving.

Sequence Properties settings

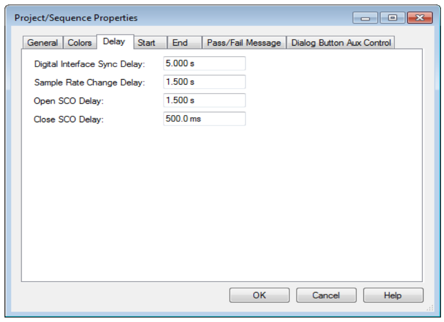
You can specify a set of properties that affect an entire sequence, and in particular the sequence's relationship with Aux Control. See Chapter 96 for more information about Aux Control.

Sequence properties settings share a dialog with project properties settings. Access the dialog from the Main menu by selecting **Project > Project/Sequence Properties**, or by choosing **Project/Sequence Properties** from the **Project** context menu (right click on the Navigator **Project** node). You can also access this dialog by clicking **Project Colors...** in the **Channel Labels** dialog. See page 54.

The **Project/Sequence Properties** dialog contains several tabbed pages. The **General** and **Colors** tabs offer global, project-wide settings, and are discussed beginning on page 32.

The Delay, Start, End, Pass/Fail Message and Dialog Button Aux Control tabs offer sequence settings and are discussed next.

Sequence Properties: Delay tab



Digital devices may require some time to establish or re-establish synchronization. When a digital configuration is changed during a sequence, delay time must be added to the sequence execution to allow the devices to synchronize.

Digital Interface Sync Delay

This delay is inserted into a sequence when a signal path that is configured with a digital input or output is activated. Default delay is 5 seconds.

Sample Rate Change Delay (digital output active)

This delay is inserted into a sequence when the output configuration is digital and a measurement with a different sample rate is activated. Default delay is 1.5 seconds.

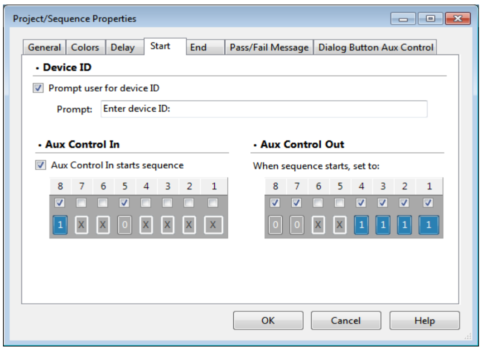
Open SCO Delay (Bluetooth Option installed and active)

This delay is inserted after a Bluetooth **Open SCO** action is sent.

Close SCO Delay (Bluetooth Option installed and active)

This delay is inserted after a Bluetooth **Close SCO** action is sent.

Sequence Properties: Start tab



Starting a Sequence with an Aux Control IN command

When this control is enabled, an **Aux Control In** command that satisfies the pattern set here will start the sequence.

Check the **Aux Control In Starts Sequence** checkbox to enable this control.

Then, set the desired **Aux Control In** pattern in the control display. To do this, activate each bit you are interested in by setting a checkmark under the bit. Then let the desired state (0 or 1) for each activated bit. The state of bits that are not activated (unchecked, state shown as “X”) will be ignored for this control.

In the illustration above, the sequence will start when **Aux Control In** bit 8 is high (1) and bit 5 is low (0). The states of bits 7, 6, 4, 3, 2, and 1 are ignored.

Setting the initial state of Aux Control Out when a sequence starts

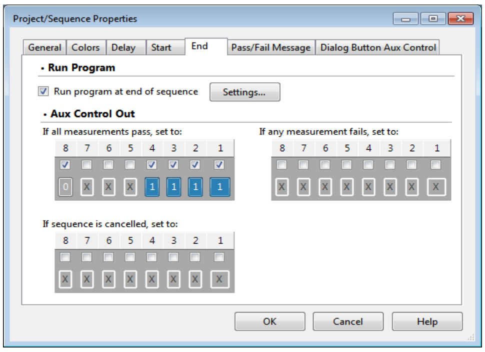
If you are using **Aux Control Out** in a sequence, the initial state of the **Aux Control Out** bits must be set to the desired pattern. If a sequence is repeated, the **Aux Control Out** bits must be reset to this pattern at the start of the repeating sequence. This control allows you to specify the initial state of **Aux Control Out** for the start of the sequence.

Check the **When Started, Set To** checkbox to enable this controls.

Set the desired **Aux Control Out** pattern in the control display. To do this, activate each bit you are interested in by setting a checkmark under the bit. Then set the desired state (0 or 1) for each activated bit. The state of bits that are not activated (unchecked, state shown as “X”) will be ignored for this control.

In the illustration above, at the start of the sequence **Aux Control Out** will be set as follows: bits 8 and 7 are low (0), bits 4, 3, 2 and 1 are high (1). The states of bits 6 and 5 are ignored; their previous state is maintained.

Sequence Properties: End tab



Setting the state of Aux Control Out when a sequence ends

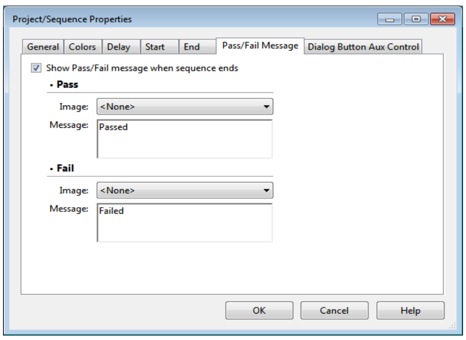
A sequence can end three ways: it is canceled, or it ends successfully with all measurements passed, or it ends successfully with at least one measurement failed. **Sequence Properties: End** allows you to send an **Aux Control Out** command for any of these sequence results.

Set the desired **Aux Control Out** pattern in the control display for any of the three end states (**All Measurements Passed, Any Measurement Failed, Sequence Cancelled**).

To do this, activate each bit you are interested in by setting a checkmark under the bit. Then set the desired state (0 or 1) for each activated bit. The state of bits that are not activated (unchecked, state shown as “X”) will be ignored for this control.

In the illustration above, when the sequence ends with all measurements passed, **Aux Control Out** will be set as follows: bits 8 and 7 are low (0), bits 4, 3, 2 and 1 are high (1). The states of bits 6 and 5 are ignored; their previous state is maintained.

Sequence Properties: Pass/Fail Message tab



An on-screen message can be shown to the operator at the end of a sequence.

To show a Pass/Fail message, click the **Show Pass/ Fail Message** checkbox.

Pass Message

If all the measurements in the sequence pass, the **Pass Message** is shown. The default message text is "Passed". Click in the message box to edit the message. You can add an image using the **Image** file browser button.

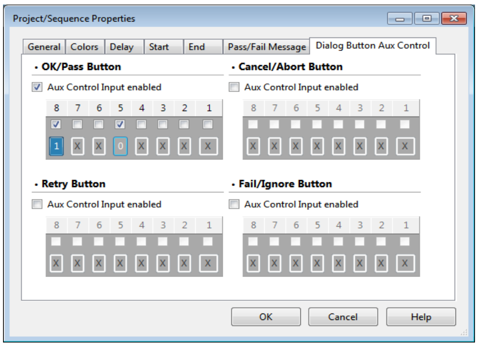
Fail Message

If one or more measurements in the sequence fails, the **Fail Message** is shown. The default message text is "Failed". Click in the message box to edit the message. You can add an image using the **Image** file browser button.

Closing the Pass/Fail Message dialog

The Pass/Fail Message dialog that is displayed to the user contains an **OK** button. The user clicks this button to close the dialog box. Alternatively, the user can close the dialog using an **Aux Control In** command. See **OK/Pass Button** in **Dialog Button Aux Control,** below.

Sequence Properties: Dialog Button Aux Control



Dialog buttons with four designations may appear in user prompts and dialogs generated during a sequence. These buttons require a response from the user, which can be accomplished by a mouse click or keyboard stroke on the automation PC, or by sending a specific signal to the Aux Control Input connector.

The four designations are:

- OK/Pass
- Retry
- Cancel/Abort
- Fail/Ignore

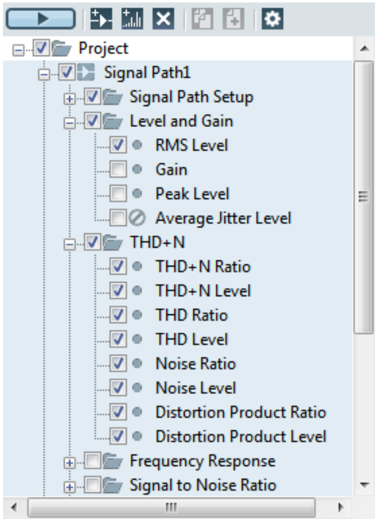
To activate any of these on-screen buttons using Aux Control, check the **Aux Control Input Enabled** checkbox for the button you wish to activate.

Then, set the desired **Aux Control In** pattern in the control display. To do this, activate each bit you are interested in by setting a checkmark under the bit. Then set the desired state (0 or 1) for each activated bit. The state of bits that are not activated (unchecked, state shown as "X") will be ignored for this control.

In the illustration above, the selected button will be activated when **Aux Control In** bit 8 is high (1) and bit 5 is low (0). The states of bits 7, 6, 4, 3, 2, and 1 are ignored.

Selecting and deselecting measurements

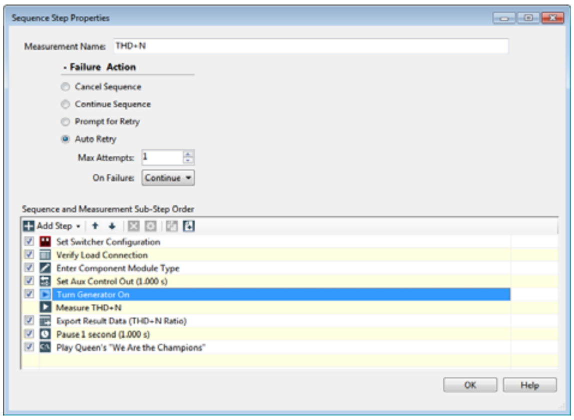
In the Navigator panel, click in the box next to any measurement to add the measurement to a sequence. A green check mark confirms your selection. Click again to remove the check mark and remove the measurement from the sequence.



If no measurements or lower branches are selected, selecting a higher branch selects all the measurements or branches that are within it. If a measurement or lower branch is already selected, selecting or deselecting a higher branch includes or excludes that branch from the sequence without changing the selection status of the measurements or branches that are within it. This allows you to choose a group of measurements in Continuous Sweep, for example, and then to include or exclude that group from a sequence with just one click.

To deselect all measurements in a branch and its lower branches, right-click on the branch and choose **Uncheck All** from the context menu.

Sequence Measurement Step Properties



Overview

Each measurement has properties associated with it that determine the behavior of the sequence step(s) for that measurement. Access the **Sequence Step Properties** dialog by selecting a measurement in the Navigator and then clicking **Edit Prompts and Properties** in the Navigator title bar, or by right-clicking on a measurement node and selecting **Edit Prompts and Properties** from the context menu.

The **Sequence Step Properties** dialog will open. At the top is a section called **Measurement Properties**, and below is a toolbar and grid displaying the **Sequence and Measurement Substep Order**.

Measurement Name

The parent measurement name from the Navigator is shown here. You can edit the name here, and your changes will be made on the Navigator branch as well.

Failure Action

When a limit (or **Auto Generator Set** action, see below) is exceeded in the selected measurement, you can direct the Sequencer to:

Cancel Sequence

Stop the sequence on a limit failure.

Continue Sequence (the default)

Flag the measurement as **Failed** and continue.

Prompt for Retry

Display a prompt indicating a limit failure, with **Cancel, Retry** and **Ignore** commands available to the operator.

Auto Retry

Retry the step up to 5 times. If the step still fails after number of attempts set in **Max Attempts**, the Sequence will either **Continue** or **Cancel**, as selected in the **On Failure** menu.

Note: If the sequence is not configured properly (calling for a file or program step that does not exist, for example), the sequence will always abort.

References

Auto Set Generator Level

For **Signal Path Setup** only, the sequence step properties includes an **Auto Set Generator Level** checkbox. When checked, this will invoke the **Auto Gen Level** feature of **Signal Path Setup > References.** If **Auto Gen Level** fails, the selected **Failure Action** will be invoked.

Copy References From

For **Signal Path Setup** only. If there is more than one **Signal Path** in the sequence, you can copy the references from another **Signal Path** into the current **Signal Path.** This allows you, for example, to copy microphone calibration settings from one **Signal Path** to another.

HDMI Source

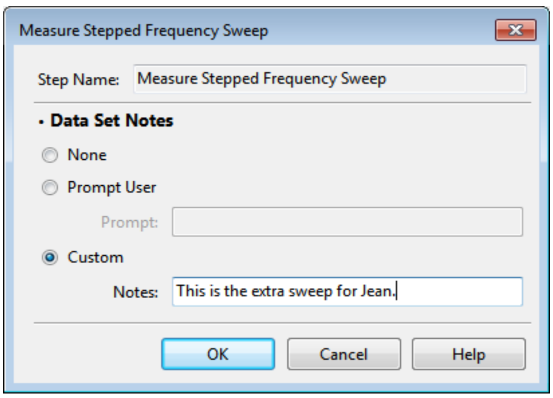
Verify DDC

For **Signal Path Setup** in **HDMI Source Output Configuration** only, the sequence step properties includes an **HDMI Source: Verify DDC** checkbox. When checked, this will verify DDC (Display Data Channel) connection with the downstream device. If DDC verification is enabled and the DDC read fails, the sequence shall fail with either a **DDC error** or a **Receiver Not Connected** error. See More About DDC and EDID on page 114.

Adding Data Set Notes to a measurement

For measurements that show Data Sets, you can cause text to be added to the Data Sets Notes field. This text will appear in the report.

Double-click on a measurement step, or right-click and select **Edit Step Properties** from the context menu. This dialog will open:



None

Choose **None** so that no Data Sets Note will appear in the report for this measurement.

Prompt User

You can enter text here that will prompt the operator to cause text to be entered in the **Notes** field while the Sequence is running. Perhaps the operator would enter an identifying number or scan a bar code in response to the prompt here.

Custom

Text you enter here will appear as a Data Sets Note in the report.

Sequence and Measurement Substep Order

You can add any number of substeps that display user prompts, require user input, run external programs, add delay, set aux control out or turn the generator on. Once added, these substeps can be enabled, disabled or deleted. Each substep's properties can be edited, and the substeps can be moved into any order.

Measurement substep

When the dialog first opens, you will see the Measurement substep for the parent measurement. This step actually invokes the parent measurement when the sequence is run.

Turn Generator On substep

For some measurements, the **Measurement** substep will be preceded by a **Turn Generator On** substep. It is not necessary to have this substep checked to run the measurement; the measurement will turn the generator ON and OFF as part of its normal operation. This substep is to turn the generator ON independent of the measurement, to allow the operator to make adjustments to the DUT or the APx generator. Place a checkmark in the box to enable the substep. Go to page 495 for more information about using this substep.

Adding Steps

Click the **Add Step** button, then select a substep type from the menu. Available substeps will vary with hardware, measurement and configuration.

- Prompt (page 484)
- User Input (page 486)
- Program (page 487)
- Delay (page 487)
- Set Digital Sync Delay (page 487)
- Appended Measurement (page 488)
- Set Aux Control Out (page 488)
- Export Result Data (page 489)
- Import Result Data (page 489)
- Import Output EQ Curve (page 489)
- PDM Control Codes (page 490)
- Bluetooth (page 491)
- Set Vdd Output (page 491)
- Set Switcher Configuration (page 491)
- Set DCX Aux Output (page 492)
- Set DCX DC Output (page 492)
- Make Result Visible (page 492)
- Send CEC Command (page 493)
- User Evaluation of Result (page 493)
- Start Analog Generator (page 493)
- Stop Analog Generator (page 494)

Each of these substeps are discussed in detail, following.

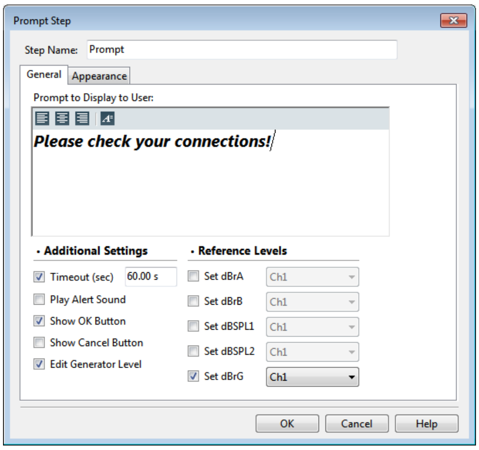
In the example above, we added a **Prompt** substep and a **User Input** substep, and then moved them above the **THD+N** measurement. Then we added the final **Program** substep at the end.

Prompt: General tab

You can add a step to any measurement in a sequence that inserts a user prompt with reply (**OK**, **Cancel**) options. Prompts allow you to provide the operator with information and the opportunity to interact with the sequence.

Right click on a measurement node in the Navigator and choose **Edit Prompts and Properties...** from the context menu. Click **Add Step**, or right-click in the list area and choose **Add Prompt...** in the context menu, and choose **Prompt...**

The following dialog will open.



Step Name:

You can name a Prompt substep to help organize your sequence.

Prompt to Display to User:

Add any text you want to appear in the prompt here. You can also paste formatted text from the Windows clipboard into the text entry field. Long text can be scrolled in the prompt.

The toolbar at the top of the text entry area provides left, center and right alignment buttons. The font style button opens a dialog where you can select a new font, change point size, color, weight, and so on.

Additional Settings

Timeout

When this box in unchecked, the prompt window will remain open until the user closes it.

When the box is checked, the value entered in the field to the right sets the maximum prompt window display time, in seconds. If the timeout is reached, the current measurement fails. Minimum timeout is 1 second, maximum is 3600 seconds (one hour).

Play Alert Sound

If the Play Alert Sound is checked, a Windows alert sound will be played through the PC. The specific sound chosen depends on the prompt icon chosen and your Windows settings. As with any sound played by Microsoft Windows, you must have a sound card installed in your PC and have the volume control and other Windows settings properly adjusted to hear the sound.

Show OK Button

Check this box to place an **OK** button on the prompt. Clicking the **OK** button closes the prompt window and continues the sequence. See Dialog Button Aux Control on page 482.

Show Cancel button

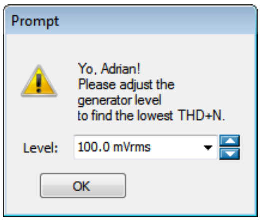
Check this box to add a Cancel button to the prompt. Clicking the Cancel button closes the prompt window and stops the sequence.See Dialog Button Aux Control on page 482.

Edit Generator Level

Not all measurements require or enable this prompt setting.

Certain measurements may require operator adjustment of the DUT or APx generator during the sequence. The **Turn Generator On** substep (page 495) must be previously set to turn the generator on. **Edit Generator Level** opens a prompt during the sequence that displays generator level controls to enable the operator to observe DUT output while making APx generator adjustments. When ready, the operator clicks **OK** to continue the sequence.

This is a sample of the Edit Generator Level prompt:



Reference Levels

These controls appear only for the **Reference Levels** measurement.

The purpose of these controls is to allow the operator to play an audio signal or make an adjustment in response to a prompt, and to save the resultant input level as a reference.

When prompted, the operator performs the required action to present a signal to the analyzer (such as playing a test signal from the DUT or using a mic calibrator), selects dBrA, dBrB, dBSPL1 or dBSPL2 and the input channel of interest, and clicks OK to save the reference(s).

If the operator has been prompted to edit the generator level (as discussed above), that level can be saved as dBrG.

Reference Levels (Acoustic Input Configuration)

When the Signal Path is set to Acoustic Input Configuration (page 167), the available input reference is dBSPL.

Prompt: Appearance tab

Right click on a measurement node in the Navigator and choose **Edit Prompts and Properties...** from the context menu. Click **Add Step,** and choose **Prompt...**

Choose the **Appearance** tab. The following dialog will open.

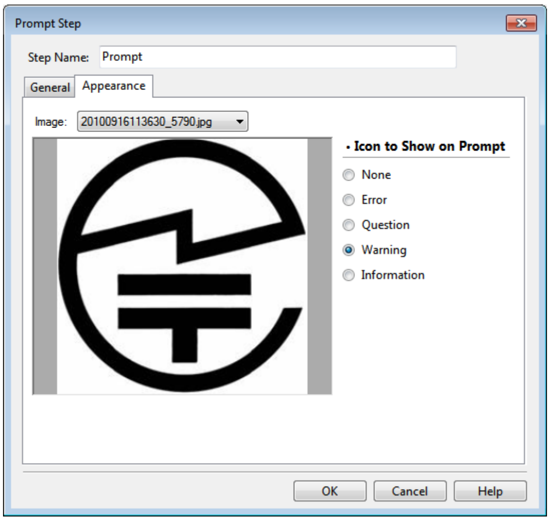


Image:

You can place an image at the top of the user prompt. This could be a photograph or diagram with additional visual information to help the user.

Click the **Image...** button to navigate to an image file. Select the image and click **Open** to place the image in the prompt. When an image is used, the image file is also attached to the project file.

Larger images will cause the prompt window size to increase to fill the available screen space, and will be scaled to fit in the prompt window.

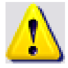
Select **<None>** to remove the image from the prompt. Image files attached to the project but not currently in use can be removed by navigating to the dialog at **File > Manage Attached Project Items** (see page 24).

Supported image file formats

- .bmp
- .jpg
- .jpeg
- .gif
- .tif
- .tiff
- .png
- .ico

Icon to Show on Prompt

Check one of the following (or None) to place a standard Windows attention icon on the prompt.

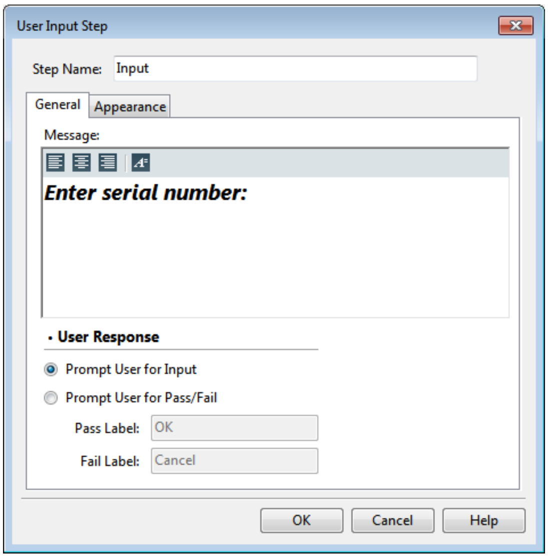
- None
-  Error
-  Question
-  Warning
-  Information

User Input (General tab)

You can add a step to any measurement in a sequence that pauses the sequence so that the operator can input information. This information can be in the form of a text string, such as a DUT serial number, or as a choice between two buttons, which can be labeled with text of your choice (PASS and FAIL, for example, or OK and CANCEL.

Right click on a measurement node in the Navigator and choose **Edit Prompts and Properties...** from the context menu. Click **Add Step**, and choose **User Input...**

The following dialog will open.



Step Name:

You can name a User Input substep to help organize your sequence. If you are requesting the serial number for input, you might name this substep “DUT serial number”. This name is also used to label the response in the sequence report.

Message:

In this field, type in the query or request to be shown to the operator on the input prompt. For example, you might request “Enter serial number.”

User Response

Choose

- **Prompt User for Input**
  if you would like the operator to respond by a text input, or
- **Prompt User for Pass/Fail**
  if you would like the operator to respond by activating a Pass or Fail on-screen button, or Aux In switch (see Dialog Button Aux Control on page 482). For this option, you can specify the on-screen label for the Pass or Fail labels.

Appearance tab

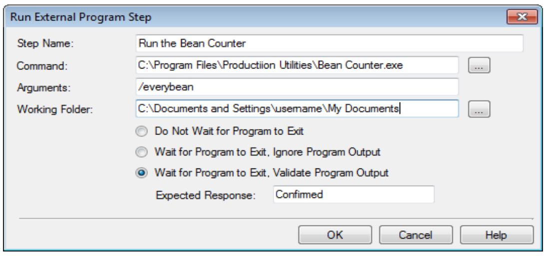
Select an optional image for the user prompt here.

Adding a Program substep

Add Program allows you to run an external program at a point in the sequence.

Click the **Add Program...**button, or right-click in the list area and choose **Add Program...** in the context menu.

The **Run External Program** dialog (shown below) will open. To edit existing **Program Step** properties, select the substep and click the **Edit Step** button, or right-click on the substep and choose **Edit Properties...** from the context menu.



Step Name:

You can name a Run External Program substep to help organize your sequence.

Command:

Type in the command for the external program here, or browse to the program using the browse button.

Arguments:

Type in the command arguments here, if any.

Working Folder:

Type in the working folder here, if necessary, or browse to the folder using the browse button.

Do Not Wait for Program to Exit

If this option is selected, the sequence will issue the command and then proceed (or, end) without waiting.

Wait for Program to Exit, Ignore Program Output

If this option is selected (the default), the sequence will pause until the program is finished.

Wait for Program to Exit, Validate Program Output

If this option is selected, the sequence will pause until the program is finished and returns a text string that matches the string entered into the **Expected Response** field.

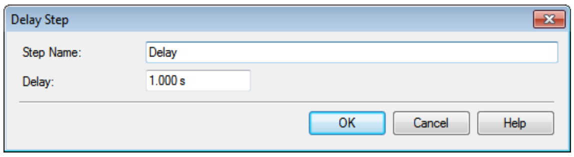
Adding a Delay substep

You can add a step to any measurement in a sequence that pauses the sequence for a specified time.

Click the Add Delay button.

Right click on a measurement node in the Navigator and choose **Edit Prompts and Properties...** from the context menu. Click **Add Step**, and choose **Delay...**

The following dialog will open:



Step Name:

You can name a Delay substep to identify the delay and help organize your sequence.

Delay:

In this field, type in the delay time in s (seconds). Minimum is 0.000 s; maximum is 300.0 s.

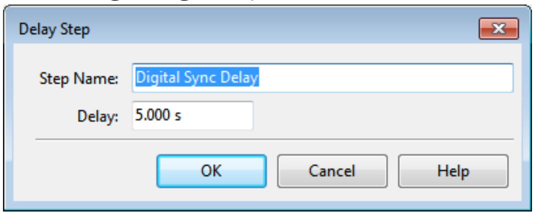
Adding a Digital Sync Delay substep

When switching to a digital input during a sequence, there may be a delay in the connection before the digital signal locks.

You can add a Digital Sync Delay step to set a maximum time to wait for digital input lock. If lock occurs before the delay time has expired, the sequence will continue immediately.

Right click on a measurement node in the Navigator and choose **Edit Prompts and Properties...** from the context menu. Click **Add Step**, and choose **Digital Sync Delay...**

The following dialog will open:



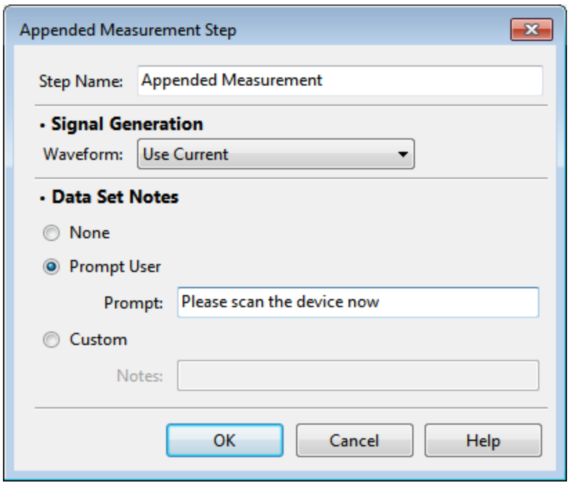
Step Name:

You can create a distinct name for this step to help organize your sequence.

Delay:

In this field, type in the delay time in s (seconds). Minimum is 0.000 s; maximum is 300.0 s.

Sequencer: Adding an Appended Measurement Step



For measurements that support **Append**, click **Add Appended Measurement** to run the measurement an additional time and append the data. You can add multiple **Add Appended Measurement** substeps to a sequence step, each invoking an additional appended measurement. This allows you to make variations on a measurement within one sequence step, rather than sequencing multiple repeated measurements.

For measurements that support more than one generator waveform choice, you can select the waveform to use for each appended measurement.

For example, in the Measurement Recorder, you may have **Split Frequency** selected as the waveform on the measurement panel, along with two generator waveform files called **Test1** and **Test2**. In the example here,

- **Use Current** would select the current waveform, **Split Frequency**;
- **Test1.wav** or **Test2.wav** would select one of the generator waveform files.

Data Sets

You can cause text to be added to the Data Sets Notes field. This text will appear in the report.

None

No Data Sets Note will appear in the report for this measurement.

Prompt User

You can enter text here that will prompt the operator to cause text to be entered in the Notes field while the Sequence is running. Perhaps the operator would enter an identifying number or scan a bar code in response to the prompt here.

Custom

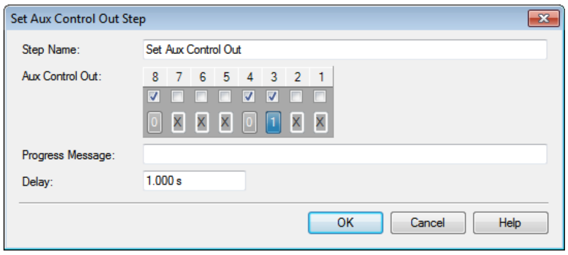
Text you enter here will appear as a Data Sets Note in the report.

Adding an Aux Control Out step

You can add a step to any measurement in a sequence to issue an Aux Control Out command, and to pause for a specified delay time after the command is issued.

Right click on a measurement node in the Navigator and choose **Edit Prompts and Properties...** from the context menu. Click **Add Step**, and choose **Aux Control Out...**

The following dialog will open.



Step Name

Enter an optional name for this step.

Aux Control Out

Set the desired Aux Control Out pattern in the control display. To do this, activate each bit you are interested in by setting a checkmark under the bit. Then set the desired state (0 or 1) for each activated bit. The state of bits that are not activated (unchecked, state shown as “X”) will be ignored for this control.

In this illustration, Aux Control Out will be set as follows: bits 8 and 2 are high (1) and bit 2 is low (0). The states of bits 7, 6, 5 and 4 are ignored; their previous state is maintained.

Progress Message

You can enter an optional text message in the Progress Message field. This message will be displayed to the operator for the length of time set in Delay, below.

Delay

An Aux Control Out command may start an external event that takes some time to complete. To wait for completion, you can enter an optional delay time here. The next substep in the sequence is not executed until the delay time has passed.

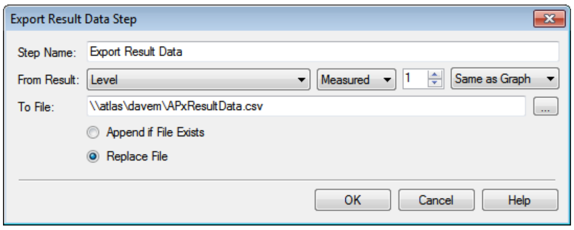
The delay time begins at the instant that the Aux Control Out command is sent.

Adding an Export Result Data step

You can add a step to any measurement in a sequence to export APx result data from that measurement.

Right click on a measurement node in the Navigator and choose **Edit Prompts and Properties...** from the context menu. Click **Add Step**, and choose **Export Result Data...**

The following dialog will open.

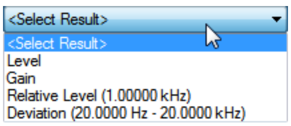


Step Name

Enter an optional name for this step.

From Result

Many measurements have more than one result. Choose the result from which you want to export data.



For XY results, you will be able to select the data set and the number of points from the additional controls (“Measured”, “2” and “Same as Graph”, above). Meter results do not have these options.

To File

Enter the path and file name here, or browse to the file. You can choose either CSV or Excel spreadsheet formats.

Append if File Exists

This option opens the file on disk, appends the current data to the end of the data in the file, and re-saves the file.

Replace File

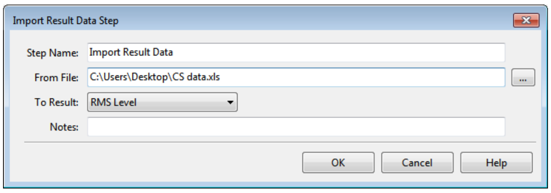
This option erases the file on disk and replaces it with a new file containing the current data.

Adding an Import Result Data step

For measurements that support data import, you can add a step in a sequence to import APx result data into that measurement.

Right click on a measurement node in the Navigator and choose Edit Prompts and Properties... from the context menu. Click Add Step, and choose Import Result Data...

The following dialog will open.



Step Name

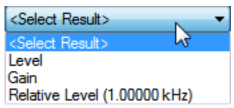
Enter an optional name for this step.

From File

Enter the path and file name here, or browse to the file.

To Result

Many measurements have more than one result. Choose the result into which you want to import data.



Be sure that the imported data and the selected result are compatible.

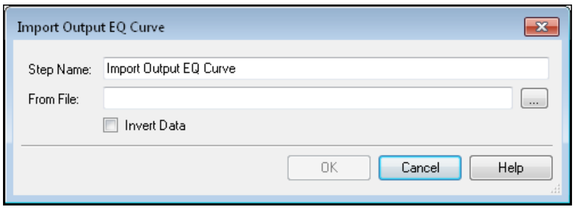
Adding an Import Output EQ Curve Step

Also see Output Equalization on page 168.

You can add a sequence step to **Signal Path Setup** to import an **Output EQ** curve. The current **Output Configuration** must be **Analog**.

Right click on the **Signal Path Setup** node in the **Navigator** and choose **Edit Prompts and Properties...** from the context menu. Click **Add Step**, and choose **Import Output EQ Curve**.

The following dialog will open.



Step Name:

Enter an optional name for this step.

From File:

Enter the path and file name here, or browse to the file.

Invert

If this checkbox is set, the EQ curve will be inverted.

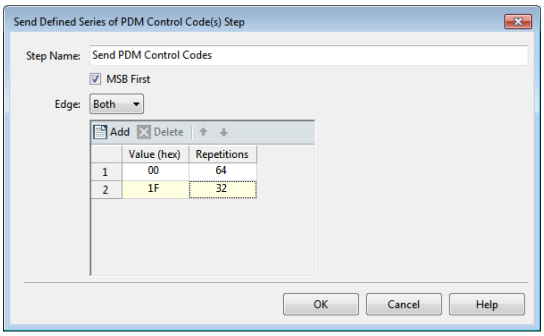
Send Defined Series of PDM Control Code(s) step

This sub-step is only available if Output Configuration for this Signal Path is set to PDM.

You can add a step to any measurement in a sequence that sends a defined series of PDM control codes.

Right click on a measurement node in the Navigator and choose **Edit Prompts and Properties...** from the context menu. Click **Add Step**, and choose **PDM Control Code(s)** > **Send Defined Series**.

The following dialog will open.



Step Name

You can create a name for this step to help organize your sequence.

MSB First

Check **MSB First** to send the Control Code MSB (Most Significant Bit) first. When **MSB First** is not checked, the MSB is sent last.

Edge

Choose **Both**, **Rising** or **Falling** to select the PDM bitstream edge to carry the Control Code data.

Grid

Add or delete PDM control codes here, and define the Value (hex) and Repetitions for each code.

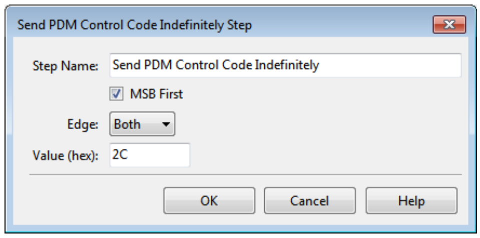
Send PDM Control Code Indefinitely step

This sub-step is only available if Output Configuration for this Signal Path is set to PDM.

You can add a step to any measurement in a sequence that sends a PDM control code indefinitely.

Right click on a measurement node in the Navigator and choose **Edit Prompts and Properties...** from the context menu. Click **Add Step**, and choose **PDM Control Code(s)** > **Send Code Indefinitely**

The following dialog will open.



Step Name

You can create a name for this step to help organize your sequence.

MSB First

Check **MSB First** to send the Control Code MSB (Most Significant Bit) first. When **MSB First** is not checked, the MSB is sent last.

Edge

Choose **Both**, **Rising** or **Falling** to select the PDM bitstream edge to carry the Control Code data.

Value (hex)

Define the **Value (hex)** for the code.

Stop PDM Control Code step

This sub-step is only available if Output Configuration for this Signal Path is set to PDM.

You can add a step to any measurement in a sequence that stops sending the PDM control code initialized by a previous **Send Code Indefinitely** step.

Right click on a measurement node in the Navigator and choose **Edit Prompts and Properties...** from the context menu. Click **Add Step**, and choose **PDM Control Code(s) > Stop Sending Code**.

Bluetooth

This sub-step is only available if Output Configuration for this Signal Path is set to Bluetooth.

Unlike other APx Output and Input Configurations, which are simply established by software selection and a cable connection, Bluetooth configurations require discovery, pairing, profile selection, etc. in an extended negotiation between the APx analyzer and the Bluetooth Device Under Test.

A number of Bluetooth sequence steps are available; however, configuring a sequence for a Signal Path utilizing Bluetooth is more complex than simply adding steps, and is discussed in detail the Bluetooth I/O chapter, beginning on page 129.

This sub-step is only when a PDM module is installed, and when neither Output Configuration nor Input Configuration for this Signal Path is set to PDM.

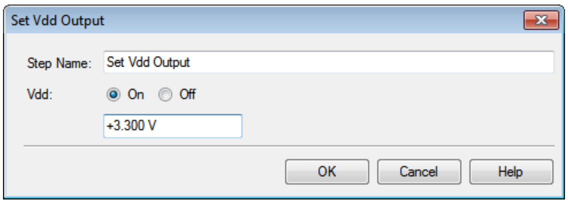
Set Vdd Output

This sub-step is available only when a PDM module is installed, and when neither Output Configuration nor Input Configuration for this Signal Path is set to PDM.

You can add a step to any measurement in a sequence that switches the Vdd dc power supply **ON** or **OFF**.

Right click on a measurement node in the Navigator and choose **Edit Prompts and Properties...** from the context menu. Click **Add Step**, and choose **Set Vdd Output...**

The following dialog will open.



Step Name

You can create a name for this step to help organize your sequence.

Vdd

The **Set Vdd Output** step feature is only available when the instrument is fitted with a PDM Option module, and when neither **Output Configuration** nor **Input Configuration** is set to PDM.

This feature enables power to be supplied to a DUT such as a MEMS microphone or an integrated circuit, while generating and/or analyzing audio in a format other than PDM.

On/Off

Choose whether this step turns the Vdd output **ON** or **OFF**.

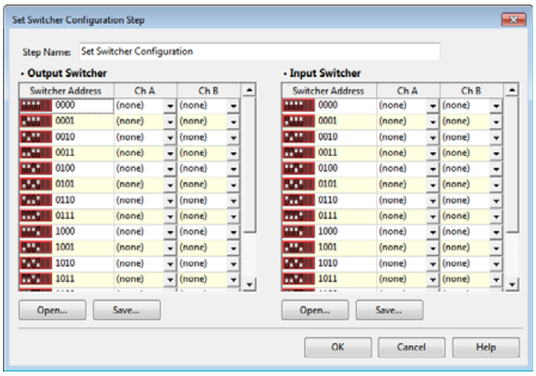
Enter the voltage required for operating power the DUT in entry field. The Vdd power is available on the PDM module at the **Vdd Supply** BNC connector, providing DC current up to 15 mA, with a voltage range of +0.8 VDC to +3.60 VDC.

Set Switcher Configuration step

In a Sequence, the initial switcher settings for each Signal Path are set in **Signal Path Setup > Switcher Settings...** for that Signal Path.

To set or change switcher settings while a Sequence is in progress, add a **Set Switcher Configuration** step.

Right click on a measurement node in the Navigator and choose **Edit Prompts and Properties...** from the context menu. Click **Add Step,** and choose **Set Switcher Configuration...** The following dialog will open:



Step Name

You can create a name for this step to help organize your sequence.

Input and Output Switcher Settings dialog

These **Input** and **Output Switcher Settings** dialog presents the same settings as the Signal Path Setup Switcher Settings dialog, discussed beginning on page 599.

Behavior

At the start of each active measurement in a sequence, the switcher configuration is set to the configuration in Signal Path Setup for the active Signal Path.

When a sequence is running, if the active measurement contains a Set Switcher Configuration step, the switcher configuration is set according to the step. The new configuration persists until the next Set Switcher Configuration step, or until the next measurement begins, when the configuration reverts to the Signal Path Setup switcher configuration as described above.

At the end of the Signal Path in a Sequence, the switcher configuration is reset to Signal Path Setup switcher configuration.

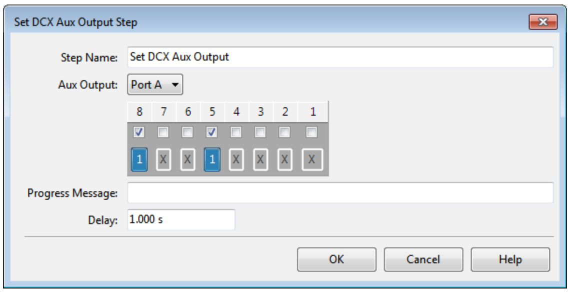
If no switchers are connected, the sequence continues normally, with no error or warning.

Set DCX Aux Output

You can add a sequence step to a measurement to make an Aux Output setting on an attached DCX-127.

Right click on a measurement node in the Navigator and choose **Edit Prompts and Properties...** from the context menu. Click **Add Step**, and choose **Set DCX Aux Output**.

The following dialog will open.



Step Name:

You can create a distinct name for this step to help organize your sequence.

Aux Output:

Choose the DCX Port from the menu, and make Aux Output settings in the display. Use a checkmark to enable a line, and set the line high or low by choosing a 1 or a 0.

Progress Message:

Enter an optional progress message here.

Delay:

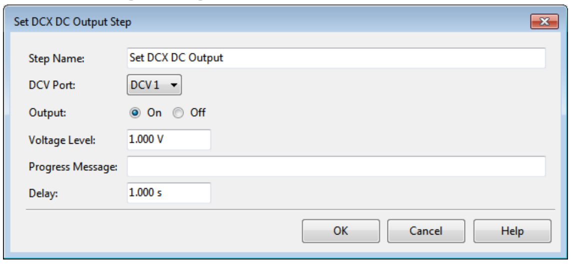
Add any delay needed for the Aux Output command and connected device to attain the required state.

Set DCX DC Output

You can add a sequence step to a measurement to set and adjust a DC Output on an attached DCX-127.

Right click on a measurement node in the Navigator and choose **Edit Prompts and Properties...** from the context menu. Click **Add Step**, and choose **Set DCX DC Output**.

The following dialog will open.



Step Name:

You can create a distinct name for this step to help organize your sequence.

DCV Port:

Choose the DCX DCV Port from the menu.

Output:

Switch the DC voltage on or off.

Voltage Level:

Set the DC voltage level here.

Progress Message:

Enter an optional progress message here.

Delay:

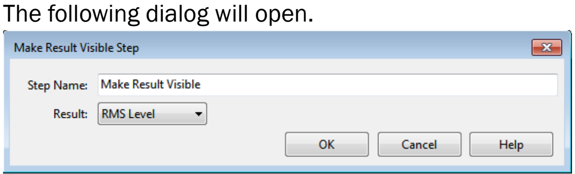
Add any delay needed for the DC Voltage Output and connected device to attain the required state.

Make Result Visible

When you run a sequence, you may have noticed that for each measurement, the most recently selected result is displayed as the measurement runs. You can add this step to cause a specific result to be displayed. This can be useful when a user must visually evaluate one or more results as a sequence runs. It may be helpful to also insert a Delay step after the Make Result Visible step to allow more time for the user to make an evaluation.

Right click on a measurement node in the Navigator and choose **Edit Prompts and Properties...** from the context menu. Click **Add Step**, and choose **Make Result Visible...**

The following dialog will open.



Step Name:

You can create a distinct name for this step to help organize your sequence.

Result:

Select the result to display from this menu.

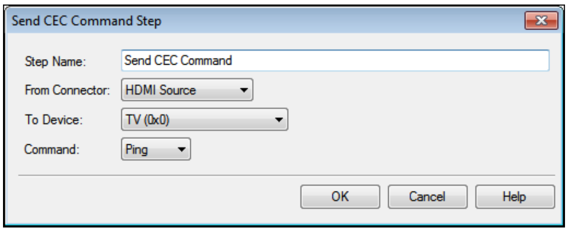
Sequencer: Adding a Send CEC Command Step

Read more about CEC on page 120.

You can add a sequence step to **Signal Path Setup** to send a CEC command from the HDMI Source or Sink connector. Either (or both) Output Configuration and Input Configuration must be currently set to HDMI to access this step.

Right click on the **Signal Path Setup** node in the Navigator and choose **Edit Prompts and Properties...** from the context menu. Click **Add Step**, and choose **Send CEC Command**.

The following dialog will open.



Step Name:

Enter an optional name for this step.

From Connector:

Select the HDMI connector (**Source, Sync, ARC Tx** or **ARC Rx**) that will send the command and receive the acknowledgement.

To Device:

Select the device to be addressed. The menu lists the 16 defined CEC logical addresses.

Command:

Select the command to send, **Ping** or **Custom**. A **Ping** is a specific CEC polling message that any addressed device should acknowledge.

When **Custom** is selected, the **Opcode** and **Operands** fields are available to add payload information to the message.

Opcode:

Enter an arbitrary opcode (operational code) here.

Operands (hex):

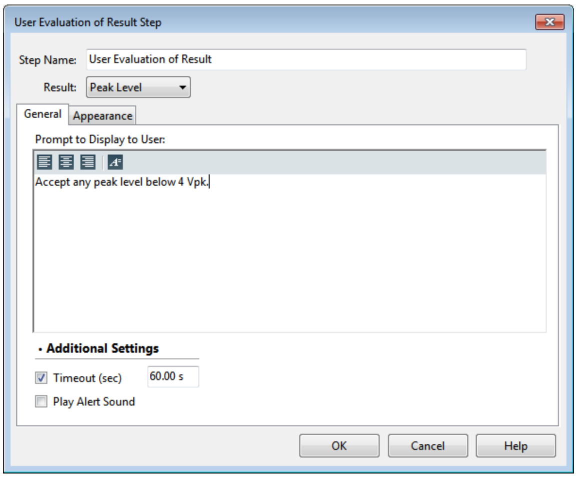
Enter arbitrary operands here, in hex.

User Evaluation of Result

Some sequences require that a user evaluate a result that is not easily evaluated by an analyzer. An example might be a stair-step pattern of automated level changes that a user can visually judge, but a machine cannot.

Right click on a measurement node in the Navigator and choose **Edit Prompts and Properties...** from the context menu. Click **Add Step**, and choose **User Evaluation of Result...**

The following dialog will open.



Step Name:

Enter an optional name for this step.

Prompt to Display to User:

Enter an optional prompt for the user here.

Timeout

Set maximum time that the sequence will wait for a response before continuing.

Play Alert Sound

Play a sound to alert the user.

Appearance tab

There is an associated **Appearance** tab in this dialog, where you can specify an image and an icon to display.

Start Analog Sine Generator

This substep is only available for the APx555, when Output Configuration is set to a non-analog output connector.

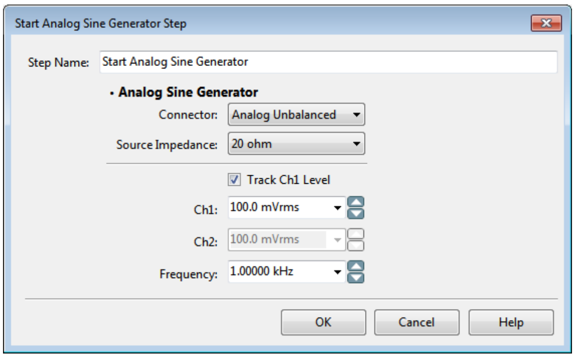
The APx555 enables the use of the analog generator as an additional signal source, when Output configuration is set to a non-analog output connection.

When this generator has been turned on in a sequence stop, you can add a sequence step to turn off the analog generator.

This step will start the analog sine generator, which, in this configuration, can be controlled independently from the DSP generator connected to a digital output.

Right click on a measurement node in the Navigator and choose **Edit Prompts and Properties...** from the context menu. Click **Add Step**, and choose **Start Analog Sine Generator...**

The following dialog will open.



Stop Analog Generator

This substep is only available for the APx555, when Output Configuration is not set to an analog output connector.

The APx555 enables the use of the analog generator as an additional signal source, when Output configuration is set to a non-analog output connection.

When this generator has been turned on in a sequence stop, you can add a sequence step to turn off the analog generator.

Right click on a measurement node in the Navigator and choose **Edit Prompts and Properties...** from the context menu. Click **Add Step**, and choose **Stop Analog Sine Generator...**

There are no additional settings for this step.

Managing sequence substeps

Enabling or Disabling substeps in the sequence

The checkbox at the left of each step row indicates whether or not that substep is enabled or disabled in the sequence; substeps are enabled (checked) by default. Uncheck the box to disable the substep.

Selecting substeps

Select a step by highlighting the substep's row using the cursor. Select multiple substeps by clicking while pressing the Shift or Ctrl keys.

Moving substeps

Substeps are processed in the sequence according to their position in the list. You can move selected substeps up or down in the list by selecting the substep(s) and dragging with the cursor, or by using the Move Selected Item(s) Up or Move Selected Item(s) Down arrows in the Toolbar, or right-click on the grid and choose Move Up or Move Down

Deleting substeps

Delete selected substeps by clicking the Delete Step(s) button, or right-click on the grid and choose Delete Step.

Editing substeps

To edit existing substep properties, select the substep and click the Edit Step button, or right-click on the substep row and choose Edit Step Properties... in the context menu.

Copying and Pasting substeps

Copy selected substeps by clicking the Copy Step(s) button, or right-click on the grid and choose Copy. Paste copied substeps by clicking the Paste button, or right-click on the grid and choose Paste. Substeps are pasted above the currently highlighted substep.

Editing substeps

To edit substep properties, select the substep and click the **Edit Step** button, or right-click on the step and choose **Edit Properties...** from the context menu.

Using Pass / Fail Limits in an automated sequence

Automated testing is usually done in conjunction with Pass / Fail limits. A limit is a value you choose as a maximum or minimum acceptable value for a measurement parameter.

You may, for example, require that no device channel have a signal-to-noise ratio less than 40 dB, or that the device's frequency response never vary more than 3 dB across a certain bandwidth. You can set limits in the signal-to-noise ratio and frequency response measurements to reflect these requirements. When the sequence is run, tests that satisfy the limits are marked as **Passed** ; those that exceed the limits are marked as **Failed** .

Displaying a message on Pass/Fail

You can display a message to the operator at the end of a sequence in response to Pass/Fail. See page 481.

Sending an Aux Control Out command on Pass/Fail

You can set the Aux Control Out state at the end of a sequence in response to Pass/Fail. See page 481.

See page 577 for more information about creating limits.

Sequences and Settling

For single value measurements, settling is applied only as a measurement or sequence is Run.

Stepped Sweep measurements are always settled, and Continuous Sweep and Signal Analyzer measurements are always unsettled.

See Chapter 92 for more information about settling.

Reports

The Report is the output document of a sequence. A report is generated when a sequence is run.

See Chapter 81 for more information about reports.

More about using the Turn Generator On substep

This substep is included in the substep order grid for some measurements. In Measurement Recorder and Signal Analyzer, it is included in the grid, but it is unavailable (grayed out) unless the measurement is configured for File Input. See Use case 2, below.

Normal operation does not require setting Turn Generator On.

You are not required to enable the Turn Generator On substep for normal operation.

All of the APx500 measurements that use a generator output will automatically turn the generator ON at the beginning of the measurement step, and OFF when the measurement is completed. However, there are situations that require the generator to be ON before the measurement begins. Here are two use cases:

Use case 1

The most common use case is to allow the operator to make adjustments to the DUT or the APx generator while observing a meter or other result, until an optimal setting for measurement is achieved.

Use case 2

Another case would be a measurement that configured for File Input that requires a DUT (such as a digital recorder) to receive a generated signal and record it to a WAV file. In the APx interface, such measurements have a Generator On button, and an Analyze button to provide this functionality. When in a sequence, the Turn Generator On step accomplishes the former, and the Measurement step accomplishes the latter.

The Turn Generator On substep is included for such cases. Move the substep to a position in the grid list before the substep that requires it (a program call to start a digital recorder, for example). Place a check in the checkbox to enable the Turn Generator On substep.

The APx500 API

Automating APx500 with the APx500 API

You can control APx500 from external programs by calling functions using the APx500 Application Programming Interface (API). The APx500 User's Manual does not cover use of the APx500 API; instead, refer to the documentation cited below.

We are currently supporting API automation for programmers using Visual Basic .NET, Visual C# .NET, and National Instruments' LabVIEW programming environment. You will find tools and information on the Audio Precision Web site at ap.com. You can download the APx API Developer Tools at http://ap.com/display/file/138. For more information about APx automation and the APx API, go to http://ap.com/products/apx/automation.

Tools from Microsoft

Visual Basic.NET

To create or edit VB.NET programs, you will need a Visual Basic editor for the .NET framework, such as Microsoft Visual Studio or Microsoft Visual Basic Express Edition. See information below regarding a free download of Visual Studio Express.

Visual C#.NET

To create or edit Visual C# .NET programs, you will need a current version Visual C# editor for the .NET framework, such as Microsoft Visual Studio or Microsoft Visual C# Express Edition. See information below regarding a free download of Visual Studio Express.

Visual Studio Express Free Download

You can download the current version of Visual Studio Express at http://www.microsoft.com/express/Windows.

Note: you must run Visual Studio or Visual Studio Express as an Administrator to open APx500 API templates and projects.

On the Audio Precision Web site at ap.com:

Refer to the file Getting_Started_with_the_APx_API.pdf.

API / Documentation Tools

- APx500 Programmer's Reference Guide
  The APx500 Programmer's Reference Guide (PRG)

documents the published API library for APx500, with support for VB.NET. The filename is “APx-500_API_PRG.chm”. This is a stand-alone Microsoft HTML Help file (H1). This file is installed with APx500 and is available via the Windows Start > All Programs > Audio Precision > APx500 n.n > API menu.

- API browser
  “APx_API_Browser.exe” is a utility that generates a complete hierarchical tree for the APx500 API. This utility is available on the Audio Precision Web site at ap.com. The program will generate documentation directly from the API files installed as components of APx500. This file is installed with APx500 and is available via the Windows Start > All Programs > Audio Precision > APx500 n.n > API menu.

API / Examples

A number of sample files have been developed to help you get started with using the APx500 API. The examples include:

- APx Information Sample
- Configure Switchers
- Control Generator Settings
- Display Meter Results
- Display Sequence Results
- Enable Signal Monitors
- Export Sample
- Manufacturing Audio Test
- Measurement Control Sample
- Read Sweep Values
- Reference Levels
- Run Measurement
- Run Sequence

API / LabVIEW Driver

National Instruments’ LabVIEW is a graphical programming language that uses icons instead of text to create applications. LabVIEW is in wide use in automated test applications. For more information about LabVIEW, visit NI’s Web site at http://www.ni.com/labview/.

Audio Precision provides a LabVIEW driver for APx500 on the Audio Precision Web site at ap.com. This driver is version specific; check the Audio Precision Web site for version information and other documentation.

The LabVIEW driver is quite comprehensive, with LabVIEW VIs for almost all of the current APx500 measurements. Visit our Web site at ap.com for more information about the LabVIEW driver.

API / Project Templates and Help

These resources are setup files that install APx API Visual Basic and Visual C# project templates. These projects include completely integrated Microsoft Help documentation for the APx API. Installers are provided for the current versions of:

- Microsoft Visual Basic Express Edition
- Microsoft Visual C# Express Edition
- Microsoft Visual Studio

Getting Started

A basic overview of the APx API is provided in

- API\Getting_Started_with_the_APx_API.pdf.

81

Reports

A report is the primary output of results from APx500 measurements. A report is created when a Sequence is run, or when a measurement is made using the Start Measurement command (found in the Navigator context menu and the Project menu).

Report types

There are two types of report:

- the APx Default Layout report, a simple, largely fixed approach. See page 497.
- the Microsoft Word report, which can be customized. See page 498.

The report type can be selected from the Type list in the Edit Properties > General, described on page 499.

For a quick output document from a single measurement view, see Printing and Exporting Results on page 501.

Your choice of report type and options will greatly affect what results are reported and how the results are presented. The Microsoft Word report, though more complicated to use than APx Default Layout report, allows you to completely control the content and presentation of the results using embedded tags.

Report Properties

Right-click on the Report node in the Navigator to open the Edit Properties dialog box.

- You can select the report Type, and add a title or notes in the Edit Properties > General tab. See page 499.
- Reports can automatically be saved at the completion of a sequence. You can select Auto-Save options in the Edit Properties > Auto-Save tab. See page 499.
- You can select and format graphics to insert in your report in the Edit Properties > Logo tab. See page 500.

What's included in a report?

A report can include information about the APx instrument used, the configuration of each signal path, and measurement results in detail (including graphic display of graphs and meters) or in a summary. Reports can include a title, a time stamp, notes, and selected images. Pass/Fail information is reported and can be used to filter the report content.

Viewing a report

If the Report node in the Navigator is checked, the report is shown at the completion of the sequence or measurement. If the node is not checked, the report is created but not shown. Click the Report node to show the report.

The APx Default Layout report is viewed in an APx500 window. Navigation, printing, save to file and exporting controls are provided in a Toolbar in the viewing window.

The Microsoft Word report is viewed in an instance of Microsoft Word invoked by APx500. Navigation, formatting, save to file, printing, etc. are controlled from within Microsoft Word.

The APx Default Layout report type

The APx Default Layout report is the default report type. You can also select it by choosing APx Default Layout as the Type in the Edit Properties > General dialog (right-click on the Report node in the Navigator). See Edit Report Properties.

An APx Default Layout report takes the form of one or more 8.5" by 11"-formatted pages created within APx500 and initially displayed in an undocked APx500 window.

Once displayed, an APx Default Layout report can be printed to any printer installed for Microsoft Windows. Reports can also be exported to a *.PDF (Adobe Acrobat), *.html (HTML, or hypertext markup language),*.rtf (rich text format), *.xls (MS Excel spreadsheet) or *.txt (text) file. Reports can be formatted with custom margins and have a custom title and company logo or other image applied.

An APx Default Layout report can include measurement setup and summary information, test results, pass / fail results (or only failed results), user prompts and meter bars or XY graphs.

APx Default Layout Report toolbar

- Edit Properties **Edit Properties** opens the Edit Properties dialog box.
  You can select the report Type, specify included information and set margins, show date, time or page numbers and add a title or notes in the **Edit Properties > General** tab. See page 499.

  Reports can automatically be saved at the completion of a sequence. You can select Auto-Save options in the **Edit Properties > Auto-Save** tab. See page 499.

  You can select and format graphics to insert in your report in the **Edit Properties > Logo** tab. See page 500.
- **Export** opens the Export browser. Export your report to PDF, HTML, RTF, XLS or CSV.
- Print... **Print** opens the Windows Print dialog.
- **Copy** copies the current page to the Windows clipboard.
- **Find** opens the Find dialog.
- **Single Page View** sets the view to one page.
- **Multiple Page View** sets the view to multiple pages.
- **Continuous Scroll** enables vertical scrolling of pages.
- **Zoom Out**
- **Zoom In**
- 100 % **Zoom menu**
- **Previous Page**
- **Next Page**
- 1/3 **Current Page Number**

Microsoft Word Report

The APx Microsoft Word report type is an option that provides much more flexibility in the selection and presentation of results, compared to the APx Default Layout report type. This is a more complicated approach, however, and requires some effort to properly format a custom layout to suit your needs.

Microsoft Word report layouts

This method applies a Contents Control tag to each report element (such as a result, result name, image, setting, etc.), and places the results and tags into an Microsoft Word layout file. A layout file acts as a template, containing the Contents Control tags and the formatting for a custom report. APx provides a default layout for Microsoft Word; you can use this file as a starting point for your custom layouts. Once you have created and saved a custom layout file by editing the default layout, you can select it as your report layout in the future.

Select Microsoft Word as the **Type** in the **Edit Properties > General** dialog. See **Edit Properties** on page 499.

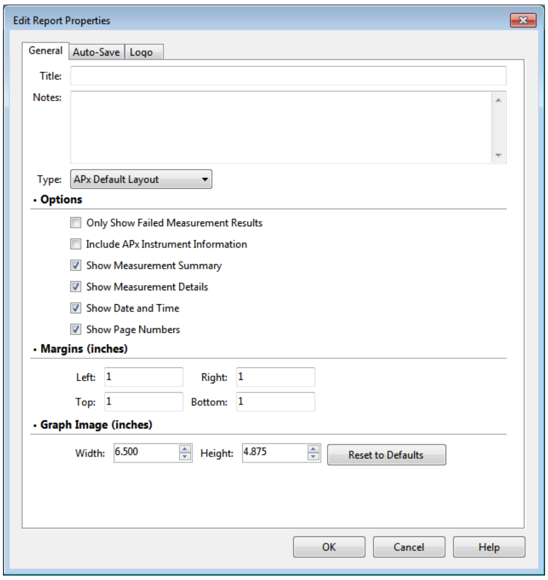
Required software

To create or edit Microsoft Word reports, you must have Microsoft Word 2013 or Microsoft Word 2010 installed on the PC. Microsoft Word 2003 and 2007 are no longer supported. Specifics follow:

Tutorials for Microsoft Word reports

There are three tutorials in the APx500 online Help to instruct you in working with the Microsoft Word report feature. We recommend that you use the tutorials from within the Help file.

Edit Properties: General tab



Choosing the Report Type

APx Default Layout

The APx Default Layout is the default report type selection. This choice creates a multipage report with graphics in the standard APx style that can be viewed in an APx window. You can export the report in a number of different formats. See APx Default Layout for detailed information about using this report type.

Microsoft Word

The Microsoft Word report type requires more effort in formatting, but can be completely customized. See Microsoft Word report for detailed information about using this report type.

APx Default Layout: Report Options

Only Show Failed Measurement Results

Only failed measurement results are reported.

Include APx Instrument Information

Adds the APx Instrument ID, its adjustment date and the APx500 software version number.

Show Measurement Summary / Show Measurement Details

These controls combine to set the degree of detail needed for your report.

APx Default Layout: Margins

Margins for the Microsoft Word style are set from within Microsoft Word.

Margins (inches)

Set page margins here. Units are inches. Defaults are 1" on each side.

APx Default Layout: Other Settings

Show Date and Time

If this box is checked, the date and time that the report was generated is printed at the bottom of each page.

Show Page Numbers

If this box is checked, the page number is printed at the bottom of each page.

Microsoft Word report layout: Report Options

You can select a default or custom layout here and can open Word to edit the layout.

Both Layout types

Title

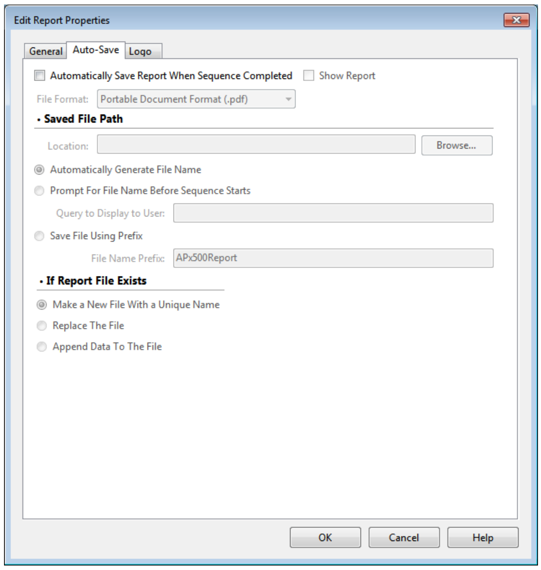
You can enter a report title with text of your choice. For the APx Default Layout, the title appears in the report title bar at the top of each page. For the Microsoft Word report type, the title appears in the "Title" tag.

Notes

You can enter notes for the report with text of your choice. The notes appear below the report title bar at the top of the first page.

You can replace the default title with text of your choice. The title appears in the report title bar at the top of each page.

Auto-Save Tab



Automatically Save Report When Sequence Completed

If this box is checked, the report is automatically saved to disk at the end of the sequence.

Show Report

If you have checked **Automatically Save Report When Sequence Completed**, you can choose to also show the report on-screen when it is generated, or to save it without showing.

File Format

For APx Default Layout reports, choose the format for automatic report save here. Options are

- PDF (Adobe Portable Document Format)
- HTML (Hypertext Markup Language)
- RTF (Rich Text Format)
- XLS (Microsoft Excel spreadsheet)
- CSV (Comma Separated Value text; data only)

For Microsoft Word reports, the format is

- .docx

Saved File Path

Location

Choose the folder for the automatically saved reports here.

Automatically Generate File Name

If this box is checked, the APx500 project name is used as the file name for automatic report saving. A time/date stamp is added to the file name. For example, a project named “My Project” and save on the afternoon of January 20, 2010 might have the name “My_Project_01_20_2010_15_14_04.docx”.

*Note: in this case, avoid the characters \ / : * ? " < > | in the Device ID. These are illegal characters in a Windows file name, and APx will replace these with underscores "_" before saving.*

Prompt for File Name Before Sequence Starts

If this box is checked, a prompt at the end of the sequence will query the user for a file name for the report (such as DUT serial number), for automatic report saving.

Query to Display to User:

Enter the text for the user filename prompt query here.

Save File Using Prefix

This option will save the report with a prefix to the file name.

File Name Prefix

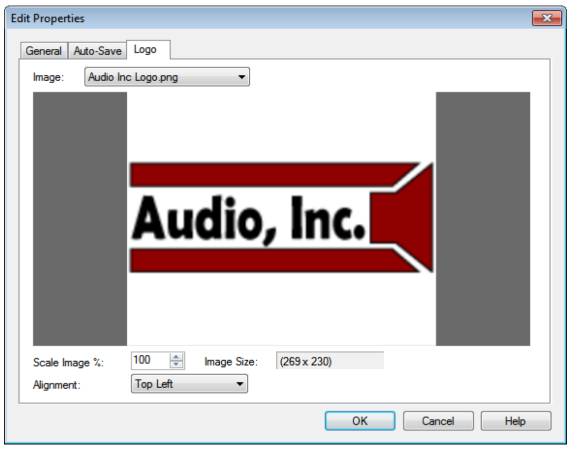
Enter the file name prefix here.

If Report File Exists

If the file already exists, you can select how Auto-Save behaves.

- **Make a new file with a unique name**
  APx will add a number (1, 2, 3, etc.) to the end of the file name, incrementing the number by 1 with each save.
- **Replace the file**
  APx will delete the previous file and save the new file with the same name.
- **Append data to the file**
  Each save operation will append the new data to the end of the file.
- **Update the data in the file** (Microsoft Word reports only)
  The file on disk will be opened, and new data will replace old data on a measurement-by-measurement basis.
  We recommend that you finalize your report layout before you use this feature. After the report is created the first time, subsequent reports use the layout in the file on disk. If you make layout changes (such as adding new tags), these changes will be ignored when the next report is generated. To save new layout changes, delete the report file on disk before creating a new report. Note that all the data in the old report file on disk will be lost.

Logo tab



You can place a logo or other image in your report. For the APx Default report style, the artwork is placed at the top of each report page, just below the report title bar. For the Microsoft Word report style, the artwork is placed at the “Logo” tag.

Logo

If an image file is loaded for this report, it is shown in the **Logo** field.

Image Size

If an image file is loaded for this report, its size in pixels is shown in the **Image Size** field.

Delete

Click the **Delete** button to remove an image file from the report.

Browse

Click the **Browse** button to open a file browser. Navigate to the image file you wish to select, and click **Open**. Supported image file types are *.bmp, *.jpg, *.gif, *.tif, *.png and *.ico.

Scale Image

You can reduce the size of the image in the report by setting **Scale Image** to less than 100%. Image aspect ratio is maintained when scaled. For the Microsoft Word report style, the artwork can be re-scaled in Word.

Alignment

You can choose whether the image is aligned to the left, the center, or the right of the report. For the Microsoft Word report style, the artwork can be re-aligned in Word.

About Image files

Image files are used by APx500 to display graphical images in Sequencer user prompts, or to display images such as company logos in reports. Image files are inserted for current use and are also attached to the project file.

Image files attached to the project but not currently in use can be removed by navigating to the dialog at **File > Manage Attached Project Items**. See page 24 for more about managing attached project items.

Supported image file formats...

- .bmp
- .jpg
- .jpeg
- .gif
- .tif
- .tiff
- .png
- .ico

Printing or Exporting Results

This feature only prints or exports results from the current view. For more powerful reporting capabilities for measurements and entire sequences, see the APx Default Layout report on page 497, or the Microsoft Word report on page 498.

To print or export results of a single measurement view, navigate to the view, run the measurement and click **Print/Export Results for this view**  from the toolbar menu at the top of the meter bar or graph display. The results are settled for the report and are compared to limits, if set.

> *The **Copy graph image to clipboard** feature does not generate a set of printable measurement results. It simply copies the graph image to the Windows clipboard. This image can then be pasted into another application. If the data on screen is unsettled, the image that is copied is also unsettled.*

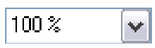
The results are shown in an undocked APx500 window. Use the window's Toolbar commands to navigate, format or to print or export the results.

Toolbar

-  **Edit Properties** opens the Edit Properties dialog box.

  You can set margins, show date, time or page numbers and add a title and notes in the **Edit Properties > General** tab.

  You can select and format a graphic to insert in your results printout in the **Edit Properties > Logo** tab.

-  **Export** opens the Export browser. Export your report to PDF, HTML, RTF, XLS or CSV.
-  **Print** opens the Windows Print dialog.
-  **Copy** copies the current page to the Windows clipboard.
-  **Find** opens the Find dialog.
-  **Single Page View** sets the view to one page.
-  **Multiple Page View** sets the view to multiple pages.
-  **Continuous Scroll** enables vertical scrolling of pages.
-  **Zoom Out**
-  **Zoom In**
-  **Zoom menu**
-  **Previous Page**

-  **Next Page**
- 

Current Page Number

Section IV: Bench Mode

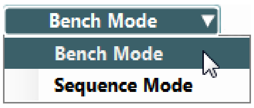
82

Bench Mode

For the APx515, Bench Mode requires a software option key. See page 166 for more information about software options.

Overview

APx500 has two operating modes: Sequence Mode and Bench Mode, each with its own workspace. APx normally launches into Sequence Mode. You can switch to Bench Mode using the selector in the upper right corner of the workspace.



Sequence Mode and Bench mode use the same underlying measurement engines, and when configured in the same way will provide identical results.

Bench Mode

Bench Mode is an alternative user interface introduced in 2014. Unlike Sequence Mode, Bench Mode does not offer defined measurements. Instead, Bench Mode provides a set of tools that can be assembled and used in many different ways. AP2700 and APWIN users will recognize similarities between the Bench Mode paradigm and earlier Audio Precision control software.

Bench Mode tools include many number bar graph meters that can each be set to one of over 12 signal parameters, real time monitors, and an extremely flexible set of sweep engines for stepped sweep, FFT, time recorder, continuous sweep and acoustic response sweeps. X and Y axes can be set to a broad range of parameters, and nested sweeps are supported.

The Bench Mode Workspace consists of three columns. The column to the left contains a collection of Signal Path Setup panels, very similar to the Signal Path Setup panels in Sequence Mode. The central column contains Generator and Analyzer settings. The right panel provides a selection of Tools displays, with a choice of meter, graph, sweep and other analysis tools.

Bench Mode Signal Path

Bench Mode supports only one signal path, and no Sequencer automation. There is no Navigator panel, since there are no Signal Paths nor sequenced measurements to navigate, select, deselect or run.

The Signal Path Setup menus are in Bench Mode, so you can configure the Bench Mode signal path inputs, outputs, filtering, switchers and so on, just as you can for a Sequence Mode signal path.

Bench Mode Generator and Analyzer

In Sequence Mode, the generator and analyzer are always associated with a particular measurement, and are often constrained to waveforms and operations suited to that measurement. In Bench Mode, the Generator and Analyzer settings are largely independent from the Bench Mode measurements you may be using.

Bench Mode Measurements and tools

Bench Mode provides a collection of tools and measurements. There are Signal Monitors, Meters, tools for initiation actions in a Bluetooth or HDMI or PDM configuration, and stepped Sweep, FFT, Recorder, Continuous Sweep and Acoustic Response measurements, each with a large suite of selectable results and displays.

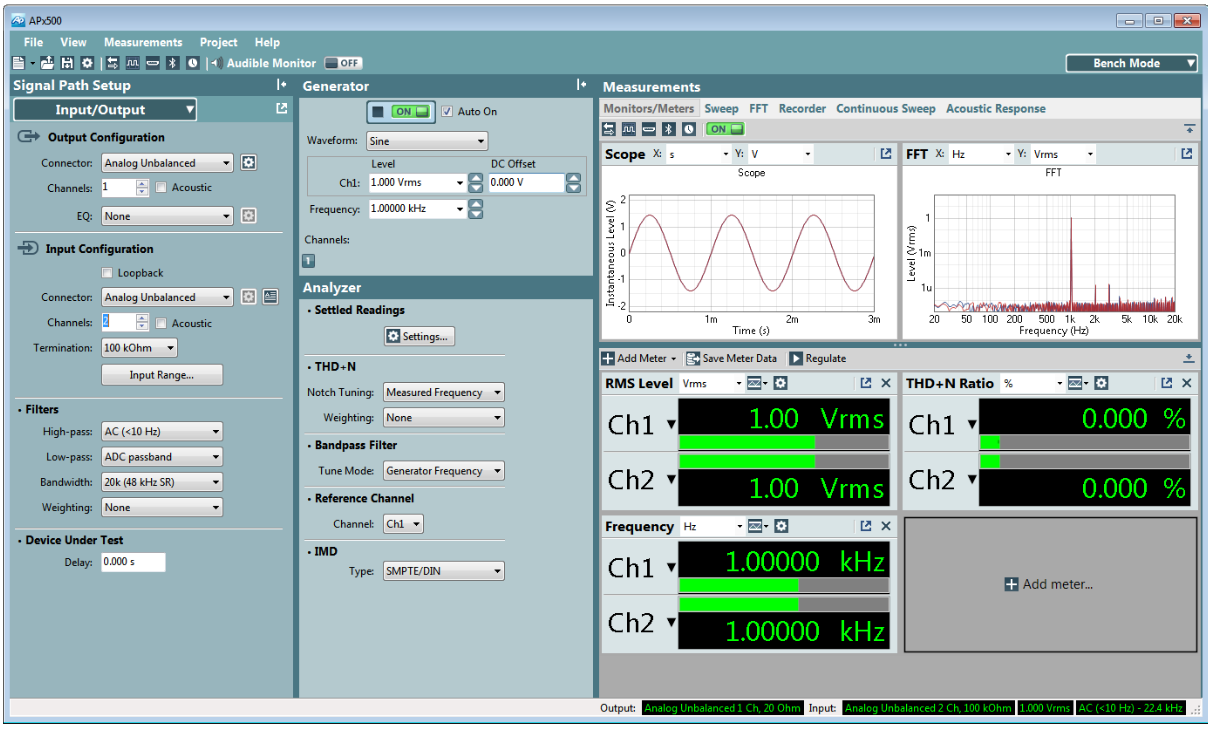
These tools and measurements are each very similar to tools and measurements in Sequence Mode, with a broader usability and application.

Hardware Support

Bench Mode and Sequence Mode support the same hardware, hardware options and accessories. Bench Mode offers slightly more capability in DCX-127 use, and in digital interface waveform analysis.

The Bench Mode Workspace

BENCH MODE WORKSPACE



SIGNAL PATH SETUP | GENERATOR AND ANALYZER | MONITORS / METERS AND MEASUREMENT TABS

Signal Path Setup

The Signal Path Setup panel and menus are located on the left of the Bench Mode workspace. The Signal Path Setup panel can be hidden.

With one exception, Signal Path Setup is identical to Signal Path Setup in Sequence Mode, with the same menus, controls and features. You can read about Signal Path Setup in Chapter 6.

Bench Mode Signal Path Setup exceptions

When Input Configuration is set to a digital input that supports interface waveform measurements, an Interface Level reading field is available on the Sequence Mode Signal Path Setup I/O menu.

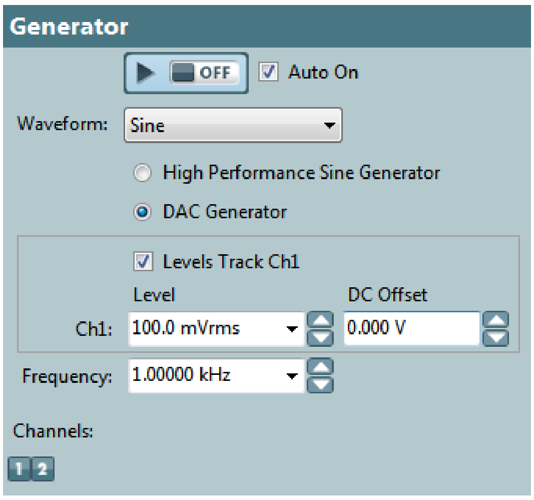
Digital Interface Level

Since there is a Digital Interface Meter available in Bench Mode, the Signal Path Setup reading field is unnecessary.

DCX measurement settings

Sequence Mode provides these settings in the context of the DCX measurements. In Bench Mode, the settings are on the Signal Path Setup DCX panel.

Generator



The Bench Mode Generator panel is located in the central set of panels in Bench Mode, above the Bench Mode analyzer panel. These panels can be hidden.

The generator is available throughout a Bench Mode session and can be adjusted at will. Three of the Bench Mode measurements (the Sweep, Continuous Sweep and Acoustic Response) temporarily take control of the generator to perform sweeps. When such sweeps are finished, the generator returns to its previous state.

On/Off

Turn the generator **ON** or **OFF** with this switch.

When **OFF**, the generator remains **OFF** unless an **Auto On** condition turns it on. When the **Auto On** condition ends, the generator reverts to **OFF**.

When **ON**, the generator remains on at its current settings unless a sweep is in progress. The sweep temporarily takes control of the generator. When the sweep is finished, the generator returns to its previous state.

Auto On

When **Auto On** is selected and the generator is off, initializing a sweep will temporarily turn the generator on. When the sweep is finished, generator is turned off.

Waveform

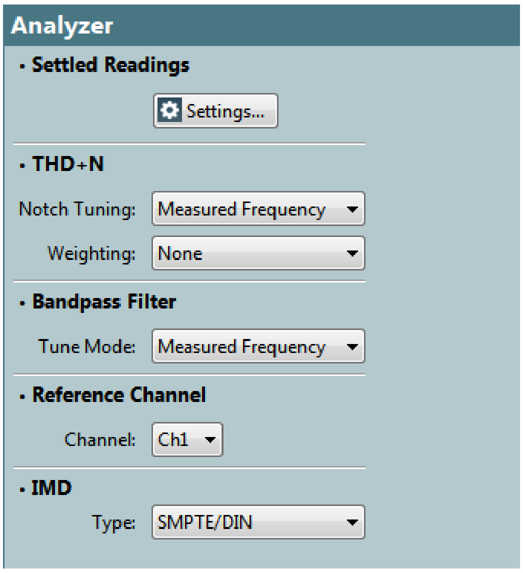
Select the generator waveform here. See Chapter 5, Signal Generation, for a full description of available generator waveforms and controls, such as Level and Frequency. The available Waveform choices change by instrument hardware and Output Configuration.

Channels

Assign the generator to output channels. By default, all active output channels are selected.

Note that **Output EQ** and other settings in **Signal Path Setup** > **Input/Output** > **Output Configuration** will affect the generator signal output. See page 52 for more information about Output EQ.

Analyzer



The Bench Mode Analyzer panel is located in the central set of panels in Bench Mode, below the Bench Mode generator panel. These panels can be hidden.

The analyzer settings are available throughout a Bench Mode session and can be adjusted at will. These settings are global within Bench Mode and affect all signal analysis.

Settled Readings

The meter readings that provide data to the Bench Mode Sweep and Recording measurements are settled. Settling is a method to identify a meter reading that represents a measured parameter accurately, excluding spurious variations caused by stimulus changes or other transient phenomena. In Bench Mode, Settling is a mechanism that provides stable, consistent results for the Sweep and Recorder tools. Read more about settling in Chapter 92.

Settings

Click the **Settings** button to open the Bench Mode **Settling** dialog, described in detail beginning on page 556.

THD+N

For meters that measure THD+N (THD+N Ratio and THD+N Level), the notch tuning and weighting filters must be specified.

See More about THD+N for a discussion of the notch filter used in THD+N measurements.

Notch Tuning

- **Generator Frequency**
  The notch is tuned to the APx generator frequency

for the current step. When the generator channels are outputting different frequencies (Split Frequency generation), the bandpass filter center is set to Frequency A.

- Measured Frequency
  The notch is tuned to the measured frequency for the current step. When the analyzer channels are receiving different frequencies, the bandpass filter for each channel is centered on the frequency in that channel.
- Fixed Frequency
  The notch is tuned to a fixed frequency selected by the user. When Fixed Frequency is selected, a Filter Freq: entry field becomes available beneath the Tune Mode control.

Frequency

When Notch Tuning is set to Fixed Frequency, enter the desired frequency here.

Weighting

A choice of weighting filters can be applied to the THD+N meter:

- A-wt.
- B-wt.
- C-wt.
- CCIR-1k
- CCIR-2k
- CCITT
- C-message

Deemphasis filters are also included, for use with pre-emphasized DUTs or stimulus signals.

- 50us de-emph.
- 75us de-emph.
- 50us de-emph. + A-wt
- 75us de-emph. + A-wt.

Notice that this is the same set of filters that can be applied to the overall input signal in Signal Path Setup > Input/Output. These filters are effectively in series, and are additive in their effect, which is rarely a desired result.

Also, consider that the Signal Path Setup > Input/Output filters are applied to the entire signal, while the filters in this area are only applied to the THD+N meter, which displays the distortion residuals.

Read more about filters in APx in Chapter 91.

Bandpass Filter

The bandpass meter differs from the RMS Level meter only in that it follows a narrow bandpass filter. The center of the filter is set by Tune Mode.

Tune Mode

- Generator Frequency
  The notch is tuned to the APx generator frequency for the current step. When the generator channels are outputting different frequencies (Split Frequency generation), the bandpass filter center is set to Frequency A.
- Measured Frequency
  The notch is tuned to the measured frequency for the current step.
  When the analyzer channels are receiving different frequencies, the bandpass filter for each channel is centered on the frequency in that channel.
- Fixed Frequency
  The notch is tuned to a fixed frequency selected by the user. When Fixed Frequency is selected, a Filter Freq: entry field becomes available beneath the Tune Mode control.

Frequency

When Tune Mode is set to Fixed Frequency, enter the desired frequency here.

Reference Channel

For Phase or Level Ratio measurements, you must specify a reference channel.

Channel

Select the reference channel here.

IMD

IMD measurements require special analysis methods.

Type

Select the IMD analysis method here. You should match the analysis mode to the generator IMD waveform used for stimulus. MOD IMD analysis is not available in Bench Mode.

- SMPTE
  This setting will analyze SMPTE and DIN IMD when used with an appropriate stimulus waveform.
- DFD
  This setting will analyze DFD IMD when used with an appropriate stimulus waveform In Bench Mode, only the d2 result is reported.
- CCIF
  This setting will analyze CCIF (ITU-R) IMD when used with an appropriate stimulus waveform. In Bench Mode, only the d2 result is reported.
- DIM
  This setting will analyze DIM IMD when used with an appropriate stimulus waveform. Audio Precision System One, System Two, Cascade and 2700 series instruments measured DIM differently than

the U1...U9 or U4+U5 modes found in the APx Sequence Mode DIM measurements, using a combination of analog filters and rms meters. In Bench Mode, DIM analysis uses Emulation mode analysis, providing DIM results compatible with results from earlier Audio Precision instruments.

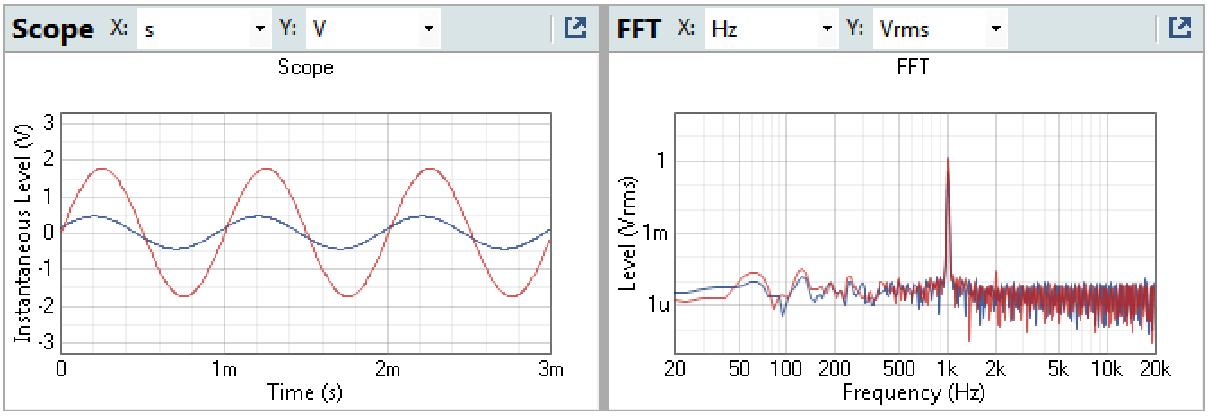
Measurements

There are six tabbed panels in the Measurements panel, each discussed in detail in the following chapters.

- Monitors/Meters, Chapter 83.
- Sweep, Chapter 84.
- FFT, Chapter 85.
- Recorder, Chapter 86.
- Continuous Sweep, Chapter 87.
- Acoustic Response, Chapter 88.

83

Monitors / Meters
(Bench Mode)



Overview

The initial Bench Mode view displays the Monitors and Meters, offering multiple real-time views of the analyzed input signals.

Scope and FFT monitors

The Scope and FFT monitors are normally docked beneath the Monitors toolbar.

These the same monitors that are used in Sequence Mode.They monitor the current analyzer inputs, displaying up to 16 channels, depending upon the input configuration. When undocked, additional monitor display controls are available. Read more about the Scope Monitor on page 31, and more about the FFT Monitor on page 31.

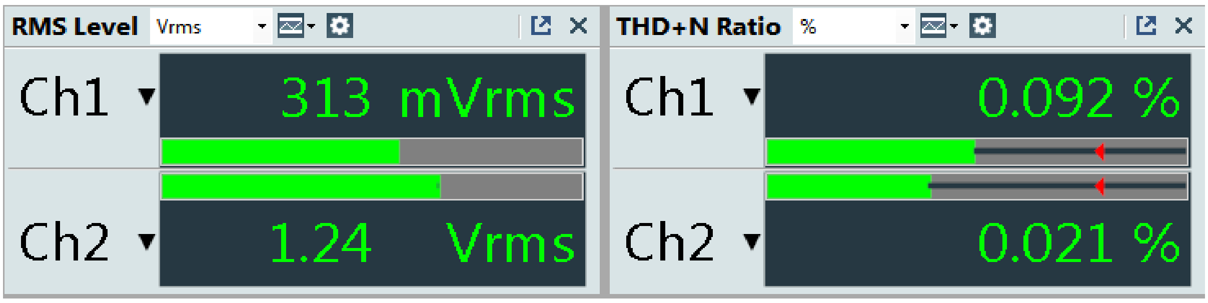
Undocking, Docking and Resizing

Click the **Undock** button to move a monitor out of the Bench Mode workspace. An undocked monitor can be resized by grabbing any of the meter edges or corners with the mouse cursor and dragging. Click the **Dock** button to return a monitor to the workspace. When undocked or opened these panels remain visible and available, even when other Bench Mode Tools, such as Sweep or Recorder are selected.

Meters

The meters are unique to Bench Mode. Each meter displays current readings as a numerical value and as a bar graph.

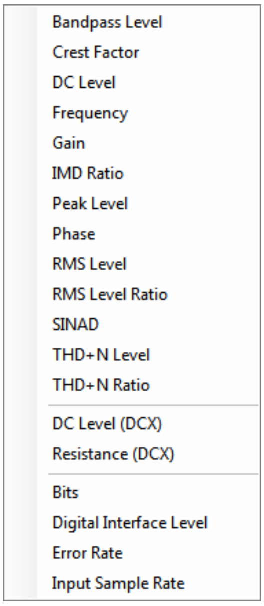
In a New Project, there are three meters shown:



- RMS Level
- Frequency
- THD+N Ratio

Adding a Meter

Click the Add Meter button and select from the menu. You can add as many meters as you want, and you can have more than one meter that displays the same parameter. Too many active meters, however, will slow system performance.



Available choices will vary with analyzer model, hardware and Signal Path Setup configuration. See detailed descriptions of Bench Mode meters on page 513.

Save Meter Data

Click Save Meter Data to open a dialog to save the current data to a csv file. Enter (or browse to) the path and filename for the csv file. Click **OK**.

If the **Append** checkbox is checked, the latest data will be added to the end of the data in the csv file. If **Append** is unchecked, previously saved data (if any) is deleted and replaced with the data in the current save operation.

The data values are unsettled, and are taken from the readings at the moment the **OK** button in the **Save Meter Data** dialog is clicked.

Regulate

The **Regulate** feature allows you to perform a regulation operation to find the parameter setting value that results in a specific reading value in a second parameter. You may, for example, want to find the generator level that produces a certain amount of distortion in the DUT. Click the **Regulate** button ▶ Regulate to open the **Regulation Settings** dialog. Read more about Bench Mode Regulation on page 514.

Arranging meters in the workspace

You can re-arrange docked meters by grabbing the meter title bar with the mouse cursor and dragging the meter to a new position.

Channels

Each meter displays 2 channels. For configurations with more than 2 input channels, you can add meters and specify additional channels.

Limits

Limits can be set for either or both channels of any meter.

Editing Limits

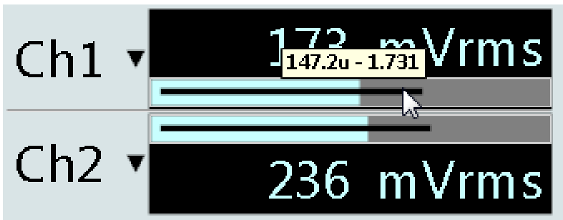
In a Bench Mode meter, select **Edit Limits** from the **Limits** menu , or right-click on the meter and select the same from the context menu.

Clear Limits

To remove all limit points, select **Clear Limits** from the **Limits** menu or from the right-click context menu.

Min/Max indicator

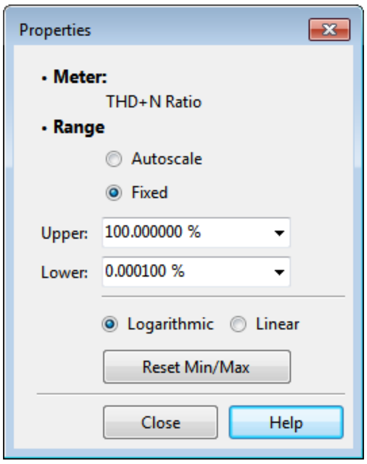
The bar graph display shows the current reading as a green bar. The excursion range of the bar graph is shown as a narrow black bar overlaying the green reading bar.



When the cursor is moved over the Min/Max indicator, a Tool Tip shows the val0ue of the minimum and maximum excursions.

The Min/Max indicator can be clear by clicking **Reset MIn/Max** in the Properties dialog.

Properties



Click the Edit Properties button to open the Properties dialog.

Meter

The meter type is displayed here

Range

Autoscale

Autoscale sets the meter range around the current signal range.

Fixed

Fixed allows you to specify a fixed graph range by **Upper** and **Lower** range values.

Upper

Set the meter range upper value here.

Lower

Set the meter range lower value here.

Logarithmic / Linear

When **Logarithmic** is selected, the meter scale follows a logarithmic rule. When **Linear** is selected, the meter scale follows a linear rule. The logarithmic scale is not available for all units.

Reset Min/Max

Click this button to reset the Meter Min/Max display.

Bench Mode Meter Descriptions

Bandpass Level

The Bandpass Level meter differs from the RMS Level meter only in that it follows a narrow bandpass filter. See page 508 for information about setting the Bench Mode Analyzer Bandpass Filter.

Crest Factor

This meter calculates and displays the Crest Factor for the selected channels. Crest factor is the peak amplitude of the waveform divided by the RMS value of the waveform, and is an indicator of the peakiness of a waveform.

DC Level

This meter reads the DC Level measured at the selected channels.

Frequency

This meter reads the frequency of the dominant tone in the signal at the selected channels.

Gain

This meter calculates and displays the gain in the selected channels. Gain is the ratio of the APx generator level to the level at the analyzer inputs.

IMD Ratio

This meter reads intermodulation distortion ratio for the selected channels. The intermodulation type (SMPTE/DIN, DFD, CCIF or DIM) as selected by the IMD Type selector in the Bench Mode Analyzer panel. See page 508 for information about setting the Bench Mode Analyzer IMD Type filter.

> *Note: The Bench Mode IMD meter is disabled when the system bandwidth is set higher than 90 kHz. The IMD stimulus signals and distortion products all fall well within the 90 kHz band, and overall system performance is improved with this meter disabled.*

Peak Level

This meter reads the AC peak level for the selected channels. For a sine wave, the peak level is one-half the peak-to-peak level, and 1.41 times the RMS level.

Phase

This meter reads the interchannel phase between the selected channels and the Reference Channel. See page 508 for information about setting the Bench Mode Analyzer Reference Channel.

RMS Level

This meter displays the RMS Level read on the selected input channels. Read more about RMS Level beginning on page 606.

RMS Level Ratio

This meter displays the ratio of the RMS Level read on the selected input channels to the RMS Level on the selected Reference Channel. See page 508 for information about setting the Bench Mode Analyzer Reference Channel.

SINAD

SINAD is a distortion ratio value, reciprocal of THD+N. Read more about SINAD beginning on page 448.

THD+N Level

THD+N Level displays the absolute RMS level of the THD+N residuals (the total signal with the fundamental removed) read on the selected input channels. Read more about THD+N beginning on page 475.

THD+N Ratio

THD+N Ratio displays the ratio of the THD+N residuals (the RMS value of the total signal with the fundamental removed) read on the selected input channels, to the RMS value of the total signal. Read more about THD+N beginning on page 475.

DC Level (DCX)

DC Level (DCX) displays the DC level measured by an optional Audio Precision DCX-127 Multifunction Module, if connected. See page 12 for information about the DCX-127.

Resistance (DCX)

DC Resistance (DCX) displays the ohmic resistance measured by an optional Audio Precision DCX-127 Multifunction Module, if connected. See page 12 for information about the DCX-127.

Bits

The Bits Meter is only available when the input is digital. The Bits Meter shows a bit-by-bit view of the embedded data in the digital signal audio word for each subframe. The two subframes correspond to Channel 1 and Channel 2.

In the bar display, the bits are numbered from the most significant bit (MSB, or bit 31 of the word) on the left, to the least significant bit (LSB, or bit 0 of the word) on the right. The Bits meter examines the signal in repeated periods of approximately 1/4 second.

When Active Bits is selected, the bar display indicates green for a bit that has changed state during the measurement period, and black for a bit that has not. The hex display shows the hexadecimal value of the difference between the changed samples.

When Data Bits is selected, the indicators display green for a bit that is at data 1 at the moment of measurement, and black for a bit that is at data 0. The hex display shows the hexadecimal value of the measured sample.

Digital Interface Level

The Digital Interface Level meter is only available when the Advanced Digital I/O module is installed, and the input is Digital Balanced or Digital Unbalanced. This meter reads the peak-to-peak level of the electrical digital interface signal (carrier).

This signal is nominally between 2 Vpp and 7 Vpp for the balanced connection when terminated in 110 ohms; it is nominally 1 Vpp for the professional unbalanced connection when terminated in 75 ohms; it is nominally 0.5 Vpp for the consumer unbalanced connection when terminated in 75 ohms.

Error Rate

This meter reads the rate of bit errors in the embedded audio of a digital signal. This meter requires a stimulus signal that can be evaluated for bit-accuracy, which includes Bit Test, Constant Value, Walking Ones and Walking Zeros.

Input Sample Rate

This meter reads the input sample rate for digital inputs.

Regulation in Bench Mode

Monitors/Meters > Regulate

Bench Mode offers a Regulate feature in the Monitors/Meters panel that allows you to perform a regulation operation to find a parameter setting value that results in a specific reading value in a second parameter. You may, for example, want to find the generator level that produces a certain amount of distortion in the DUT.

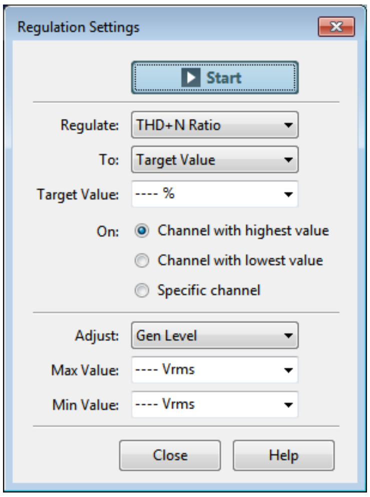
Click Regulate to open the editing dialog.

Sweep Point Regulation

A regulation feature is also available for the Bench Mode Sweep. See page 520.

Edit Regulation Settings Dialog

When accessed from Monitors/Meters, the Regulation Settings dialog includes a Start button. When accessed from the Sweep panel, regulation is automatically started at each sweep point.



The settings in this dialog are arranged rather like two sentences describing a course of action:

“REGULATE [regulated parameter] TO [value] ON [channel]. ADJUST [adjusted parameter].”

Regulate:

Select the parameter to be regulated. For example, if you are searching for the generator level that results in a specific THD+N Ratio value (the Target, Min or Max Value), select THD+N Ratio here.

To:

Select the specific result value you want to attain. Choices are

- Target Value
  Enter a value in the Target Value field.
- Maximum Value
  The effective target value will be the highest value found in the regulation process.
- Minimum Value
  The effective target value will be the lowest value found in the regulation process.

On:

Choose the channel to measured for the result value. Choices are

- Channel with the highest value
- Channel with the lowest value
- Specific Channel
  This selection makes a channel selector menu available.

Adjust:

Select the parameter to adjust. This parameter will be automatically varied between maximum and minimum values in an attempt to attain the desired value for the regulated parameter.

Max Value:

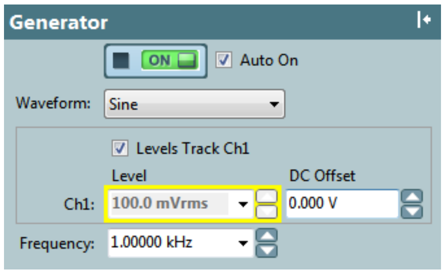
Set a maximum value for the adjusted parameter.

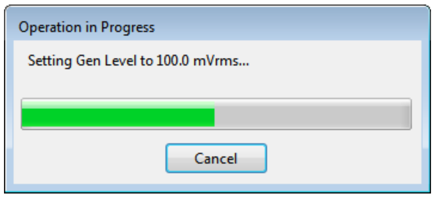
Min Value:

Set a minimum value for the adjusted parameter.

Progress view

During regulation, the adjusted parameter setting is highlighted on-screen, and a progress dialog or progress bar comments on the settings as they are varied.





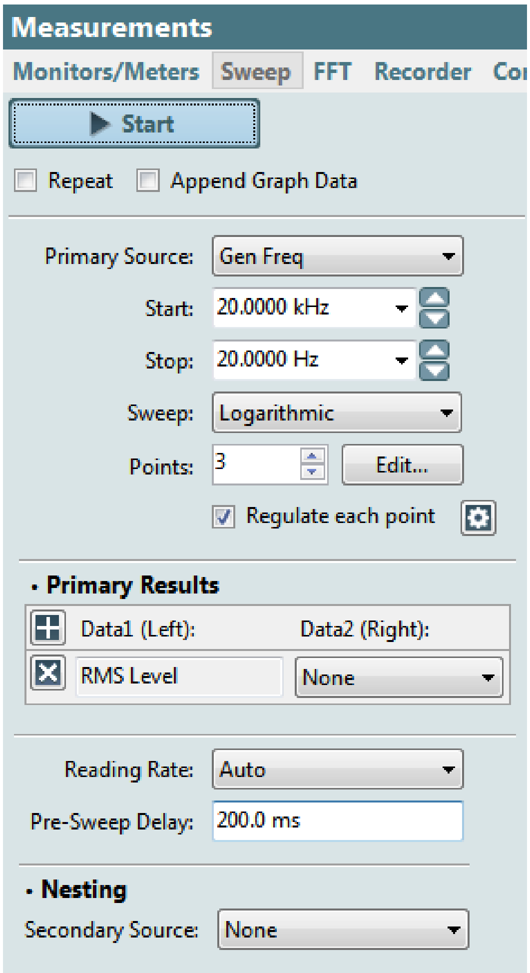
When regulation is successful in a Monitors/Meters operation, the desired value for the regulated parameter is attained, and the adjusted parameter setting retains the adjusted value. When regulation is unsuccessful, an error message is displayed.

84

Sweep (Bench Mode)

For the APx515, Bench Mode requires a software option key. See page 166 for more information about software options.

Read about the Bench Mode workspace, Generator and Analyzer panels in Chapter 82.



Overview

Bench Mode Sweep is a general-purpose stepped-sweep engine that sweeps a primary source parameter (on the X axis) in a number of steps, providing a graph of the response of one or two result parameters (on the Y axes). Typical sweeps are a frequency response sweep, where the generator frequency is swept across the audio band while the DUT's output level is graphed, or a level response sweep, where the generator level is swept across the DUT's range, while the DUT THD+N is graphed.

Sweep in Bench Mode can be considered a combination of the Stepped Frequency Sweep and Stepped Level Sweep as presented in Sequence Mode, with some Bench Mode changes and additions. You can read more about these measurements in Chapters 76 and 77.

Generator and Analyzer settings

In Bench Mode, these settings are made in the Generator and Analyzer panels. See pages 506 and 507.

Nested Sweeps

A Bench Mode Sweep can be run iteratively at different steps of an additional parameter (the secondary source) in what is referred to as a nested sweep. See Nesting, below on page 519.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. Read about the Signal Path Setup menu in Chapter 6.

Start

To make a sweep, click **Start**. The Sweep will start at the first point and continue until the results for the last point are plotted.

For File Input configuration, the Start button becomes the Analyze button.

Repeat

When the **Repeat** checkbox is checked, the sweep (or nested sweep) repeats continually. Unchecking the **Repeat** checkbox or clicking **Stop** will cancel **Repeat**.

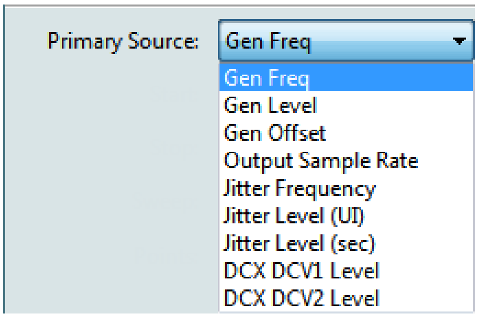
Append Graph Data

Measured data are grouped in a **Data Set**, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 573.

Primary Source

Select the primary sweep source here. This is the parameter that will be swept and plotted on the X-axis. Available choices are dependent upon instrument hardware, options, Input/Output settings and Bench Mode Generator settings. Choices may include:



Output Sample Rate is only available when Output Configuration > Connector is set to a digital output.

If Output Configuration > Connector is set to None (External), or Input Configuration > Connector is set to a File input, you must use a Sweep Table. Read about using Sweep Tables beginning on page 154.

Start

Set the sweep start value here.

Stop

Set the sweep stop value here.

Sweep

Choose

- **Logarithmic** point spacing
- **Linear** point spacing
- **Custom**
  Edit the **Sweep Points** to create a **Custom** sweep.

Some Primary Sources allow only Linear spacing.

Points

Set the number of sweep points here.

Edit

Open the **Sweep Points** dialog to edit, import or export the **Sweep Points** table.

Step Size

For **Linear** sweeps, enter the **Step Size** here.

Regulate each point

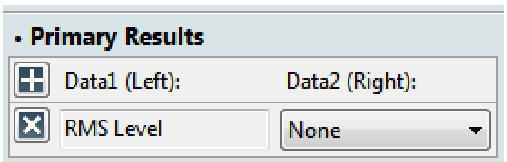
Check this box to regulate each sweep point according to the settings in the Regulation Settings dialog, discussed below on page 520. Click the **Edit...** button to open the dialog.

Primary Results

In Bench Mode Sweep, Primary results consist of a Bench Mode meter reading (the Data) on the left or right Y-axis, and a Bench Mode sweep Primary Source on the X-Axis. See Bench Mode Meter descriptions in Chapter 83.

There is an exception to this rule: when the Primary Source is set to Generator Level, four additional results are available, all providing the option of plotting the measured level (rather than the generator level) on the X-axis.

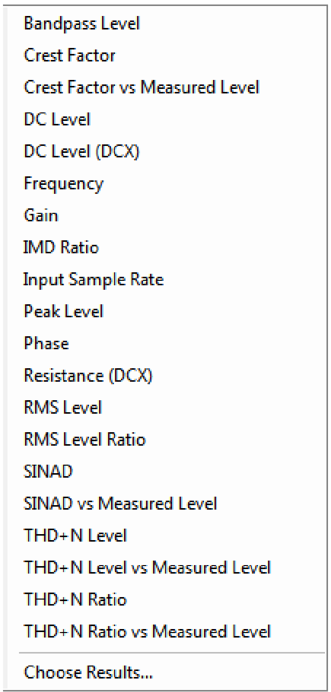
Initially, only one result (RMS Level) is set, plotted as Data1 on the left Y axis of the graph.



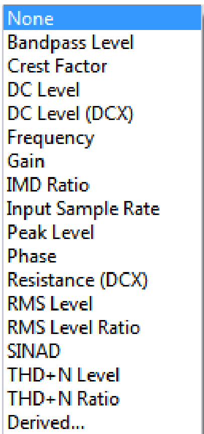
To add more primary results to the left axis, click the Add Data1 (Left) button and select one or more results.

To add a primary or derived result to the right axis, open the Data2 (Right) menu for a given existing Data1 result, and select a result.

Available choices will vary with analyzer model, hardware and Signal Path Setup configuration. Choices may include:



Data 1 Menu



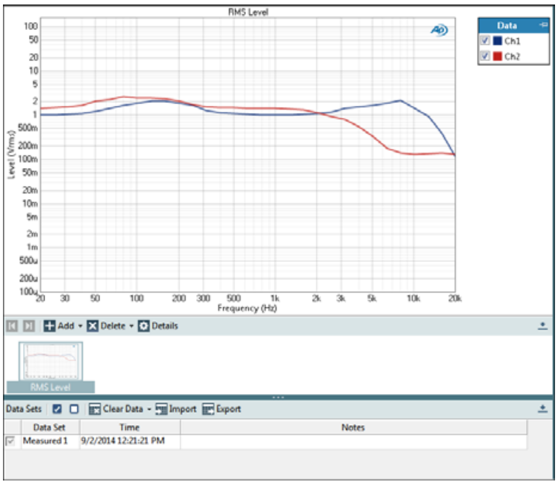
Data 2 Menu

The "vs Measured Level" results are only available when the Primary Source is Generator Level.

Remove a row by clicking the **Delete** button. Both left and right axis results are removed. Results can also be added or deleted from the Selector filmstrip.

Note that in the Data 1 menu, you can select **Choose Results...** to open the **Add Primary Results** dialog, where you can add multiple results. Also, in the **Data 2** menu, you can select **Derived...** to add **Derived Results**. See Chapter 95 for more information about Derived Results.

Typical Result Display



Settling

If Settling is enabled in the **Settling** dialog (**Analyzer > Settled Readings > Settings...**), the data for each point in the sweep is settled before plotting. Read more about Settling in Bench Mode beginning on page 507.

Reading Rate

Sweep uses a software meter to (RMS Level, for example) to read the value of the input signal for each selected result at each sweep step. **Reading Rate** sets a maximum number of times the input signal will be read per second. The default for Sweep is **Auto**.

In most cases the number of readings will be very close to the specified value. However, the rate is dependent upon PC performance and channel count, and may be lower. Read more about Reading Rate beginning on page 159.

Pre-Sweep Delay

If the generator is off when the Sweep started, you may want to delay the beginning of the sweep to allow any transients to dissipate, or to allow the DUT to warm to some nominal operating temperature. Enter a delay time between 0 seconds and 5 seconds in the **Pre-Sweep Delay** field.

Nesting

Bench Mode sweeps can be nested. See Nested Sweeps on page 161.

Secondary Source

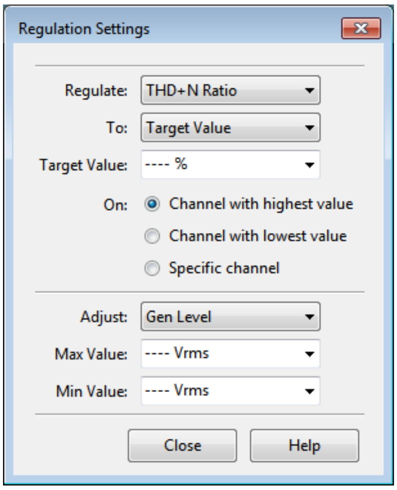
Select the secondary sweep source here. This is the outside, nesting parameter that is varied to change the conditions for the primary sweep through a number of iterations. Available choices are dependent upon instrument hardware, options, Input/Output settings and Bench Mode Generator and other settings.

Sweep Point Regulation

A regulation feature is available for the Bench Mode Sweep. When the Sweep panel **Regulate each point** checkbox is checked, regulation is performed on every sweep point before it is displayed. Click the **Edit...** button to open the editing dialog.

Edit Regulation Settings Dialog

When accessed from the Sweep panel, regulation is automatically started at each sweep point.



The settings in this dialog are arranged rather like two sentences describing a course of action:

“REGULATE [regulated parameter] TO [value] ON [channel]. ADJUST [adjusted parameter].”

Regulate:

Select the parameter to be regulated. For example, if you are searching for the generator level that results in a specific THD+N Ratio value (the Target, Min or Max Value), select THD+N Ratio here.

To:

Select the specific result value you want to attain. Choices are

- Target Value
  Enter a value in the Target Value field.
- Maximum Value
  The effective target value will be the highest value found in the regulation process.
- Minimum Value
  The effective target value will be the lowest value found in the regulation process.

On:

Choose the channel to measured for the result value. Choices are

- Channel with the highest value
- Channel with the lowest value
- Specific Channel
  This selection makes a channel selector menu available.

Adjust:

Select the parameter to adjust. This parameter will be automatically varied between maximum and minimum values in an attempt to attain the desired value for the regulated parameter.

Max Value:

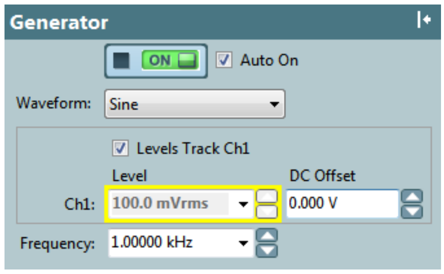
Set a maximum value for the adjusted parameter.

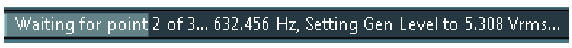
Min Value:

Set a minimum value for the adjusted parameter.

Progress view

During regulation, the adjusted parameter setting is highlighted on-screen, and a progress dialog or progress bar comments on the settings as they are varied.





When regulation is successful, the point is plotted and the sweep moves to the next point.

When regulation is unsuccessful, an error message is displayed.

85

FFT
(Bench Mode)

For the APx515, Bench Mode requires a software option key. See page 166 for more information about software options.

Read about the Bench Mode workspace, Generator and Analyzer panels in Chapter 82.

Overview

Bench Mode FFT is a general purpose diagnostic tool that uses a high-resolution FFT technique to provide several different display results. Read More About FFTs beginning on page 442.

FFT in Bench Mode is essentially the same as the Signal Analyzer measurement that is presented in Sequence Mode, with some Bench Mode additions. Read more about the Signal Analyzer in Chapter 73.

Generator and Analyzer settings

In Bench Mode, these settings are made in the Generator and Analyzer panels. See pages 506 and 507.

Nested Sweeps

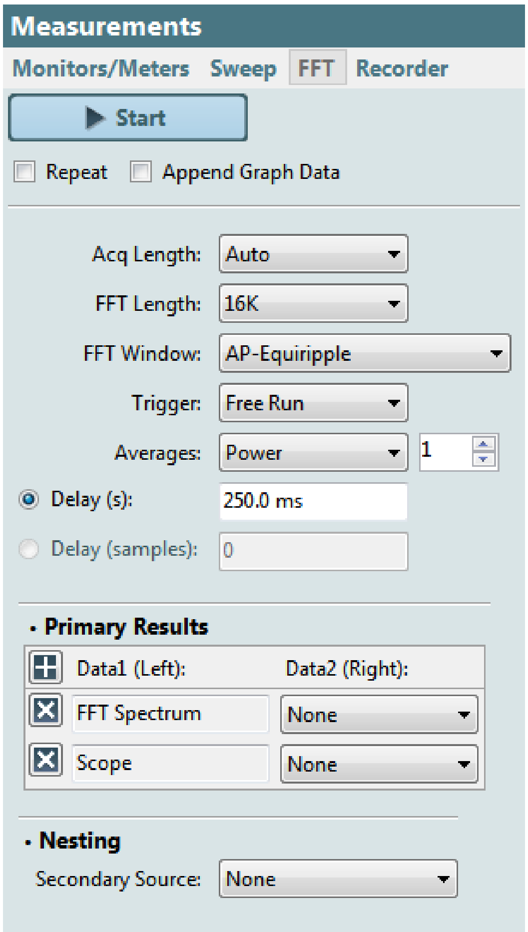
A Bench Mode FFT can be run iteravely at different steps of a fourth parameter (the secondary source) in what is referred to as a nested sweep. See Nesting, below on page 524.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. Read about the Signal Path Setup menu in Chapter 6.

Start

To run FFT, click **Start**. The signal selected in the Generator (if any) will be applied to the DUT, and an FFT of the DUT's output will be graphed here.



Repeat

When the **Repeat** checkbox is checked, the FFT acquisition (or nested acquisitions) repeats continually. Unchecking the **Repeat** checkbox or clicking **Stop** will cancel **Repeat**.

Append Graph Data

Measured data are grouped in a Data Set, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the Append Graph Data box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 573.

Acq. Length

Set the acquisition length here, in seconds or in samples. Auto, the default, sets the acquisition length to match the FFT length.

FFT Length

Set the FFT record length here. Options from 256 points to 1.2M points are available, depending upon the number of channels in use; default is 16K points.

FFT Window

FFT acquisitions must either be synchronous or have one of a number of amplitude windows applied to provide useful data for interpretation. Each window function brings advantages and disadvantages. The default selection, AP-Equiripple, is a proprietary Audio Precision FFT window that is an excellent choice for most FFT measurements.

Read about FFT windows beginning on page 443.

Trigger

- Free Run
  When trigger is set to Free Run, the acquisition begins when the Start button is clicked.
- +0 Crossing
  When trigger is set to +0 Crossing, the acquisition begins at the waveform's first positive-going zero crossing, for the input channel selected.
- Generator
  Generator is only available if the Generator is configured to play a file (Waveform: Browse for file...). When Trigger is set to Generator, the acquisition begins at the file's first sample. A new acquisition is triggered each time the generator waveform loops back to its first sample.
- External
  External is only available for analyzers equipped with the Advanced Master Clock. The acquisition begins when a trigger pulse is received at the rear-panel Trigger In BNC connector. Set the Trigger Input in Signal Path Setup > Triggers to the desired voltage threshold and pulse edge.

Averages

For non-periodic waveforms such as noise, averaging multiple acquisitions can provide a more useful view. By setting Averages to more than 1, you can specify how many acquisitions to average. Default is 1; maximum is 1000.

- Power
  The scope result does not use power averaging. Power averaging affects only the frequency domain results.
- Synchronous
  When Trigger is set to Generator, Synchronous Averaging is an option. Synchronous averaging affects both the time domain and the frequency domain results.

Read More About Averaging beginning on page 444.

Max Hold and Min Hold are not averaging functions, but use the multiple acquisitions available to the Averages feature to find maximum or minimum values for each FFT bin.

- Max Hold
  Max Hold displays the highest value for each bin found in all the acquisitions set in Averages. As the acquisitions progress, the value in each bin can only go up.
  The scope result does not use Max Hold. This selection only affects the frequency domain results.
- Min Hold
  Min Hold displays the lowest value for each bin found in all the acquisitions set in Averages. As the acquisitions progress, the value in each bin can only go down.
  The scope result does not use Min Hold. This selection only affects the frequency domain results.

Delay (s) / Delay Samples

The Delay (s) / Delay Samples setting allows you to delay the acquisition by a specific period of time (or, for a Generator Trigger, a specific number of samples) after the FFT Start. If the Generator is set to Auto On, this effectively lets the generator or the generator waveform to run for a time before acquisition.

This is sometimes necessary to avoid analysis of spurious signals created when using a coded generator file with a DUT that requires some time to match the coded format, or to allow the filter to settle when AC coupled.

For Delay (s), enter the delay in seconds. Minimum time is 0.0 s; maximum is 60.0 s; the default is 250 ms.

If **Trigger** is set to **Generator**, you can choose to enter the delay in Samples rather than seconds. Minimum sample count is 0; maximum is 12,960,000.

Signal

The APx515 does not support the PDM I/O module.

The **Signal** field is only available when **Input Connector** in **Signal Path Setup** is set to **PDM**.

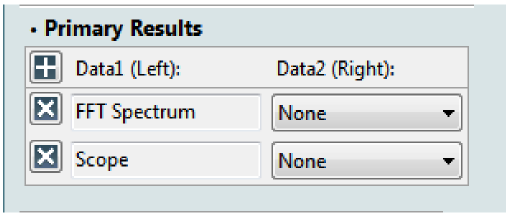
- **Decimated Audio**
  The decimated PDM audio (at the baseband sample rate and bandwidth) is the input signal.
- **PDM Bitstream**
  The PDM bitstream (at the PDM bit clock rate and bandwidth) is the input signal. This is useful to view the out-of-band noise and other components of the PDM bitstream.

Note: When the PDM input decimation is set to x1, (see Settings for PDM Input on page 138) the PDM signal is not decimated. In that case, either of the Signal selections in the Signal Analyzer will present the undecimated PDM bitstream for the analyzer acquisition.

Primary Results

In Bench Mode FFT, Primary results consist one of four FFT results on the left or right Y-axis. The X-axis is scaled in the result domain (either frequency domain or time domain).

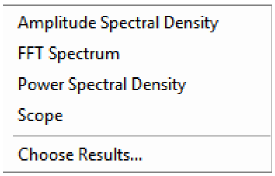
Initially, only two results are set, plotted as Data1 on the left Y axis of the graph.



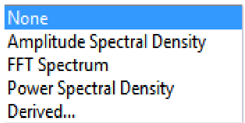
To add more primary results to the left axis, click the Add Data1 (Left) button and select one or more results.

To add a primary or derived result to the right axis, open the Data2 (Right) menu for a given existing Data1 result, and select a result.

Choices include:



Data 1 Menu

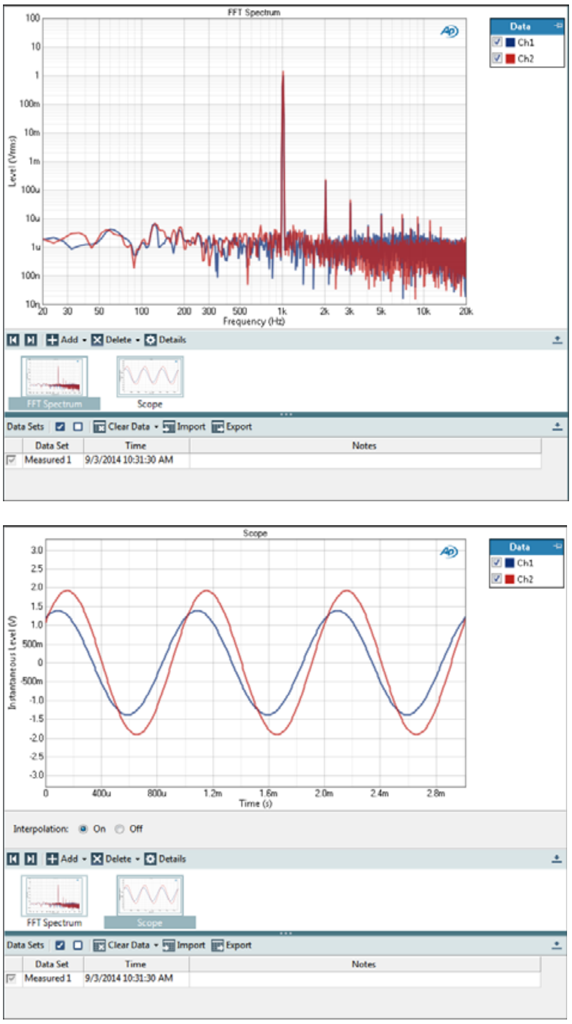


Data 2 Menu

Remove a row by clicking the **Delete** button. Both left and right axis results are removed. Results can also be added or deleted from the Selector filmstrip.

Note that in the Data 1 menu, you can select **Choose Results...** to open the **Add Primary Results** dialog, where you can add multiple results. Also, in the **Data 2** menu, you can select **Derived...** to add **Derived Results**. See Chapter 95 for more information about Derived Results.

Typical Result Displays



Spectrum and scope result displays.

Settling

FFT results are always unsettled.

Nesting

Bench Mode FFTs can be nested. See Nested Sweeps on page 161.

Secondary Source

Select the secondary sweep source here. This is the outside, nesting parameter that is varied to change the conditions for the recording through a number of iterations. Available choices are dependent upon instrument hardware, options, Input/Output settings and Bench Mode Generator and other settings.

86

Recorder
(Bench Mode)

For the APx515, Bench Mode requires a software option key. See page 166 for more information about software options.

Read about the Bench Mode workspace, Generator and Analyzer panels in Chapter 82.

Overview

Bench Mode Recorder is a general-purpose diagnostic tool that provides a record of a number of measurements versus elapsed time. The record can be very long (up to one week), providing a means of monitoring the output of a device under test over an extended period of time. The Recorder does not require a specific test signal. It can be used with any audio signal within the input range of the analyzer, or with no signal.

The Recorder also provides a means of recording the audio acquisition to a file. See **Save to File**, below on page 527.

Recorder in Bench Mode is similar to Measurement Recorder presented in Sequence Mode, with some Bench Mode changes and additions. You can read about the Measurement Recorder in Chapter 57.

Generator and Analyzer settings

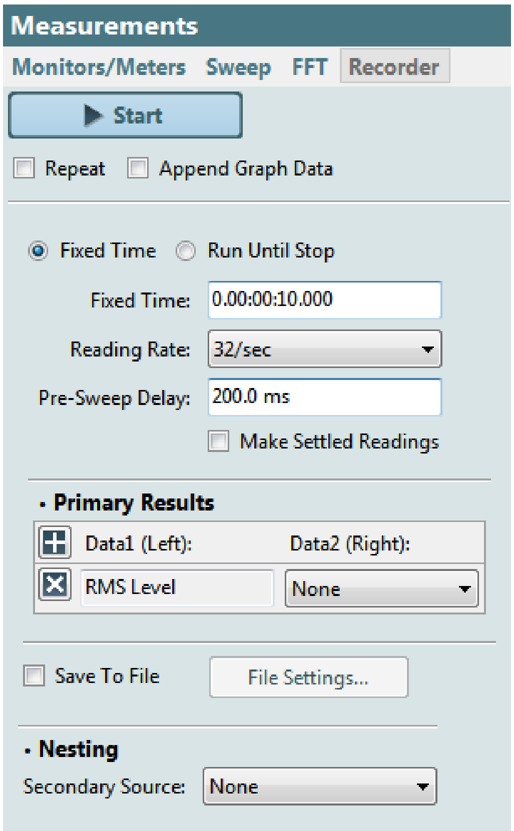
In Bench Mode, these settings are made in the Generator and Analyzer panels. See pages 506 and 507.

Nested Sweeps

A Bench Mode Sweep can be run iteravely at different steps of an additional parameter (the secondary source) in what is referred to as a nested sweep. See Nesting, below on page 527.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. Read about the Signal Path Setup menu in Chapter 6.



Start

To make a measurement recording, click **Start**. The measurement recording will begin and the graph will display as the record is acquired. The record will continue until the end time set in **Fixed Time**, or until **Stop** is pushed.

Note: For digital inputs, sample rate must be stable before the Recorder begins. When *playing a digital external source (such as a DVD), we advise that you start the playback device first, then click the Recorder Start button.*

Repeat

When the **Repeat** checkbox is checked, the recording (or nested sweep) repeats continually. Unchecking the **Repeat** checkbox or clicking **Stop** will cancel **Repeat**.

Note: if Append Graph Data is checked while Repeat is set, a great deal of data will be quickly amassed and the computer memory or storage may be overwhelmed.

Append Graph Data

Measured data are grouped in a **Data Set**, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the **Append Graph Data** box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 573.

Primary Source

For the Bench Mode Recorder, the Primary Source plotted on the graph X-axis is always **Time**, in seconds (s).

Fixed Time

Fixed Time sets the length of the measurement record. Minimum time is 0.1 s, maximum time is 7 days, or 12 million points at the selected reading rate, whichever is less.

Enter the desired time in the **Fixed Time** field following this pattern:

d:hh:mm:ss.s

where d = days, hh = hours, mm = minutes, ss = seconds. Days and fractional seconds are optional.

Run Until Stop

This selection will record until 12 million points are acquired, unless **Stop** is clicked.

Reading Rate

Recorder uses a software meter to (RMS Level, for example) to read the value of the input signal for each selected result at each sweep step. **Reading Rate** sets a maximum number of times the input signal will be read per second. The default for Recorder is **32/sec.**

In most cases the number of readings will be very close to the specified value. However, the rate is dependent upon PC performance and channel count, and may be lower. Read more about Reading Rate beginning on page 159.

Pre-Sweep Delay

If the generator is off when the Recorder is started, you may want to delay the beginning of the recording to allow any transients to dissipate, or to allow the DUT to warm to some nominal operating temperature. Enter a delay time between 0 seconds and 5 seconds in the Pre-Sweep Delay field.

Make Settled Readings

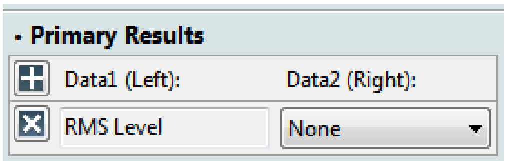
Settling parameters are set in the **Settling** dialog available in the Bench Mode **Analyzer** panel. You can engage or disengage settling for the Recorder here. When the **Make Settled Readings** checkbox is checked, every Recorder reading is settled, which can slow the result display, depending upon Settling settings and the nature of the incoming signal. Read about Settling in Bench Mode beginning on page 507.

When the **Make Settled Readings** checkbox is unchecked, the raw, unsettled data is plotted.

Primary Results

In Bench Mode Recorder, Primary results consist of a Bench Mode meter reading (the Data) on the left or right Y-axis, and Time on the X-Axis. See Bench Mode Meter descriptions in Chapter 83.

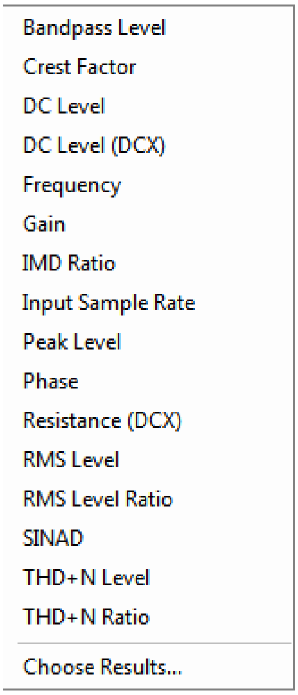
Initially, only one result (RMS Level) is set, plotted as Data1 on the left Y axis of the graph.

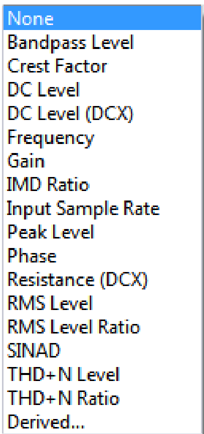


To add more primary results to the left axis, click the Add Data1 (Left) button and select one or more results.

To add a primary or derived result to the right axis, open the Data2 (Right) menu for a given existing Data1 result, and select a result.

Available choices will vary with analyzer model, hardware and Signal Path Setup configuration. Choices may include:





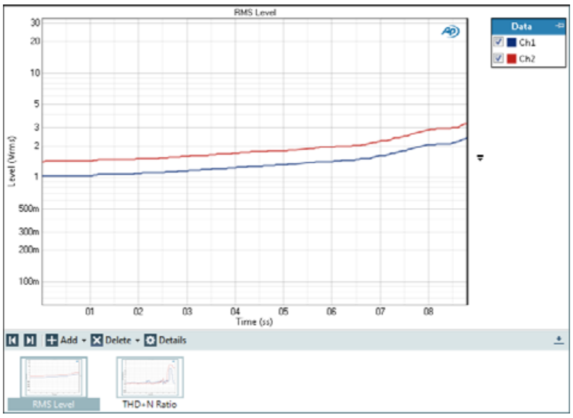
Data 1 Menu

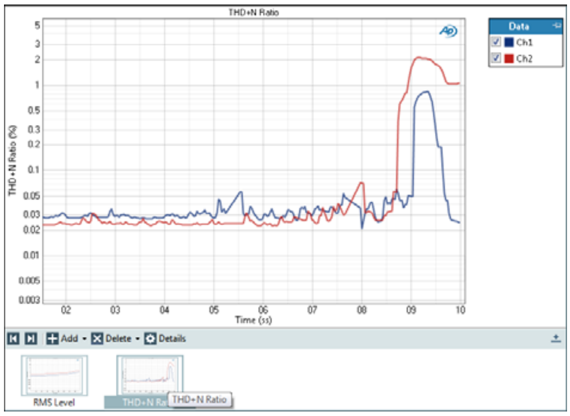
Data 2 Menu

Remove a row by clicking the **Delete** button. Both left and right axis results are removed. Results can also be added or deleted from the Selector filmstrip.

Note that in the Data 1 menu, you can select **Choose Results...** to open the **Add Primary Results** dialog, where you can add multiple results. Also, in the **Data 2** menu, you can select **Derived...** to add **Derived Results**. See Chapter 95 for more information about Derived Results.

Typical Result Displays





These results show a 10 second recording of Level and THD+N results, while the DUT gain is increased.

Save To File

You can record the audio signal to a computer file as the signal is being acquired. Click **Save to File** before you run the measurement recorder. See **Recording Audio to a File** on page 347 for detailed information.

File Settings

When saving a audio signal to a file, click **File Settings** to specify the file name and location.

Nesting

Bench Mode Recorder **Fixed Time** records can be nested. **Run Until Stop** records cannot be nested. See Nested Sweeps on page 161.

Secondary Source

Select the secondary sweep source here. This is the outside, nesting parameter that is varied to change the conditions for the recording through a number of iterations. Available choices are dependent upon instrument hardware, options, Input/Output settings and Bench Mode Generator and other settings.

87

Continuous Sweep
(Bench Mode)

For the APx515, Bench Mode requires a software option key. See page 166 for more information about software options.

Read about the Bench Mode workspace, Generator and Analyzer panels in Chapter 82.

Overview

Bench Mode Continuous Sweep is a brief log-swept sine wave (a Farina log chirp) that moves continuously across a specified range of frequencies. The DUT output is acquired by the analyzer and is mathematically processed to provide a number of results. For more information about continuous sweep measurements, go to page 220.

Continuous sweep based measurements are not supported in External Source or File Input configurations.

Continuous Sweep in Bench Mode is essentially the same measurement that is presented in Sequence Mode, with some Bench Mode additions. You can read about the Continuous Sweep measurement in Chapter 28.

Generator and Analyzer settings

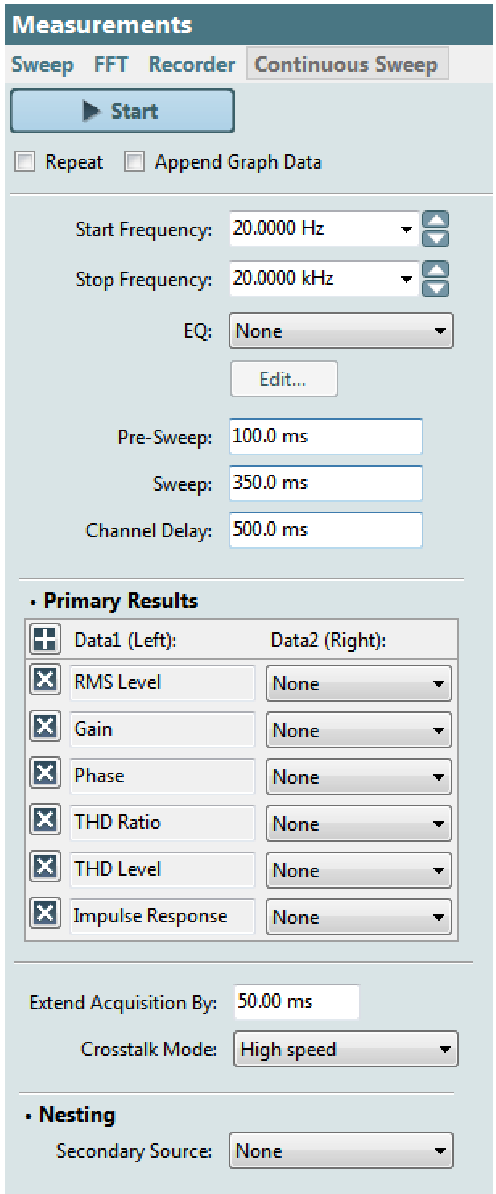
In Bench Mode, these settings are made in the Generator and Analyzer panels. See pages 506 and 507.

Nested Sweeps

A Bench Mode Continuous Sweep can be run iteravely at different steps of an additional parameter (the secondary source) in what is referred to as a nested sweep. See Nesting, below on page 531.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and out-



puts, references and other settings. Read about the Signal Path Setup menu in Chapter 6.

Start

To make a Continuous Sweep, click Start. The generator will output a continuous sweep to the DUT on the selected generator channels at the level set in the Generator panel. In a short period of time, the signal will be acquired and processed, and each of the small graphs in the Selector will be populated with results. Simply click the result view you want.

Repeat

When the Repeat checkbox is checked, the sweep (or nested sweep) repeats continually. Unchecking the Repeat checkbox or clicking Stop will cancel Repeat.

Note: if Append Graph Data is checked while Repeat is set, a great deal of data will be quickly amassed and the computer memory or storage may be overwhelmed.

Append Graph Data

Measured data are grouped in a Data Set, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the Append Graph Data box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 573.

Primary Source

The primary source for a continuous sweep measurement is the log-swept sine signal. The range between the Start and Stop frequencies provides the X-axis for most of the result graphs.

Start Frequency and Stop Frequency

These controls set the frequency range of the sweep.

EQ

You can equalize the generator signal to make a pre-emphasized stimulus. See Generator Equalization (page 170) for a general discussion of this feature.

Choosing generator equalization

Select None, Relative or Absolute from the EQ drop-down menu.

- None applies no equalization to the generator.
- Relative applies the gain or loss specified in the EQ table to the current generator level for each channel.
- Absolute sets the generator level for all channels to the level specified in the EQ table. Note: when Absolute is selected, current channel level settings are lost.

Edit

If you have chosen to equalize the generator signal, you can edit the EQ table here. See Edit EQ Table (page 170).

Pre-Sweep and Sweep duration fields

A continuous sweep stimulus consists of a sine wave swept logarithmically across the sweep range in the period of time set in these fields.

To enable the DUT to stabilize before the beginning of the defined sweep measurement, there is variable Pre-Sweep time. The actual sweep stimulus signal begins below the defined Start Frequency and is swept upward, reaching the Start Frequency at the end of the Pre-Sweep time. The sum of the times in Pre-Sweep and Sweep is the total sweep length.

- The default for Pre-Sweep is 100.0 ms. Minimum is 0 s; maximum is 1.0 s.
- The default for Sweep is 350.0 ms. Minimum is 50.0 ms; maximum is 5 s.

Longer Pre-Sweep settings allow more time for DUT stabilization, but increase measurement time. Longer Sweep settings provide greater resolution and signal-to-noise ratio, but increase measurement time.

Channel Delay

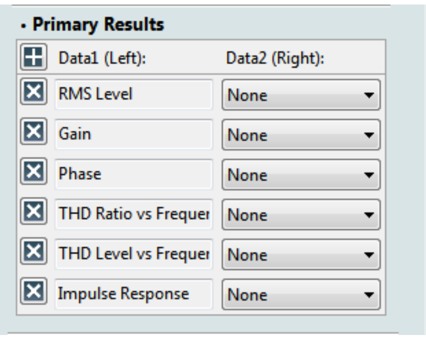
The Channel Delay control allows you to specify a delay before the Start moments for each channel when in one of the High Accuracy Crosstalk modes.

If the Channel Delay is zero or very short, the slew rate limiting of a DUT or the time constant in AC Coupling can cause a transient when energy from the sweep in a previous channel appears in the analysis of the current channel. The default Channel Delay is 500 ms. If DC coupled and the DUT does not have appreciable slew rate limiting, Channel Delay can be set to a very low value, or 0.

Primary Results

In Bench Mode Continuous Sweep, Primary results include 16 sets of data derived from the sweep.

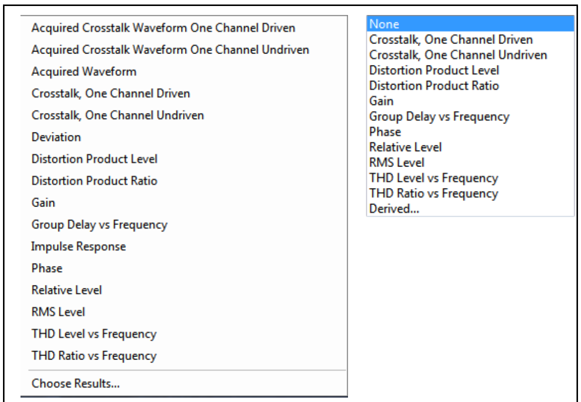
Initially, only six results are set, plotted as Data1 on the left Y axis of the graph.



To add more primary results to the left axis, click the Add Data1 (Left) button and select one or more results.

To add a primary or derived result to the right axis, open the Data2 (Right) menu for a given existing Data1 result, and select a result.

Available choices will vary with analyzer model, hardware and Signal Path Setup configuration. Choices may include:

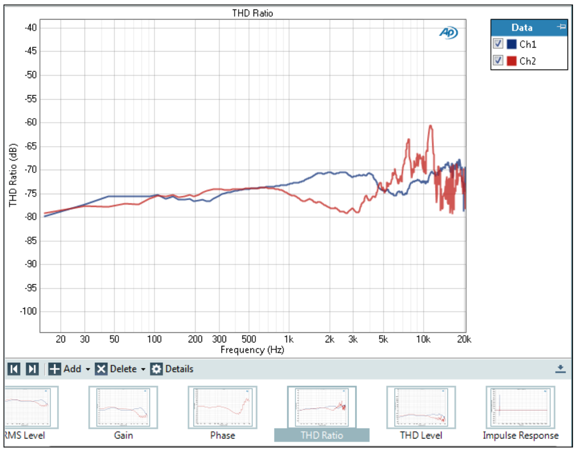


Data 1 Menu Data 2 Menu

Remove a row by clicking the **Delete** button. Both left and right axis results are removed. Results can also be added or deleted from the Selector filmstrip.

Note that in the Data 1 menu, you can select **Choose Results...** to open the **Add Primary Results** dialog, where you can add multiple results. Also, in the **Data 2** menu, you can select **Derived...** to add **Derived Results**. See Chapter 95 for more information about Derived Results.

Typical Result Display



Continuous Sweep result displays are shown in detail beginning on page 213.

Settling

Continuous sweep results are always unsettled.

Extend Acquisition By

By default, a continuous sweep measurement makes an acquisition slightly longer than the stimulus, to include possible time-delayed artifacts created in the DUT. By default, the acquisition is extended 50 ms longer than the stimulus.

In some cases, you may want to extend the acquisition further. Enter a new value in the **Extend Acquisition By:** field. Minimum extension is 0 s; maximum is 3 s.

Crosstalk Mode

Choose the mode of crosstalk measurement (or **None**). This selection changes the sweep alignment in time across channels, and may affect other continuous sweep results. For more information, see Continuous Sweep Crosstalk modes on page 222.

Choices are

- None
- **High speed** (default)
- High accuracy, one channel driven
- High accuracy, one channel undriven

Nesting

Bench Mode Continuous Sweeps can be nested. See Nested Sweeps on page 161.

Secondary Source

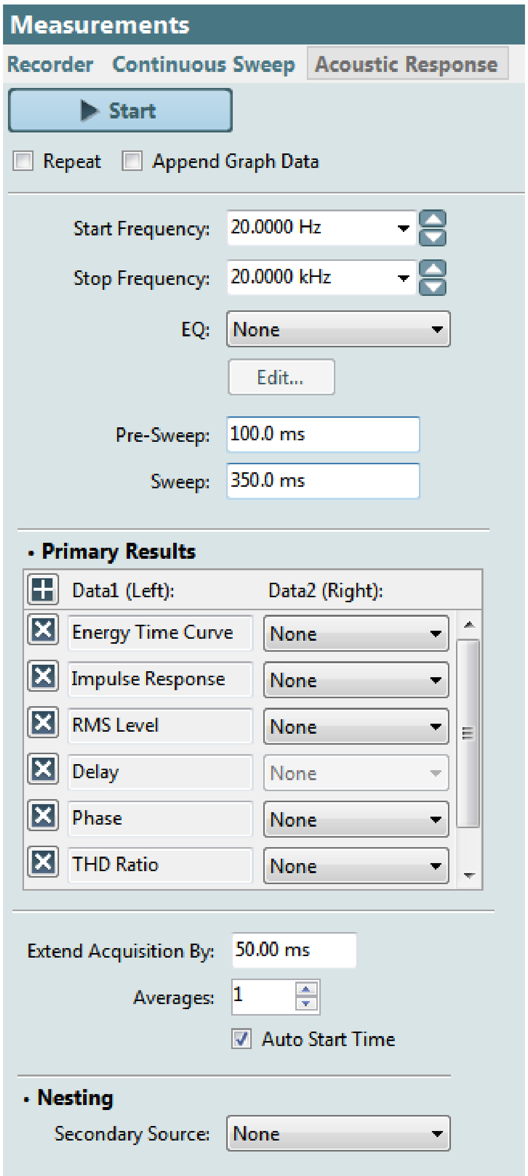
Select the secondary sweep source here. This is the outside, nesting parameter that is varied to change the conditions for the primary sweep through a number of iterations. Available choices are dependent upon instrument hardware, options, Input/Output settings and Bench Mode Generator and other settings.

88

Acoustic Response
(Bench Mode)

For the APx515, Bench Mode requires a software option key. See page 166 for more information about software options.

Read about the Bench Mode workspace, Generator and Analyzer panels in Chapter 82.



Overview

Bench Mode Acoustic Response uses a continuous sweep, with the addition of a time window feature to enable quasi-anechoic acoustic measurements.

The Acoustic Response measurements are designed to facilitate loudspeaker testing using an APx500 Series analyzer with a measurement microphone and an audio power amplifier.

Typically, the APx500 generator output is connected to the power amplifier, which in turn is connected to the loudspeaker under test. A measurement microphone is located in the acoustic field of the loudspeaker, and the output of the microphone preamplifier is connected to an APx500 analyzer input. A specially configured continuous response measurement (called the Acoustic Response measurement) is made.

For more information about continuous sweep measurements, go to page 220.

Continuous sweep based measurements are not supported in External Source or File Input configurations.

Acoustic Response in Bench Mode is essentially the same measurement that is presented in Sequence Mode, with some Bench Mode additions Read more about Acoustic Response and in particular, tips for setting up you measurements, in Chapter 21.

Generator and Analyzer settings

In Bench Mode, these settings are made in the Generator and Analyzer panels. See pages 506 and 507.

Nested Sweeps

A Bench Mode Acoustic Response sweep can be run iteravely at different steps of an additional parameter (the secondary source) in what is referred to as a nested sweep. See Nesting, below on page 536.

Operation

If you have not yet set up your test, first go to Signal Path Setup to select and configure inputs and outputs, references and other settings. Read about the Signal Path Setup menu in Chapter 6.

Start

To make an Acoustic Response measurement, click Start. The generator will output a continuous sweep to the DUT on the selected generator channels at the level set in the Generator panel. In a short period of time, the signal will be acquired and processed, and each of the small graphs in the Selector will be populated with results. Simply click the result view you want.

Be sure that you have set the generator Level low enough that the loudspeaker will not be damaged. Also, be sure you have set the generator Start and Stop frequencies so that the loudspeaker will not be damaged by out-of-band signals. See Loudspeaker Testing using Acoustic Response beginning on page 173, for more information.

Repeat

When the Repeat checkbox is checked, the sweep (or nested sweep) repeats continually. Unchecking the Repeat checkbox or clicking Stop will cancel Repeat.

Note: if Append Graph Data is checked while Repeat is set, a great deal of data will be quickly amassed and the computer memory or storage may be overwhelmed.

Append Graph Data

Measured data are grouped in a Data Set, which contains all acquisition results. Normally, the current graph data are deleted each time you start a new measurement. If the Append Graph Data box is checked, appended data are grouped as additional data sets, as are any imported data. The Data Sets Panel displays the Data Set(s) in the measurement, and provides a visibility checkbox, a Time acquisition field, and a Notes field for each data set.

Read more about Data Sets beginning on page 573.

Primary Source

The primary source for a continuous sweep measurement is the log-swept sine signal. The range between the Start and Stop frequencies provides the X-axis for most of the result graphs.

Start Frequency and Stop Frequency

These controls set the frequency range of the sweep.

EQ

You can equalize the generator signal to make a pre-emphasized stimulus. See Generator Equalization (page 170) for a general discussion of this feature.

Choosing generator equalization

Select None, Relative or Absolute from the EQ drop-down menu.

- None applies no equalization to the generator.
- Relative applies the gain or loss specified in the EQ table to the current generator level for each channel.
- Absolute sets the generator level for all channels to the level specified in the EQ table. Note: when Absolute is selected, current channel level settings are lost.

Edit

If you have chosen to equalize the generator signal, you can edit the EQ table here. See Edit EQ Table (page 170).

Pre-Sweep and Sweep duration fields

A continuous sweep stimulus consists of a sine wave swept logarithmically across the sweep range in the period of time set in these fields.

To enable the DUT to stabilize before the beginning of the defined sweep measurement, there is variable Pre-Sweep time. The actual sweep stimulus signal begins below the defined Start Frequency and is swept upward, reaching the Start Frequency at the end of the Pre-Sweep time. The sum of the times in Pre-Sweep and Sweep is the total sweep length.

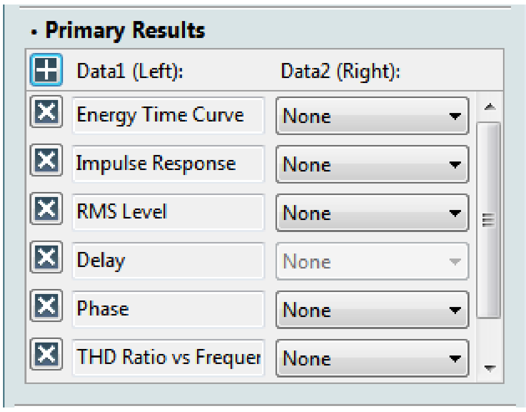
- The default for Pre-Sweep is 100.0 ms. Minimum is 0 s; maximum is 1.0 s.
- The default for Sweep is 350.0 ms. Minimum is 50.0 ms; maximum is 5 s.

Longer Pre-Sweep settings allow more time for DUT stabilization, but increase measurement time. Longer Sweep settings provide greater resolution and signal-to-noise ratio, but increase measurement time.

Primary Results

In Bench Mode Acoustic Response, Primary results include 16 sets of data derived from the sweep.

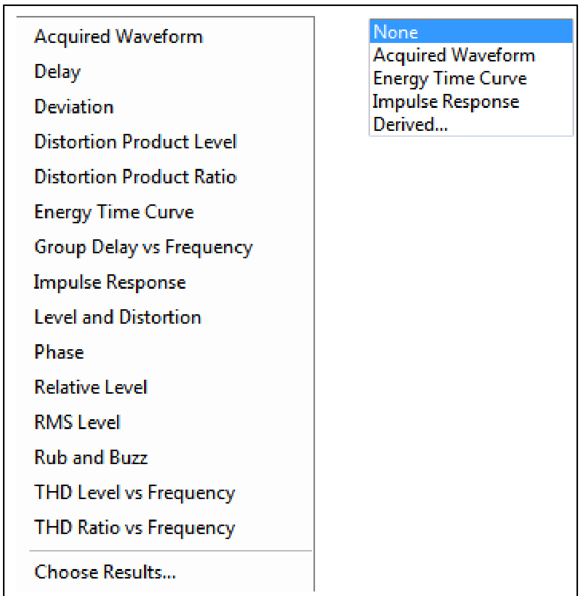
Initially, only six results are set, plotted as Data1 on the left Y axis of the graph.



To add more primary results to the left axis, click the Add Data1 (Left) button and select one or more results.

To add a primary or derived result to the right axis, open the Data2 (Right) menu for a given existing Data1 result, and select a result.

Available choices will vary with analyzer model, hardware and Signal Path Setup configuration. Choices may include:

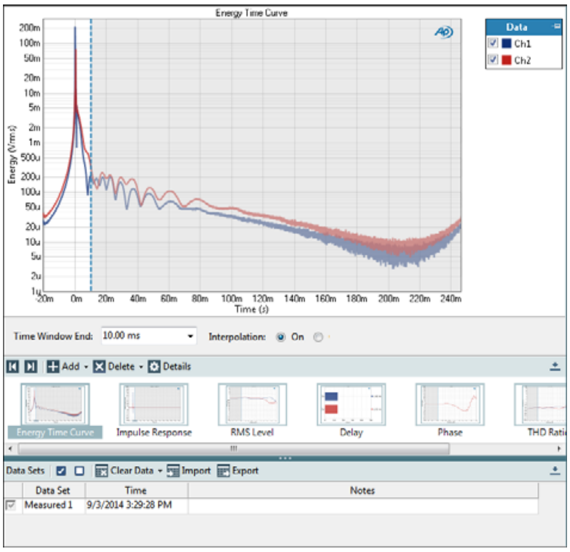


Data 1 Menu Data 2 Menu

Remove a row by clicking the **Delete** button. Both left and right axis results are removed. Results can also be added or deleted from the Selector filmstrip.

Note that in the Data 1 menu, you can select **Choose Results...** to open the **Add Primary Results** dialog, where you can add multiple results. Also, in the **Data 2** menu, you can select **Derived...** to add **Derived Results**. See Chapter 95 for more information about Derived Results.

Typical Result Display



Acoustic Response result displays are shown in detail beginning on page 176.

Settling

Continuous sweep results are always unsettled.

Extend Acquisition By

By default, a continuous sweep measurement makes an acquisition slightly longer than the stimulus, to include possible time-delayed artifacts created in the DUT. By default, the acquisition is extended 50 ms longer than the stimulus.

In some cases, you may want to extend the acquisition further. Enter a new value in the **Extend Acquisition By:** field. Minimum extension is 0 s; maximum is 3 s.

Auto Start Time

When **Auto Start Time** is checked, the **Time Window Start** adjustment control is not available, and **Time Window Start** is automatically set to the generator **Start** time. **Time Window End** is initially set just after the decay of the impulse, but can be adjusted. This is the default behavior and is correct for the vast majority of acoustic tests. For more detail and optional manual control of **Time Window Start**, see Adjusting the Time Window on page 176, and Deselecting Auto Start Time on page 177.

Averages

Averaging several sweeps can remove uncorrelated data from the results (for acoustic measurements, this is usually background noise). The sweep will be run the number of times set in the Averages field, and the results will be averaged for display.

Nesting

Bench Mode Acoustic Responses can be nested. See Nested Sweeps on page 161.

Secondary Source

Select the secondary sweep source here. This is the outside, nesting parameter that is varied to change the conditions for the primary sweep through a number of iterations. Available choices are dependent upon instrument hardware, options, Input/Output settings and Bench Mode Generator and other settings.

Section V: Reference

89

IEC60958 Metadata

Set Status Bits / Set User Bits

For more information about status bits and user bits, go to page 355.

This topic discusses **setting** the status bits and user bits to be embedded in the digital output signal. For information about **reading** the status bits on a digital input, go to the Metadata Monitor: Status Bits on page 32. For information about recording changes in status bits or users bits, see Chapter 58, the Metadata Recorder.

Status bits are supported in the transport streams for the Digital Balanced, Digital Unbalanced, Digital Optical and Digital HDMI inputs.

The Digital Serial Input/Output (DSIO) protocol does not support embedded metadata.

Setting Status Bits

In one of the supported digital modes, from Signal Path Setup click **Settings > Status/User Bits**. The **Set Status Bits / User Bits** dialog will open.

Auto

When **Auto** is selected, APx500 chooses the consumer application or the professional application depending upon your digital output choices. The selected application is shown in the checkbox name: **Auto (Consumer)**, for example. In **Auto**, all status bits are set to default values. User bits are set to 00.

When **Auto** is not selected, the status bits and user bits can be set to any available value.

Subframes A and B are the same (not available in HDMI)

Each frame of audio in the transport stream has 2 subframes, A and B. These are sometimes called channels A and B. For the AES3/SPDIF transport stream, the status bits for A and B are typically set to the same values, but can be set independently.

When **Subframes A and B are the same** is checked, any setting made in subframe A status bits is copied to subframe B. When **Subframes A and B are the same** is not checked, the entry fields for subframe B become available, and status bits for the subframes can be set independently.

Note: some early versions of APx instrument hardware do not support independent setting of subframe A and subframe B status bits.

For HDMI, the Status Bits for subframes A and B are always set to the same values, and this control is not available.

Auto Increment Local Address

This setting is only available when the instrument is fitted with an Advanced Digital I/O module. When checked, the Local Address increments in integers at the sample rate.

Auto Increment Time of Day

This setting is only available when the instrument is fitted with an Advanced Digital I/O module. When checked, the Time of Day increments in integers at the sample rate.

User Defined CRC ChA

This setting is only available when the instrument is fitted with an Advanced Digital I/O module. Enter any hex number in the adjacent field.

User Defined CRC ChB

This setting is only available when the instrument is fitted with an Advanced Digital I/O module. Enter any hex number in the adjacent field.

Display Grids

Plain Text display and Hex display

Status bits can be set from the grid displays, reading the plain text interpretations and choosing settings from drop-down menus in the grid. Status bits can also be set by entering hex values for selected bytes in the Status Bits hex display. These two displays are coupled, and a change in one will be reflected by the corresponding change in the other.

User bits

You can arbitrarily set user bits to any available value by entering hex values for selected bytes in the User Bits hex display. User bits are associated with frames, not subframes; they are not affected by status bit subframe A or B settings.

> *Note: some early versions of APx instrument hardware do not support setting of user bits.*

How Digital Output Configuration Choices Affect Status Bits

In APx500, making a digital output format selection also affects digital metadata by setting the application for the Auto mode on the Set Status Bits / User Bits panel. The application for Auto mode is set as follows:

Digital Unbalanced Output

You can set Digital Unbalanced to either Application: Consumer or Application: Professional on the Settings panel for this output. The default is Application: Consumer. Note that these choices also set the output voltage for the unbalanced output. See Settings for Digital Unbalanced Output for more information.

The Application settings can be overridden by deselecting Auto on the Set Status Bits / User Bits panel.

The electrical settings persist, regardless of metadata settings.

Digital Balanced Output

The Digital Balanced transport stream is output with professional application status bits, when Auto is set on the Set Status Bits / User Bits panel. This application setting can be overridden by deselecting Auto.

The Digital Balanced transport stream is always output with a peak-to-peak level of 5.0 V, within the recommendations of AES3 and IEC60958-4.

Digital Optical Output

The Digital Optical transport stream is output with consumer application status bits, when Auto is set on the Set Status Bits / User Bits panel. This application setting can be overridden by deselecting Auto.

HDMI Source

The HDMI transport stream carries an interpretation of the AES3/IEC60958 protocol, including status bits. These status bits are set to consumer application, when Auto is set on the Set Status Bits panel. This application setting can be overridden by deselecting Auto.

HDMI ARC Tx

The ARC transport stream carries IEC60958 audio, although with different physical layer characteristics. Status bits are set to consumer application, when Auto is set on the Set Status Bits panel. This application setting can be overridden by deselecting Auto.

Digital Serial Output

The Digital Serial Input/Output (DSIO) protocol does not support embedded metadata.

PDM

The PDM Input/Output does not support embedded metadata.

90

Supported Audio File Formats

...for import as Generator Waveforms

The following audio file formats can be imported as Generator Waveforms (Arbitrary Waveforms).

Format	*Detail*	*Sample Rate*	*Filename Extension*
Linear PCM			
WAVE_FORMAT_PCM (Type 1)	8 to 32 bit mono or stereo†	Any rate in the range of 4 kHz—216 kHz	*.wav
WAVE_FORMAT_EXTENSIBLE	8 to 32 bit multichannel†	Any rate in the range of 4 kHz—216 kHz	*.wav
WAVE_FORMAT_PCM (proprietary mapping)*	8 to 32 bit multichannel†	Any rate in the range of 4 kHz—216 kHz	*.wav
AP legacy waveform	24 bit mono or stereo	Any rate in the range of 4 kHz—216 kHz	*.agm, *.ags
Coded			
Dolby Digital	5.1 surround	32 kHz or 48 kHz	*.ac3
Dolby Digital Plus	7.1 surround	48 kHz	*.ec3
Dolby TrueHD	8 channel	48 kHz, 96 kHz and 192 kHz	*.mlp
MAT (Dolby HD container)	stream	48 kHz, 96 kHz and 192 kHz	*.mat
dts Digital Surround	5.1 surround	48 kHz and 96 kHz	*.dts
dts Digital Surround compact	5.1 surround	48 kHz and 96 kHz	*.cpt
dts-HD High Resolution Audio	6 channel	48 kHz and 96 kHz	*.dtshd
dts-HD Master Audio	8 channel	48 kHz, 96 kHz and 192 kHz	*.dtshd

* Ambiguities within WAVE_FORMAT_PCM allow undocumented multiple channel configurations. This particular proprietary format is in common use, and can be created in Sony Sound Forge and other waveform editors.

† For 32-bit audio, proprietary 32-bit fixed point .wav formats are supported.

...for import as File Input

The following audio file formats can be imported for analysis using the File Input feature. Note that only files from acquisitions saved by APx500 will contain cross-domain scaling data.

Format	*Detail*	*Sample Rate*	*Filename Extension*
Linear PCM			
WAVE_FORMAT_PCM (Type 1)	8 to 32 bit mono or stereo†	Any rate in the range of 4 kHz—216 kHz	*.wav
WAVE_FORMAT_EXTENSIBLE	8 to 32 bit multichannel†	Any rate in the range of 4 kHz—216 kHz	*.wav
WAVE_FORMAT_PCM (proprietary mapping)*	8 to 32 bit multichannel†	Any rate in the range of 4 kHz—216 kHz	*.wav

* Ambiguities within WAVE_FORMAT_PCM allow undocumented multiple channel configurations. This particular proprietary format is in common use, and can be created in Sony Sound Forge and other waveform editors.

† For 32-bit audio, proprietary 32-bit fixed point .wav formats are supported.

...for export as Record to File (linear audio)

These audio file formats can be exported and saved to disc from Measurement Recorder, Noise Recorder and Multitone Recorder.

Format	*Detail*	*Sample Rate*	*Filename Extension*
Linear PCM			
WAVE_FORMAT_PCM (Type 1)	8 to 32 bit mono or stereo†	For analog acquisitions, 6 kHz, 8 kHz, 12 kHz, 16 kHz, 24 kHz, 44.1 kHz, 48 kHz, 96 kHz or 192 kHz. If fitted with BW52, 624 kHz, 1248 kHz and 2496 kHz are also available. For digital acquisitions, files are saved at the same sample rate as the acquisition.	*.wav
WAVE_FORMAT_EXTENSIBLE	8 to 32 bit multichannel†	as above.	*.wav
WAVE_FORMAT_PCM (proprietary mapping)*	8 to 32 bit multichannel†	as above.	*.wav

* Ambiguities within WAVE_FORMAT_PCM allow undocumented multiple channel configurations. This particular proprietary format is in common use, and can be created in Sony Sound Forge and other waveform editors.

† For 32-bit audio, proprietary 32-bit fixed point .wav formats are supported.

...for export as Record to File (coded audio)

These audio file formats can be exported and saved to disc from Measurement Recorder, digital acquisitions only. Coded audio (typically multichannel Dolby or DTS surround sound, coded in an IEC61937 bitstream) is saved in the incoming format.

Format	*Sample Rate*	*Filename Extension*
Coded Audio		
Dolby Digital	File is saved at the same sample rate as the acquisition.	*.ac3
Dolby Digital Plus	File is saved at the same sample rate as the acquisition.	*.ec3
Dolby TrueHD	File is saved at the same sample rate as the acquisition.	*.mlp
dts Digital Surround	File is saved at the same sample rate as the acquisition.	*.dts
dts Digital Surround compact	File is saved at the same sample rate as the acquisition.	*.cpt
dts-HD	File is saved at the same sample rate as the acquisition.	*.dtshd
raw encoded bitstream	File is saved at the same sample rate as the acquisition.	*.bin

91

Bandwidth Limiting and Weighting Filters

Introduction

Bandwidth limiting or weighting filters are commonly used in distortion and noise measurements. Bandwidth limiting filters cut off the extremes of the frequency spectrum to eliminate low or high frequency signals that might confuse results. Weighting filters approximate the characteristics of a transmission system or of human hearing with a more complex filter curve.

Always state the filtering used when reporting distortion or noise results.

Signal Path Setup: Filters

New with APx500 version 4.0 are Signal Path Filters. In Sequence Mode, each signal path has a set of filters that, when applied, affect the entire signal path. Bench Mode Signal Path Setup has the same set of filters available.

Local Measurement Filters

In addition to the Signal Path Filters, some Sequence Mode measurements and some Bench Mode Analyzer features have local filters available.

Signal Path Filters

The Filters controls on the Setup > Input/Output panel enable high-pass, low-pass, bandwidth and weighting filters. In Sequence Mode, these filters affect all the incoming audio throughout the current signal path; in Bench Mode, the filters affect all the incoming audio for the entire Bench Mode project.

> *Note: We do not recommend using input filtering when using the Impedance/Thiele-Small measurement or the Loudspeaker Production Test measurement. Impedance curves derived with input filtering in effect might be in error. For low-frequency drivers, be sure that the input signal path is DC-coupled.*

High-pass (analog or digital audio signals)

- DC
  This selection provides no high-pass filtering. The signal is DC coupled.

Read more about DC in APx analyzers beginning on page 240.

- AC (<10 Hz)
  This selection provides AC coupling flat to 20 Hz, and about 0.1 dB down at 10 Hz.
- Butterworth
  You can specify a custom 4-pole Butterworth high-pass filter here, by entering a value for the corner frequency in the field; the acceptable range is 10 Hz to 90 kHz (standard analog input), 10 Hz to 1 MHz (APx555 or BW52 analog input), 10 Hz to 100 kHz (digital input). The signal is AC coupled.
- Elliptic
  You can specify a custom 5-pole elliptic high-pass filter here, by entering a value for the corner frequency in the field; the acceptable range is 10 Hz to 90 kHz (standard analog input), 10 Hz to 1 MHz (APx555 or BW52 analog input), 10 Hz to 100 kHz (digital input). The filter has a passband ripple of 0.01 dB and a sharp roll-off to –60 dB. It closely matches the response of the frequency-domain filters in earlier versions of APx500. The signal is AC coupled.

High-pass (jitter signals)

- 700 Hz (AES3)
  This filter conforms to the AES3 recommendation for intrinsic jitter measurement.
- Butterworth
  You can specify a custom 4-pole Butterworth high-pass filter here, by entering a value for the corner frequency in the field; the acceptable range is 50 Hz to 150 kHz. The signal is AC coupled.
- Elliptic
  You can specify a custom 5-pole elliptic high-pass

filter here, by entering a value for the corner frequency in the field; the acceptable range is 50 Hz to 150 kHz. The filter has a passband ripple of 0.01 dB and a sharp roll-off. The signal is AC coupled.

Note that the corner frequency for the high-pass filter cannot be higher than the corner frequency for the low-pass filter.

Low-pass (analog audio signals)

ADC Passband

All analysis in APx analyzers is performed in digital signal processing (DSP), so all analog input signals are digitized using high-performance analog-to-digital converters (ADCs). To avoid aliasing, ADC designs effectively filter the audio to just below 1/2 the sample rate (Fs/2, the Nyquist frequency). Therefore, selecting an ADC sample rate sets a de facto low-pass filter. In APx500 this filtering is referred to as *ADC Passband*.

Bandwidth

For ADC Passband, these bandwidth choices are available:

- 20 kHz (SR 48 kHz)
- 45 kHz (SR 96 kHz)
- 90 kHz (SR 192 kHz)
- 250 kHz (SR 624 kHz) (APx555 or BW52 option)
- 500 kHz (SR 1.248 MHz) (APx555 or BW52 option)
- 1 MHz (SR 2.496 MHz) (APx555 or BW52 option)

AES17 (20 kHz)

This filter is an 8-pole elliptic filter with a corner frequency at 20 kHz, satisfying the AES17 recommendation for converter measurements and other measurements in the presence of high out-of-band noise. When this filter is selected, the ADC sample rate is set to 48 kHz.

AES17 (40 kHz)

This filter is an 8-pole elliptic filter with a corner frequency at 40 kHz, satisfying the AES17 recommendation for converter measurements and other measurements in the presence of high out-of-band noise. When this filter is selected, the ADC sample rate is set to 96 kHz.

Butterworth

You can specify a custom 8-pole low-pass Butterworth filter here, by entering a value for the corner frequency in the field. The acceptable range is 10 Hz to 1 MHz (when supported by hardware). An 8-pole Butterworth filter is not as sharp as an elliptic filter. The Butterworth filter is –3 dB at the corner frequency and falls off at a rate of 48 dB per octave.

When this filter is selected, the ADC sample rate is set to the next range above the requested corner frequency. The final filter curve will reflect the combination of the Butterworth filter and the ADC Passband filter.

Elliptic

You can create a custom 8-pole low-pass elliptic filter here, by entering a value for the corner frequency in the field. The acceptable range is 1 kHz to 1 MHz. When this filter is selected, the ADC sample rate is set to the next range above the requested corner frequency. The elliptic filter has a passband ripple of 0.01 dB and a very sharp roll-off to –60 dB.

When this filter is selected, the ADC sample rate is set to the next range above the requested corner frequency. The final filter curve will reflect the combination of the elliptic filter and the ADC Passband filter. Since the elliptic filter has such a sharp roll-off, the combined response will be dominated by the elliptic filter.

Low-pass (digital signals)

Fs/2

The bandwidth of a digital audio signal cannot exceed one-half the sample rate (Fs/2). The Fs/2 setting passes a digital signal at its maximum bandwidth, without any additional low-pass filtering.

Butterworth

You can specify a custom 8-pole low-pass Butterworth filter here, by entering a value for the corner frequency in the field. The acceptable range is 10 Hz to 100 kHz.

Elliptic

You can specify a custom 8-pole low-pass elliptic filter here, by entering a value for the corner frequency in the field. The acceptable range is 10 Hz to 100 kHz.

Low-pass (jitter signals)

- 700 Hz (AES3)
  This filter conforms to the AES3 recommendation for intrinsic jitter measurement.
- Butterworth
  You can specify a custom 8-pole Butterworth high-pass filter here, by entering a value for the corner frequency in the field; the acceptable range is 50 Hz to 150 kHz. The signal is AC coupled.
- Elliptic
  You can specify a custom 8-pole low-pass elliptic filter here, by entering a value for the corner frequency in the field; the acceptable range is 50 Hz

to 150 kHz. The filter is implemented in the time domain, and has a passband ripple of 0.01 dB and a sharp roll-off. The signal is AC coupled.

Note that the corner frequency for the low-pass filter cannot be lower than the corner frequency for the high-pass filter.

Weighting

APx500 provides weighting and deemphasis filters for noise and distortion measurements. When applied from Signal Path Setup, weighting filters affect the full signal. This may be appropriate for a noise measurement, but for a THD+N measurement it is recommended that you use a filter dedicated to a THD+N measurement or meter, which will only filter the residual signal.

See Local Measurement Filters, below.

APx500 provides these weighting filters:

- A-wt.
- B-wt.
- C-wt.
- CCIR-1k
- CCIR-2k
- CCITT
- C-message

Deemphasis filters are also included, for use with pre-emphasized DUTs or stimulus signals.

- 50us de-emph.
- 75us de-emph.
- 50us de-emph. + A-wt
- 75us de-emph. + A-wt.

Local Measurement Filters

Measurement filters are “local”, affecting only the measurement (Sequence Mode) or meters (Bench Mode) to which the filter is attached.

Sequence Mode

Many Sequence Mode measurements have filter controls that affect that measurement only. These filters use the same DSP algorithms and have the same characteristics as an equivalent Signal Path Filter.

High-pass (analog or digital audio signals)

- Signal Path
  This selection applies no local high-pass filter at the measurement level. Only the filtering set in Signal Path Setup > Filters (which could range from none to highly filtered) will be in effect.
- Butterworth
  You can specify a custom 4-pole Butterworth high-pass filter here, by entering a value for the corner frequency in the field; the acceptable range is 10 Hz to 90 kHz (standard analog input), 10 Hz to 1 MHz (APx555 or BW52 analog input), 10 Hz to 100 kHz (digital input). The signal is AC coupled. Selecting a filter here overrides any high-pass filter set in Signal Path Setup > Filters.
- Elliptic
  You can specify a custom 5-pole elliptic high-pass filter here, by entering a value for the corner frequency in the field; the acceptable range is 10 Hz to 90 kHz. The filter has a passband ripple of 0.01 dB and a sharp roll-off to -60 dB. It closely matches the response of the frequency-domain filters in earlier versions of APx500. The signal is AC coupled. Selecting a filter here overrides any high-pass filter set in Signal Path Setup > Filters.

High-pass (jitter signals)

- Signal Path
  This selection applies no local high-pass filter at the measurement level. Only the filtering set in Signal Path Setup > Filters (which could be none, or substantial) will be in effect.
- Butterworth
  You can specify a custom 4-pole Butterworth high-pass filter here, by entering a value for the corner frequency in the field; the acceptable range is 50 Hz to 150 kHz. The signal is AC coupled. Selecting a filter here overrides any high-pass filter set in Signal Path Setup > Filters.
- Elliptic
  You can specify a custom 5-pole elliptic high-pass filter here, by entering a value for the corner frequency in the field; the acceptable range is 50 Hz to 150 kHz. The filter has a passband ripple of 0.01 dB and a sharp roll-off to –60 dB. The signal is AC coupled. Selecting a filter here overrides any high-pass filter set in Signal Path Setup > Filters.

Note that the corner frequency for the high-pass filter cannot be higher than the corner frequency for the low-pass filter.

Low-pass (analog audio signals)

- Signal Path
  This selection applies no local low-pass filter at the measurement level. Only the filtering set in Signal Path Setup > Filters (which could range from none to highly filtered) will be in effect.
- Butterworth
  You can specify a custom 8-pole low-pass Butterworth filter here, by entering a value for the corner frequency in the field. The acceptable range is

10 Hz to 1 MHz (when supported by hardware). An 8-pole Butterworth filter is not as sharp as an elliptic filter. The Butterworth filter is –3 dB at the corner frequency and falls off at a rate of 48 dB per octave. Selecting a filter here overrides any low-pass filter set in Signal Path Setup > Filters.

Note that you cannot specify a corner frequency higher than the current ADC passband upper frequency limit.

- Elliptic
  You can create a custom 8-pole low-pass elliptic filter here, by entering a value for the corner frequency in the field. The acceptable range is 1 kHz to 1 MHz. When this filter is selected, the ADC sample rate is set to the next range above the requested corner frequency. The elliptic filter has a passband ripple of 0.01 dB and a very sharp roll-off. Selecting a filter here overrides any low-pass filter set in Signal Path Setup > Filters.

Note that you cannot specify a corner frequency higher than the current ADC passband upper frequency limit.

Low-pass (digital audio signals)

- Fs/2
  The bandwidth of a digital audio signal cannot exceed 1/2 the sample rate (Fs/2). The Fs/2 setting passes a digital signal at its maximum bandwidth, without any additional low-pass filtering.
- Butterworth
  You can specify a custom 8-pole low-pass Butterworth filter here, by entering a value for the corner frequency in the field. The acceptable range is 10 Hz to 100 kHz.
- Elliptic
  You can specify a custom 8-pole low-pass elliptic filter here, by entering a value for the corner frequency in the field. The acceptable range is 10 Hz to 100 kHz.

Low-pass (jitter signals)

- 700 Hz (AES3)
  This filter conforms to the AES3 recommendation for intrinsic jitter measurement.
- Butterworth
  You can specify a custom 8-pole low-pass Butterworth filter here, by entering a value for the corner frequency in the field; the acceptable range is 50 Hz to 150 kHz. The filter is implemented in the time domain. The signal is AC coupled.
- Elliptic
  You can specify a custom 8-pole low-pass elliptic filter here, by entering a value for the corner frequency in the field; the acceptable range is 50 Hz to 150 kHz. The filter is implemented in the time domain, and has a passband ripple of 0.01 dB and a sharp roll-off. The signal is AC coupled.

Note that the corner frequency for the low-pass filter cannot be lower than the corner frequency for the high-pass filter.

Weighting

APx500 provides weighting and deemphasis filters for noise and distortion measurements. A weighting filter, if selected, is applied to the noise or distortion results only, but not to the entire acquisition or measurement. The filters have the same characteristics as the Signal Path weighting filters, listed previously in this chapter.

Relationship between Sequence Mode Signal Path and Measurement Filters

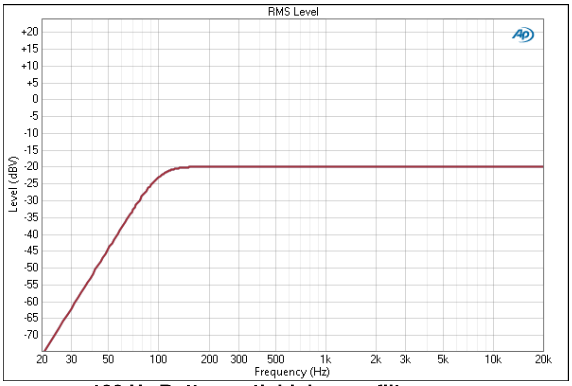
Keep these rules in mind:

1. Setting a measurement (local) low-pass or high-pass filter overrides the corresponding Signal Path filter, while the measurement is active. Measurement (local) low-pass and high-pass filters affect the full signal for that measurement.
2. Setting a measurement (local) weighting filter DOES NOT override the Signal Path weighting filter; also, the measurement (local) weighting filter affects only the THD+N residuals. If you set both the Signal Path and measurement weighting filters, you will be filtering the signal twice: once the full signal, once the residuals only.
3. When a custom low-pass filter is requested as a Signal Path filter, the ADC sample rate may be shifted to accommodate the request, as explained in the previous section.
   However, when a custom low-pass filter is requested as a measurement (local) filter, the ADC sample rate is never changed; instead, the corner frequency entry field is constrained to be less than the bandwidth allowed by the current ADC sample rate.

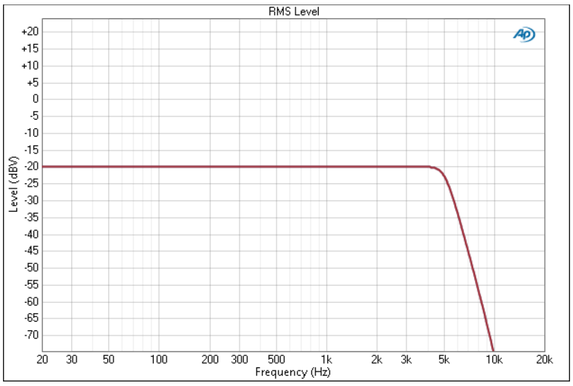
Bench Mode

The Bench Mode analyzer provides a “local” THD+N Weighting filter that is only active for THD+N and Noise meters, with the same selections and characteristics as the Sequence Mode Weighting measurement filters described above.

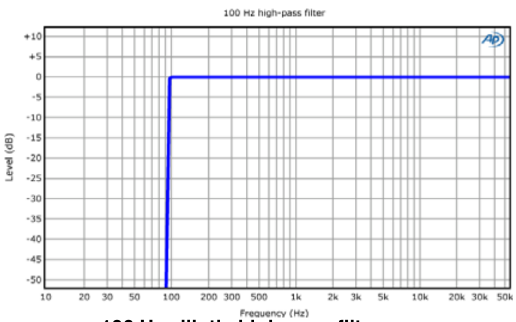
Typical Filter Curves



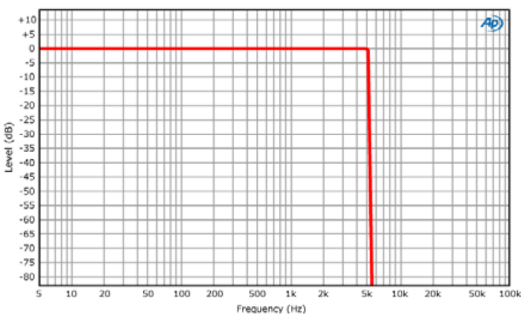
100 Hz Butterworth high-pass filter curve



5 kHz Butterworth low-pass filter curve

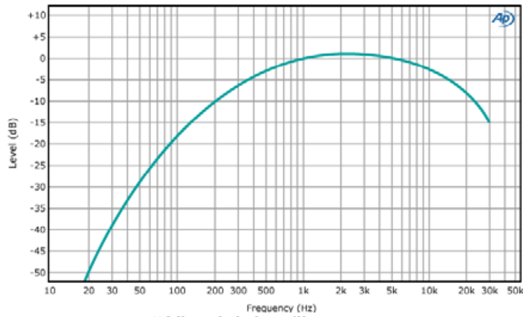


100 Hz elliptic high-pass filter curve

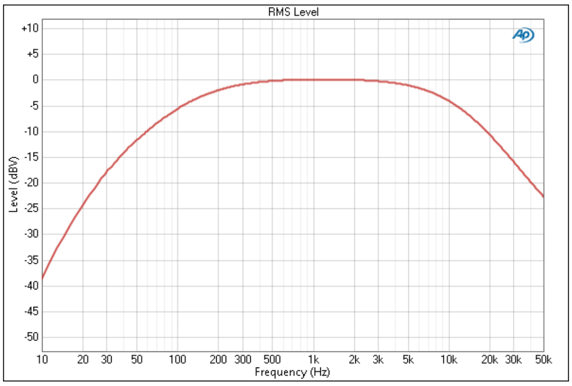


5 kHz elliptic low-pass filter curve

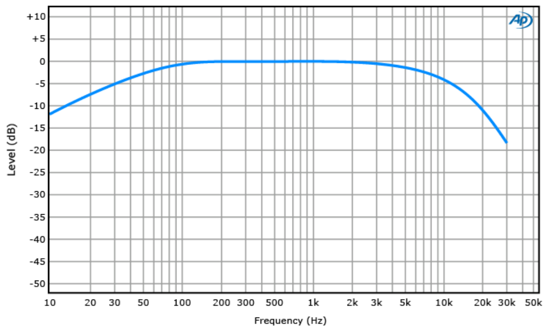
Weighting and deemphasis filter curves



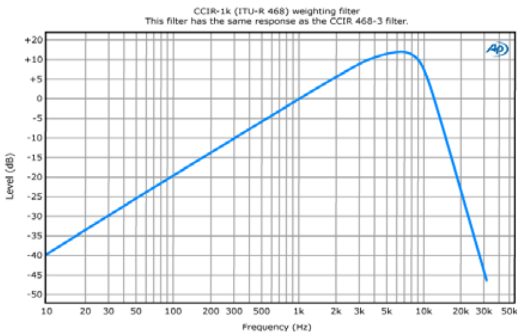
"A" weighting filter curve



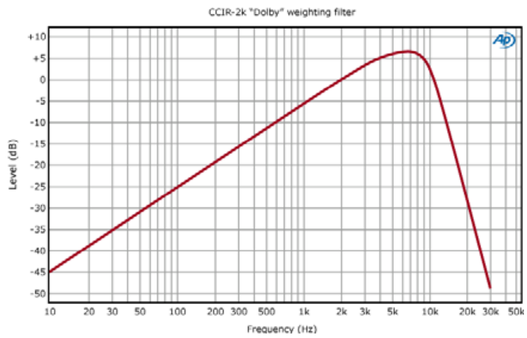
"B" weighting filter curve



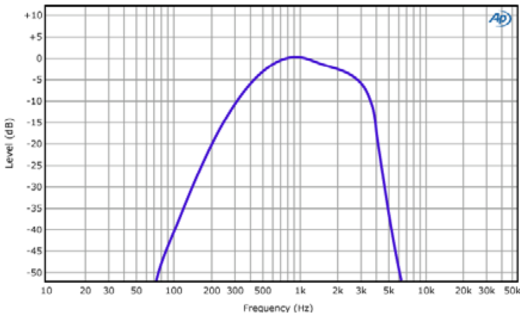
"C" weighting filter curve



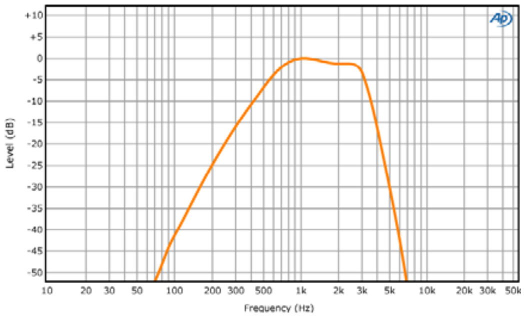
CCIR-1k (ITU-R 468) weighting filter curve



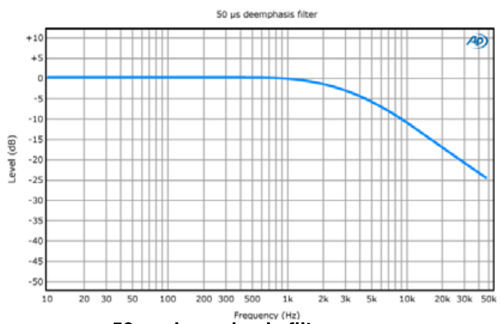
CCIR-2k “Dolby” weighting filter curve



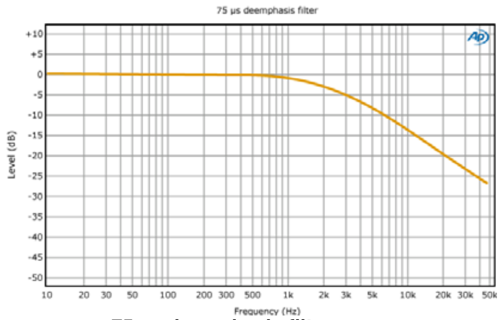
CCITT weighting filter curve



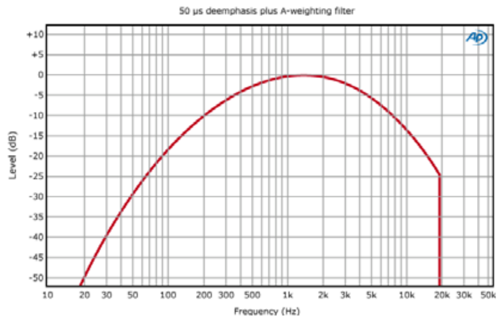
C-Message weighting filter curve



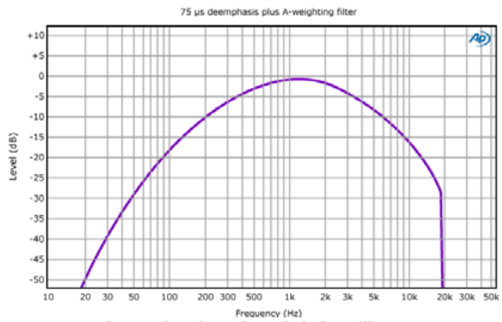
50 μs deemphasis filter curve



75 μs deemphasis filter curve



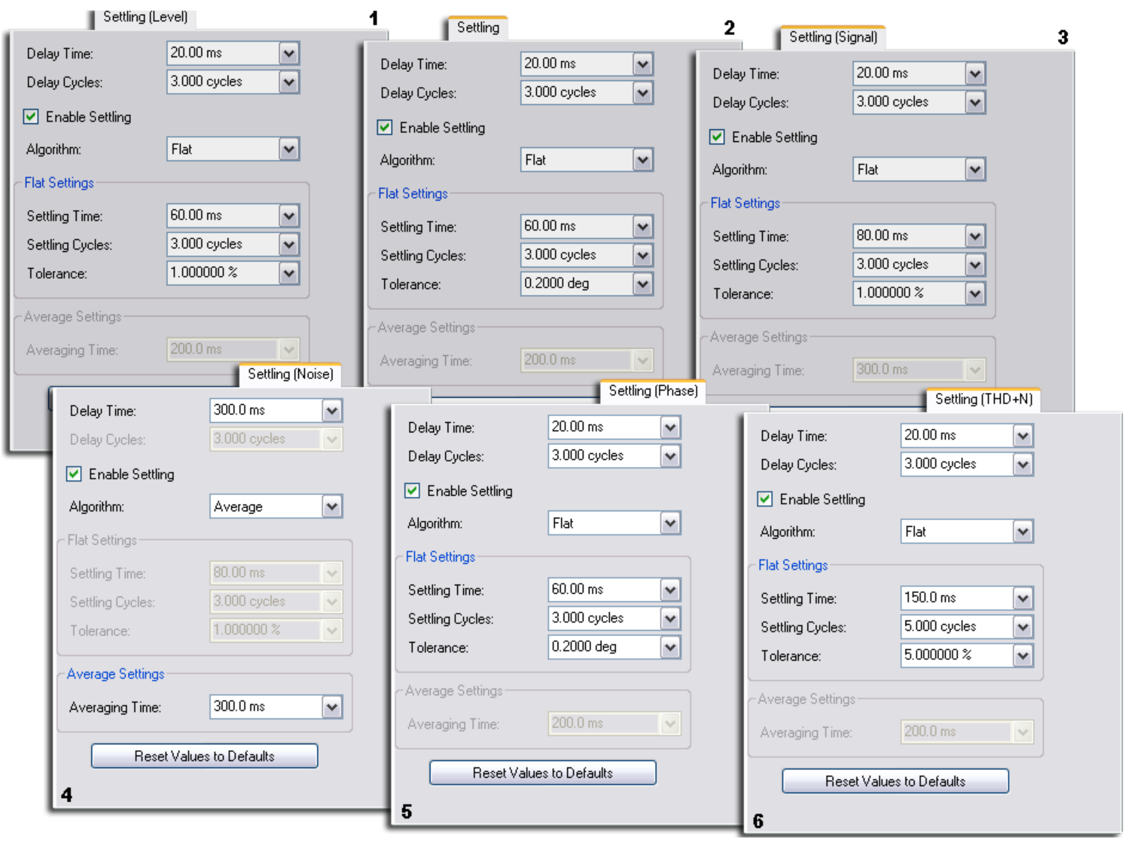
50 μs deemphasis + A weighting filter curve



75 μs deemphasis + A weighting filter curve

92

Ranging and Settling



Ranging

Audio amplifier circuits are required to operate over vast dynamic ranges: 80, 100 or even 120 dB or more. Distortion and noise at the extremes are givens.

The need for ranging

The input circuits of an audio meter or analyzer employ switchable attenuation or gain to accommodate the extremes of dynamic range for accurate, high-resolution measurement.

Fixed ranging

When an input signal is low, the analyzer gain should be set to amplify that signal to an optimum level for measurement. When an input signal is high, the analyzer gain should be reduced to prevent overload and to bring the signal down into the optimum measurement range.

Most meters and analyzers provide a way for you to manually set ranging to accommodate the input signal level. In Audio Precision instruments, the ranging choices are shown in a dialog that allows you to select the maximum level you expect.

For the APx585 and 586, six ranges are available for both unbalanced and balanced inputs:

- 0 Vrms to 320 mVrms
- 320 mVrms to 1 Vrms
- 1 Vrms to 3.2 Vrms
- 3.2 Vrms to 10 Vrms
- 10 Vrms to 32 Vrms
- 32 Vrms to 100 Vrms

For the APx525 family, a seventh range is added. For the unbalanced inputs it is:

- 100 Vrms to 160 Vrms

For the APx525 family balanced inputs, the seventh range is extended to a higher voltage:

- 100 Vrms to 300 Vrms

Choose the lowest range whose upper limit is greater than the expected maximum input signal.

Autoranging

Autoranging employs a circuit that measures the input voltage and automatically selects the correct range for the signal. As the signal level varies, the range will be switched up or down to keep the signal in the best range for measurement.

Autoranging works well for steady signals or signals whose levels change slowly. Most test stimulus signals are consistent, making autoranging a good choice for most testing.

When autoranging isn't suitable

However, signals that are very dynamic (such as music, voice or noise signals) may cause the autoranging circuits to switch too often for your test purposes. Signals with very fast pulses or spikes change too quickly for the autoranging circuit to react and may cause incorrect measurements due to ranging errors.

In these cases, evaluate the signal range carefully, or estimate the range if you must, and manually set the analyzer ranging to accommodate the input signal.

Also, autoranging takes time. When the signal levels are known and a sequence would require many range changes, setting ranging to Fixed will provide faster testing.

Track first channel range

For convenience, in APx500 ranging for channels above 1 are set to follow the channel 1 range setting. This behavior can be defeated by unchecking the **Track first channel range** checkbox.

Autoranging for Continuous Sweep and Frequency Response measurements

Both Continuous Sweep and Frequency Response use a modified autoranging process to find the correct input range.

For Continuous Sweep and Frequency Response, the sweep acquisition begins in a particular range called the range floor, and moves up from there if necessary. By default, the range floor is the lowest range (0 Vrms to 320 mVrms), but you can specify a higher range by entering the value in the **Min Range** field in the Advanced Settings dialog.

If the initial range is correct for the signal, the acquisition is made and the data are processed and displayed. If this range is too low, data from the out-of-range sweep are discarded. The input range is moved up and the sweep is repeated. This process may result in several sweep attempts before the correct range is determined; however, since the sweeps are very fast (typically one second) the total acquisition time is short.

If your sweep is a step in an automated sequence that you would like to run as fast as possible, you can optimize the speed of the ranging process by setting the range floor to the correct range for the measurement.

Settling

Practical devices will take a finite time to stabilize when an input signal changes, and at each sweep point a DUT will have an initial moment of instability. Also, practical audio analyzers require a minimum time to acquire sufficient data for proper analysis. These instabilities can vary with frequency, level, periodicity of signal and other factors. To provide consistent and accurate measurements, APx500 uses settling mechanisms to obtain stable, settled readings.

When settling is enabled, the analyzer pauses before reporting the result for each measurement or sweep point, providing time for DUT stabilization. The settling mechanisms attempt to make the pause as short as possible consistent with obtaining a settled reading. Since DUTs often take longer to stabilize when stimulated by a low frequency and acquisitions must be longer for sufficient low frequency resolution, you will notice that the sweep progresses more slowly through the low frequencies. This is due to settling.

APx500's settling mechanisms are optimized for fast, stable readings, but some of the settling parameters are available for adjustment. Depending upon the DUT, the nature of your measurement and your required results, you may find that adjusting the default settling parameters will give you faster or more consistent sweeps.

Settling in Sequence Mode

Measurements that can be settled

Signal to Noise Ratio and Crosstalk measurement are always settled, both for on-screen display and for reports. Stepped Sweeps are settled at each point, both for on-screen display and for reports.

Single value “meter” measurements in APx500 (Level and Gain, THD+N, Interchannel Phase, DC Level and the IMD measurements) appear on-screen as real-time measurements, unsettled.

When a meter measurement is started from the Navigator, (whether singly or as part of a sequence) these measurements are settled before the results are reported.

These defaults can be overridden. Settling can be defeated by unchecking the Enable checkbox and setting delays to zero in the Advanced Settings dialog.

Measurements that cannot be settled

Continuous Sweep measurements (including Frequency Response measurements) do not have settling available. Instead, the stimulus includes a pre-sweep signal to avoid transient-induced “pops” in the DUT.

The Signal Analyzer views are not settled. The Signal Analyzer FFT Spectrum view employs selectable averaging to reduce transient and non-periodic components in the results.

Adjusting Settling Parameters

Settling is normally enabled with default settings. These settings are correct for most measurements and do not require adjustment. Settling is applied when a report is generated, either singly or as part of an automated sequence.

If you would like to optimize the settling for speed or to accommodate a challenging signal, the Settling tab (or tabs) in Advanced Settings provide options.

Settling tabs

Some measurements in APx500 measure only one type of signal and require only one set of settling parameters. Others measure two or even signal types and have independent settling parameters for each. SNR is an example, where the signal and the noise are settled with different settings. Stepped Frequency sweeps have three settling tabs.

In the various Advanced Settings dialogs, one or more Settling tabs allow you to view and adjust the settling parameters. The illustration on page 552 shows the settling tabs and default settings for

1. Settling for Level measurements.
2. Settling for Interchannel Phase measurements.
3. Settling for SNR Signal measurements.
4. Settling for SNR Noise measurements.
5. Settling for Stepped Sweep Phase measurements
6. Settling for Stepped Sweep THD+N measurements.

In the explanations below, the default settings for each Settling tab are described.

Initial delay

To avoid spurious transient signals caused by DUT instability at the onset or change of the stimulus tone, the first moment of the acquisition is not measured. Measurement begins after the initial delay. The initial delay is calculated using the values entered in the Delay Time and Delay Cycles fields. The actual delay time is the greater of:

- the value set in Delay Time, or

  Delay time defines an initial measurement delay in seconds. The effect of this delay is the same at all frequencies.

  Minimum delay time is 0 s.

  Maximum delay time is 10 s.

  Defaults:
  20.00 ms for all measurements except SNR (Noise), which is 300.00 ms.

- the value set in Delay Cycles multiplied by the period of the generated stimulus signal.

  Delay cycles defines an initial measurement delay in number of cycles of the stimulus waveform. This setting produces a greater time delay at low frequencies.

  Minimum delay cycles is 0.

  Maximum delay cycles is 100.

  Defaults:
  3.000 cycles for all measurements except SNR (Noise), for which the Delay Cycles control is not available.

These settings interact to produce a frequency-dependent delay curve with an inflection at the point where Delay Time and the time calculated using Delay Cycles are the same. For SNR (Noise) the delay time is independent of frequency.

Initial delay settings are observed even if the Enable Settling checkbox is unchecked. Initial delay can be set to none by entering 0 in both fields.

Time and Cycles in Settling dialogs

For both the pre-settling Delay and the Flat algorithm Settling window, the calculation of total time from the values entered in the Time and Cycles fields follows the same rules:

The time is the greater of:

- the value set in Delay (or Settling) Time, or
- the value set in Delay (or Settling) Cycles value multiplied by the period of the generated signal.

In both cases, these settings produce intersecting curves, as shown in the diagram. With default values, this has the effect of lengthening the time at lower frequencies.

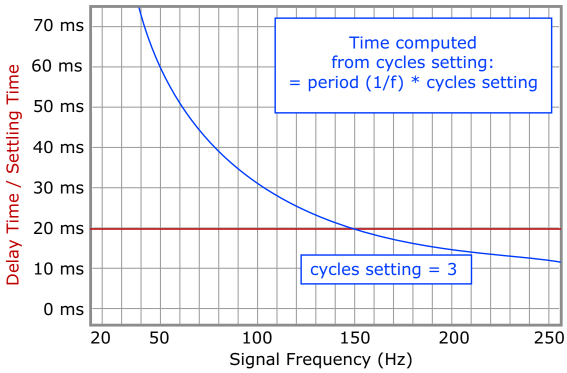


Illustration of both Initial delay and Settling delay calculation.

Example 1

The period of a 1 kHz sine wave is 1 ms. For a measurement at 1 kHz with typical Delay settings, the value of Time = 20 ms, and the calculated value of Cycles is 3 × 1 ms = 3 ms. Actual setting is the greater value, or 20 ms. The 3 cycles setting exceeds 20 ms at 150 Hz, which has a period of 6.667 ms.

Example 2

The period of a 100 Hz sine wave is 10 ms. For a measurement at 100 Hz with the same settings, the value of Time = 20 ms, and the calculated value of Cycles is 3 × 10 ms = 30 ms. Actual setting is the greater value, 30 ms.

Settling algorithms

After the initial delay, APx500 uses one of two settling algorithms. **Flat** is usually more suitable for periodic signals and is the default for most measurements. **Average** is sometimes more suitable for aperiodic (noise-dominated) signals. You can select either algorithm when available. **Enable Settling** must be checked to use a settling algorithm; if the box is unchecked, the signal will be measured directly after the initial **Delay Time / Delay Cycles** time has passed.

> **Defaults:**
> Flat is the default algorithm for all measurements except SNR (Noise), for which Average is the default algorithm.

The Flat Algorithm

The flat algorithm settings describe a time/level window within which a series of measurements must fall for the last point to be considered settled. The width of the window is the time calculated using the values entered in the **Settling Time** and **Settling Cycles** fields. The actual settling time is the greater of:

- the value set in Settling Time {default: 60.00 ms}, or

  In conjunction with Settling Cycles, Settling Time defines the width of the Flat algorithm time window in seconds. The effect of this setting is the same at all frequencies.

  Minimum settling time 0 s.

  Maximum settling time is 10 s.

  Defaults:
  60.00 ms for all measurements except the two Stepped Sweep (THD+N) tabs, which are 150.00 ms, and SNR (Noise) which is 80.00 ms, if Flat is selected.

- the value set in Settling Cycles multiplied by the period of the generated stimulus signal.

  In conjunction with Settling Time, Settling Cycles defines the width of the Flat algorithm time window in seconds. This setting produces a wider time window at low frequencies.

  Minimum settling cycles is 0.

  Maximum settling cycles is 100.

  Defaults:
  3.000 cycles for all measurements except the two Stepped Sweep (THD+N) tabs, which are 5.000 cycles, and SNR (Noise), for which the Settling Cycles control is not available.

These settings interact to produce a frequency-dependent delay curve with an inflection at the point where **Settling Time** and the time calculated using **Delay Cycles** are the same.

The height of the window is the instantaneous measured level multiplied by:

- twice the value set in Tolerance.

  As each acquired point is measured, the Flat algorithm window is redefined around its level. The height of the window is ±{Tolerance percentage}, and is always centered vertically above and below the measured value.

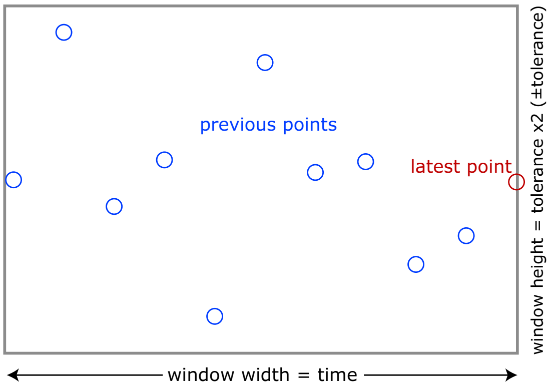
  Minimum tolerance percentage is ±0%.

  Maximum tolerance percentage is ±1000%.

  Defaults:
  1 % for all measurements except the two Stepped Sweep (THD+N) tabs, which are 5 %, and the SNR (Noise), for which the Flat Algorithm selection is not available.

Flat algorithm window

When using the Flat settling algorithm, it is useful to imagine a "window" that qualifies the latest point in a series of measurements as a "settled" measurement.



Flat algorithm settling window.

The Settling Time setting defines the width of the window in time. The current or latest point is shown at the right of the window. Previous points are shown to the left of the latest point. The number of points in the window depends on the measurement rate and the width of the window (Settling Time).

The Tolerance setting defines the height of the window. Since Tolerance is figured as a plus/minus percentage, a 1% Tolerance setting (±1%) defines a window height of 2%. The latest point is centered at 0%.

If any of the points in the window width (Settling Time) fall outside the window height (±Tolerance), the algorithm moves to the next measurement point for a new evaluation. When all the points in the window width are also within the window height, the measurement is considered settled and the latest point is reported as the measurement value.

If the flat algorithm cannot produce a settled result in a time of ten windows, the settling is considered failed and the data in the most recent time of five windows are averaged to produce a result.

The Settling Average Algorithm

This algorithm averages the data to be settled over a period of time to produce a result. The period is set by:

- the value set in Averaging Time {default: 200.0 ms}.

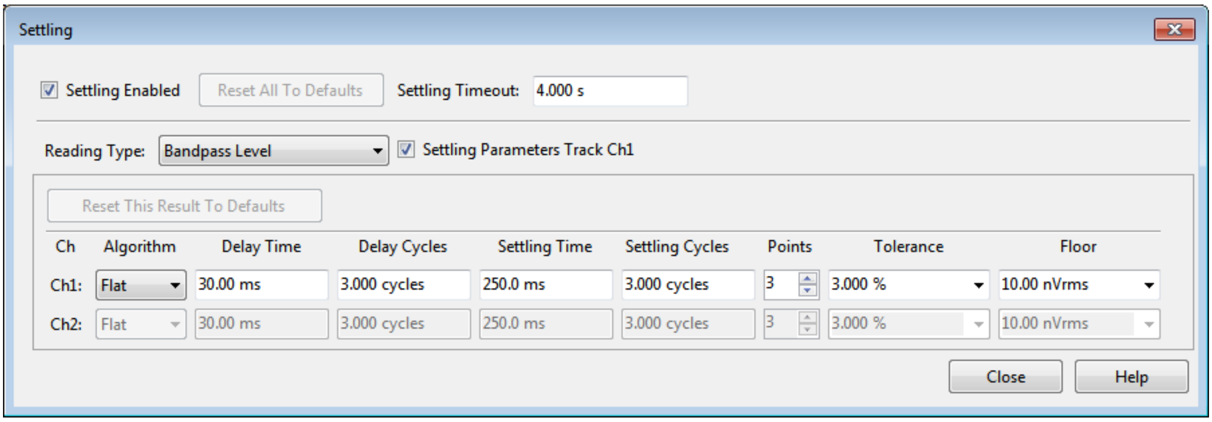
  Minimum averaging time is 0 s.
  Maximum averaging time is 10 s.

  Defaults:
  200.00 ms for all measurements except the SNR (Noise) tab, which is 300.00 ms.

Reset Values to Defaults

Click **Reset Values to Defaults** to restore default settings.

Settling in Bench Mode



See Time and Cycles in Settling dialogs on page 553, and Flat Algorithm window on page 555.

Application of Settling

In **Bench Mode**, **Settling** is applied at each step of a **Sweep**, and can optionally be applied at each reading in the **Recorder**.

The **Monitors/Meters, FFT, Continuous Sweep and Acoustic Response** are always unsettled.

Settling Enabled

This checkbox enables or disables **Settling**.

Reset All to Defaults

This switch resets all settling parameters for all results to the default values.

Settling Timeout

This value sets the maximum time that APx will wait for a settled reading. If the reading is not settled within this time, the reading is marked with a “T” mark (timeout) on the graph. The 6 previous readings are averaged, and the averaged value is used for the failed reading.

Settling Parameters Track Ch1

When **Settling Parameters Track Ch1** is set, all input channels are set to the same settling parameters for each result type. When **Settling Parameters Track Ch1** is unchecked, each channel’s settling parameters can be set individually.

Result Type menu

Select the result that you'd like to examine. Each result type has been populated with default settling parameters appropriate for that result. You can edit the parameters for each result type (and for each channel, if **Settling Parameters Track Ch1** is not set).

Reset to This Result to Defaults

This button resets only the parameters for the selected result type.

Algorithm

Choose **None**, **Flat** or **Average**.

Flat

The **Flat Algorithm** reports a data point when the signal is so stable that the difference between any two consecutive readings, when expressed as a percentage, is:

- less than the percentage value specified in the **Tolerance** field, and
- that this degree of stability is held through the longer of
    - the number of consecutive readings specified in the **Points** field, or
    - the length of time specified in the **Settling Time** field, or
    - the length of time that is the sum of the periods of the number of cycles (at the current measured frequency) specified in the **Cycles** field.

Average

The **Average** settling algorithm takes the mathematical average of the readings acquired in

- the number of consecutive readings specified in the **Points** field, or
- the consecutive readings made in the time specified in the **Settling Time** field, or
- the consecutive readings made in the length of time that is the sum of the periods of the number of cycles (at the current measured frequency) specified in the **Cycles** field, whichever is greater.

The **Average** algorithm is particularly useful when the signal is fundamentally noisy and might never settle within a practical tolerance.

Delay Time

Enter the **Delay Time** here. Delay time defines an initial measurement delay in seconds. The effect of this delay is the same at all frequencies. Minimum delay time is 0 s. Maximum delay time is 10 s. Information gathered during the **Delay Time** interval is ignored.

Settling Time

When the **Algorithm** is **Flat**, enter the **Settling Time** here. When the **Algorithm** is **Average**, enter the **Averaging Time** here.

Along with settling **Cycles**, **Settling Time** defines the width of the **Flat** or **Average** algorithm time window in seconds. If **Cycles** is zero, the width of the time window is the same at all frequencies.

- Minimum **Settling Time** is 0 s.
- Maximum **Settling Time** is 5 s.

Cycles

Enter the number of **Cycles** here. This specifies a length of time that is the sum of the periods of the number of cycles set, at the current measured frequency. This causes settling to be slower, but to have fewer failures at low frequencies.

Cycles, along with **Settling Delay**, defines the width of the **Flat** and **Average** algorithm time window in seconds. If **Cycles** is greater than zero, the time window is wider (settling for a longer time) at low frequencies.

- Minimum **Cycles** is 0.
- Maximum **Cycles** is 500.

Points

Enter the **Points** here. This defines the minimum number of data points to be evaluated for settling.

Tolerance

This field is available when **Algorithm** is **Flat**. This specifies the amount of variability you are willing to accept from test to test.

Floor

This field is available when **Algorithm** is **Flat**. When working at the bottom of a measurement range, it can be difficult to get readings that satisfy settling parameters. Readings below the Floor value are ignored; instead, the Floor value is reported as a result.

93

Results: Meters, Graphs, and Tables

Overview

APx500 measurements have one or more primary results. You can also specify Derived results, User Defined results, and Imported results. Results are viewed in Result Displays.

Primary results

Each measurement has one or more *primary* results associated with it. A primary result is the direct result of a measurement; a primary Level result, for example, is the measured amplitude of the audio signal. Most measurements have several primary results associated with them. Frequency Response, for example, has four: Level, Gain, Relative Level and Deviation.

A new measurement initially presents its complete set of primary results in the Navigator tree and the Selector filmstrip.

A batch measurement (page 160) primary result initially has one *Data Set*, called Measured1. This is the data gathered from measuring the acquisition. For measurements and results that support appending, multiple Data Sets can be appended to primary results and are called Measured2, Measured3 and so on. Meter measurements (page 158) do not support appending measurement data.

Derived results

A measurement can also have one or more user-defined Derived results. A Derived result is a data point or set of data points derived from an existing result (called the source result) by a mathematical computation such as averaging or normalizing. Derived results can be attached to primary results, imported results or existing Derived results.

See Chapter 95 for more information about Derived results.

User Defined results

You can define a new result and add it to a measurement. The data for a Defined result is read from a file on disk at measurement run time, or when the (Refresh) button is clicked. See page 574 for more information about Defined results.

Imported data

Data can be imported into an XY Graph result from a file on disk. The data may have been exported from an APx measurement, or saved from another application, or created as an arbitrary list of values. Any compatible data can be imported into a result to form a new Data Set for that result. Imported Data Sets are called Imported1, Imported2 and so on. Bar Graph results do not support importing data from a file on disk.

See page 573 for more information about importing result data.

Result Displays

Bar Graphs/Meters

Meter measurements (page 158) have only one result parameter per channel. Meters typically show results as Bar Graph displays (also called Meters). Meters also display numerical results on the Bar Graph and in the Graph Data grid. As real-time measurements, meter measurement Bar Graphs and numerical displays are constantly updating.

Some batch measurements have one or more single-parameter results, which are displayed as Bar Graphs. See more about Bar Graphs starting on page 561.

XY Graphs

Most results of batch measurements (page 160) have two parameters per channel (level vs. frequency, for example) and are displayed as XY Graphs. The data being graphed can be viewed as numerical results in the Graph Data grid. See more about XY Graphs starting on page 565.

Other result displays

Some results of a few APx measurements (Metadata Recorder, POLQA, PESQ and Impedance/Theile-Small are examples) do not use either Bar Graphs or XY

Graphs. Displays for such measurements are discussed in the individual measurement chapters.

Lists and tables

The graphic displays for both Bar Graphs and XY Graphs are generated from underlying data points. You can view and export a list (for Meter results) or a table (for XY or other results) containing the data values. See Graph Data on page 573.

Working with results

Navigating results

Results are represented as branch nodes on the Navigator tree display, and as thumbnails in the Selector filmstrip display. The currently selected result is displayed in the Graph panel.

Primary or derived results can be added to or deleted from a measurement, and can be renamed to suit your needs. Multiple instances of a result can be added to a measurement.

Adding, deleting or renaming result

Add, delete or rename a primary or a derived result to a measurement using any of these methods:

- Navigator
  Right-click on a result and choose Add Primary Result, Add Derived Result, Delete Result or Rename.
- Selector toolbar
  Select a result and click Add or Delete in the Selector toolbar.
- Selector filmstrip
  Right-click on a result and choose Add Derived Result, Delete or Rename.
- Selector filmstrip
  Right-click between results and choose Add Primary Result.
- Project menu
  Select a result in the Navigator or the Selector, and from the main Menu bar choose Project > Add Primary Result, Project > Add Derived Result or Project > Delete Result.

In addition, you can add a derived result by right-clicking on the result in the large Graph result display and choosing Add Derived Result, or by clicking the Add Derived Result button in the Graph toolbar.

Selecting a result

To select a result, click on the result branch in the Navigator, or on the result thumbnail in the Selector.

In addition, you can select an existing derived result by first selecting the source result and then choosing Go to Derived from the Project menu or from the context menus available in the Navigator, the Selector or the Graph.

You can also select a source result by first selecting the derived result and then choosing Go to Source from the Project menu or from the context menus available in the Navigator, the Selector or the Graph.

Viewing a result display

View a result display in the Graph panel by selecting the result.

Viewing result details

To view result details, select the result and click Details in the Selector toolbar. This is particularly useful when working with Derived Results, as it reveals the relationships between source and derived results. See page 572.

Moving a result

In the Selector, grab a result thumbnail with the mouse cursor and drag it to a new position in the Selector. The result positions in the Selector determines the relative result locations in a report.

Printing or exporting a result

You can print or export a single result by selecting the result, then clicking Print/Export this Result. Such results are always unsettled. See Settling on page 552.

Including a result in a sequence

Any result that is checked in the Navigator, and whose source measurement is checked, will be included in a sequence.

Including a result in a Sequence report

If Report is checked in the Navigator, any result that is included in a sequence will be included in the sequence report.

You can also run an individual measurement by right-clicking on the measurement in the Navigator and choosing Start Selected Measurement from the context menu. If Report is checked in the Navigator, any result that is checked for that measurement will be included in the measurement report. If Settling is available for the result, it is applied.

“----” (invalid) results

Some configurations or results are invalid, and tables or graphs that display such a result show four dashes “----” followed by the current unit symbol.

Such a result is not a number. It represents an invalid condition for measurement, or no data.

For example, in phase measurements one channel is selected as a reference channel. As such, it has no numerical value, and its result is displayed as "----deg". A reference channel is ignored for pass/fail evaluation and for derived results computations.

In another example, a THD+N Ratio measurement with no input provides nothing for the distortion ratio algorithm to work with. The THD+N Ratio result for a channel with no input is "----%". Such a result will cause a failure when compared against limits.

Similarly, an unlocked digital input has no valid audio result. The result is displayed as "----dBFS".

Derived results attached to "----" results

When a derived result is attached to a result that includes a reference channel (such as a phase result), the reference channel is ignored in the computations.

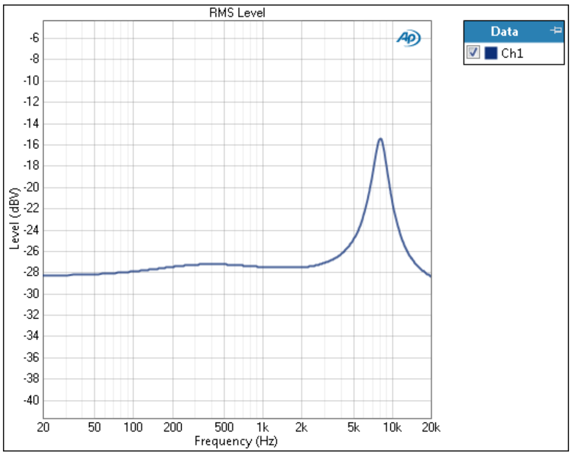
When a derived result is attached to a result that includes one or more channels with "----" results, and one or more of such channels is included in the result specification, the derived result is "----".

Introduction to Result displays

In APx500, measurement results are shown as meter Bar Graphs, as XY graphs or as values in a table or grid. These results can be viewed on-screen, printed or output as data. Results can be also compared to *limits,* so that you can evaluate the performance of your DUT against a performance requirement. This chapter looks at the features of each type of display, the properties you can set for each display, and how to apply limits to your results.

Meter displays show a one-parameter Bar Graph result for each channel. An example is a simple level measurement, as shown here. Channel 1 is measured at 0 dBV.

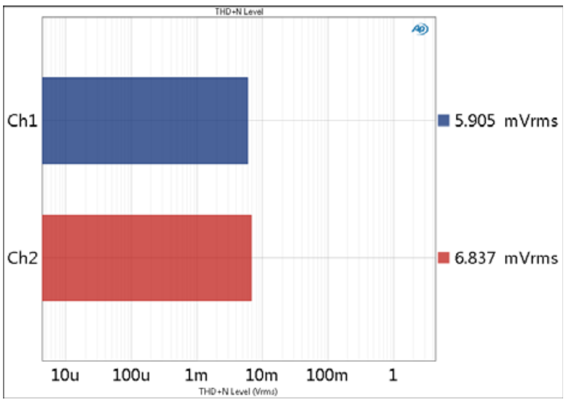
Graphs show the relationship between two parameters for each channel. The example below shows the level of Channel 1 as it varies by frequency. The DUT has a 12 dB peak in the response, centered at about 6.5 kHz.



Tables and grids can show one, two or more parameters per channel, with values for each parameter arranged in a column like a spreadsheet. You'll find grid displays in the Signal Monitor view and in the Reference Levels dBr A and B settings. The data behind each meter bar and Graph display can also be viewed as a table. The table to the right shows part of the data for the frequency response Graph above.

Bar Graph (Meter) Displays

Bar Graph (Meter) displays are used for those measurements that measure only one parameter per channel, providing a clear view of this parameter across many channels. APx500 also uses a vertical Bar Graph display for distortion product measurements, where several parameters are displayed but the view is limited to one channel.



The Bar Graph Panel

The Bar Graph Panel contains the Bar Graph and the Bar Graph toolbar. Below the Bar Graph Panel is the Selector filmstrip, where you can choose which result to view, add results, etc.

Result Name

The name of the result is at the top of the Graph. You can change the result name in the Graph Properties dialog, in the Selector or in the Navigator/Sequencer.

Measurement Timestamp

The current date and time are shown at the top right of the Graph. You can enable or disable the time-stamp in the Project/Sequence Properties dialog.

Zoom

You can zoom in and out using the scroll wheel on your mouse; if you do not have a scroll control, you can hold the **Ctrl** button while dragging the mouse pointer left or right in the Graph area or on the X-axis scale. Click **Set Zoom/Pan to Original** in the tool-bar or context menu to return the Graph to its original size.

Pan

Click anywhere in the Graph or in the X-Axis scale and drag to move the Graph image.

X-Axis

Click on the chevron icon (next to the X-axis title) to open a menu that provides access to **Units** and **Log/ Linear** settings, **Autoscale On/Off** and the **Graph Properties** dialog.

Bar Graph toolbar

The Bar Graph (Meter) toolbar extends across the top of the Graph Panel. The commands in the toolbar are also available in the context menu invoked by right-clicking in the Graph area.

Save Graph image to disk

Click this button to save the meter Bar Graph to disk as an image file. A file browser window opens to enable you to name the file and browse to a folder. Supported file formats are

- .bmp
- .jpg
- .emf
- .png
- .pdf

This image can then be imported into another application. If the data on screen is unsettled, the image that is saved is also unsettled.

Copy Graph image to clipboard

Click this button to copy the meter Bar Graph to the Windows clipboard as a bitmap image. This image can then be pasted into another application. If the data on screen is unsettled, the image that is copied is also unsettled.

Print/Export Results for this view

Click this button to print or export the results of the current measurement. If limits are set, the results are compared to the limits.

Zoom in

Click this button and then drag the mouse pointer across the Graph to define an area. When you release the mouse button, the display will zoom to that area.

Zoom out to original

Click this button to return a zoomed or panned display to its original magnification and location.

Fit view to data

This control autoscales the graph to the current data, at the moment the **Fit view to data** button is clicked.

Show Graph data

Click this button to open a data view window that displays the meter data. From this window, you can copy the data to the Windows clipboard as text, or export the data as a Microsoft Excel spreadsheet file or CSV (comma separated value) text file.

Draw/Edit Limits

This button opens a menu where you can select one of the two following choices:

 ***Draw Limits***

Click this button to open the Draw Limits dialog.

 ***Edit Limits***

Click this button to open the Edit Limits dialog. See page 577 for information about limits.

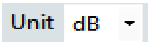
Edit Graph Properties

Click this button to open the Edit Graph Properties dialog. See page 563 for information about editing meter Bar Graph properties.

(Pan)

To pan (slide the X axis of the Graph without changing the current magnification), click and drag anywhere in the Graph, or click and drag in the X-axis scale.

Units

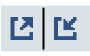


A measurement produces a result whose value can be expressed within a certain family of measurement units appropriate to that parameter, so the list of unit choices available on the meter bar display is determined by the type of measurement.

Choose a measurement unit for your display from the Units drop-down list. Some units are absolute; others calculate values in reference to values set in reference fields in APx500.

See Chapter 98 for more information about units of measurement.

Undock and Dock



At the right of the Bar Graph toolbar is the Undock button. This places the meters in a new window (undocks them) that can be moved and resized. When a meter display is undocked, clicking the Dock button will return it to its fixed location in the workspace.

Additional features on the context menu

Right-click in the Graph area to open a context menu. The commands on the toolbar are also available here, with these additional commands:

Edit annotations

You can add text annotations to the Graph display. These annotations are included with the Graph in the Report and when copying or saving the Graph image.

Right-click on the Graph, and select choose **Edit Annotations** from the context menu. This opens the Annotation Toolbox.

To add an annotation, click the button in the Annotation Toolbox, and drag the mouse cursor to create a box on the Graph. Using the Toolbar, choose font size, style and color. Use the **Fill Color** tool to color the box background. Type in the box to enter text for the annotation.

To edit an existing annotation, open the Annotation Toolbox and select the annotation box. Change the font style, size or color in the toolbar. Select and edit the type in the box.

To move the text annotation box, click on the box frame and drag the box to a new position.

To resize the text annotation box, click on a box corner and drag to resize the box.

To delete a text annotation, open the Annotation Toolbox, select the annotation box to be deleted, and click the **Delete Annotation** button.

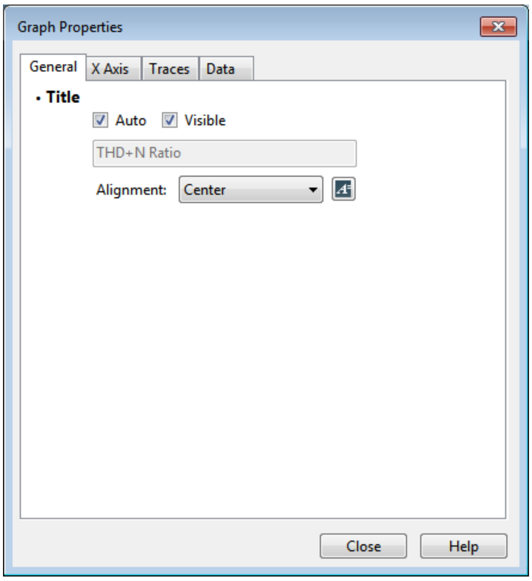
Autoscale

Autoscale is a shortcut that sets the Graph Properties **Range** checkbox to **Auto**. Right-click on the Graph, then select **Autoscale** > **X Axis** from the context menu.

Meter Bars: Edit Graph Properties

The Graph Properties dialog allows you to edit the meter Bar Graph title, X-Axis title and range.

General tab



Title

For most results, when **Auto** is checked the default name for the measurement result (RMS Level, for example) is entered into the Title field. For a Defined Result, when **Auto** is checked the Title string from the referenced file is entered.

When **Auto** is unchecked, you can enter any title into the Title field.

Visible

When **Visible** is checked the title is applied to the Graph.

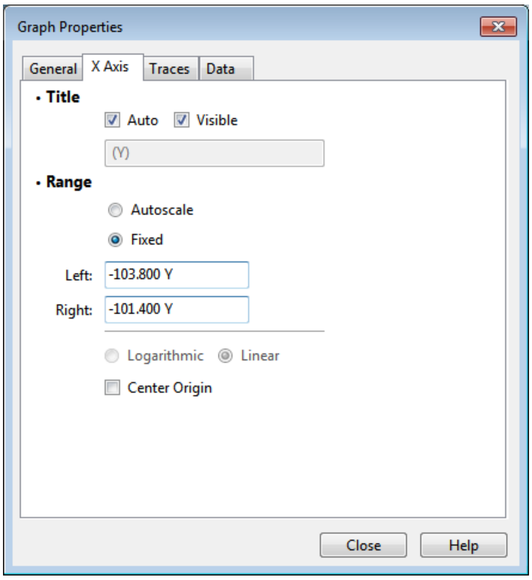
Alignment

Select Left, Center or Right title alignment.

Font...

The **Font...** button opens the Windows font browser. You can modify the title font family, style, size, color and so on from this dialog. Default font is MS Sans Serif 8 point.

X-Axis tab



Title

When Auto is not checked, you can enter a title for the X-Axis in the Title field.

Auto

When Auto is checked, the X-Axis label shows the parameter and the units (Level [dBV], for example).

Visible

When Visible is checked the X-Axis title is applied to the Graph.

Range

Range sets the values at the left and right edges of the Graph. Enter values in the Left and Right range fields. You can also select units from the range field drop-down menus.

Autoscale

Autoscale sets the range around the current signal characteristics.

Logarithmic

When Logarithmic is checked, the X-Axis scale follows a logarithmic rule.

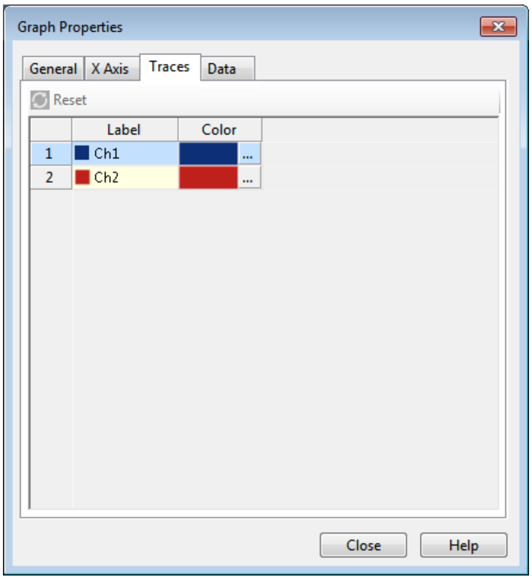
Linear

When Linear is checked, the X-Axis scale follows a linear rule.

Center Origin

For a Bar Graph Defined Result, you can specify whether or not the Graph origin (0 point) is centered.

Traces tab



Open the **Graph Properties** dialog and go to the **Traces** tab. By default, the settings are derived from the global setting made in the **Labels** dialog, accessed from **Signal Path Setup**, and from the global settings made in the **Colors** tab, accessed from the **Project/ Sequence Properties** dialog. In the **Traces** tab you can edit the local settings for each Bar Graph channel, overriding the global settings.

Label

Enter a new **Label** for the channel in this field. This setting is local and will only affect this result.

Color

Click on the button to open a color picker. Choose a new **Color** for this channel. This setting is local and will only affect this result.

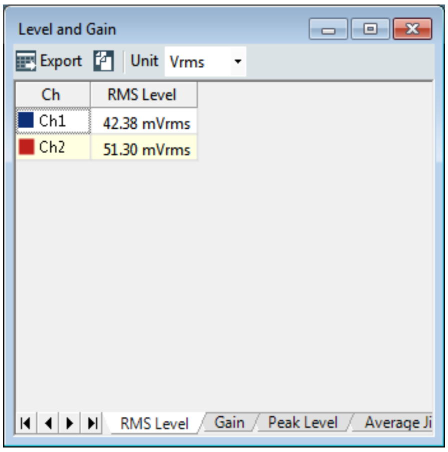
Alternatively, you can right click in a displayed value in a meter result (Bar Graph display) to open a context menu and edit the **Label** and **Color** for that channel.

For more information about setting global labels, see page 54. For information about global colors and line styles, see page 32.

Data Tab

The Data Tab is used for User Defined Results. See page 574.

Graph data for meters



The data window for meters shows the values for each meter bar.

Tabbed Pages

For measurements that have more than one result view, tabs at the bottom of the data grid give you access to the Graph data for the other results.

Units

For each page, you can select units that differ from the initial Graph units.

Export

Click the Export button to save the data for that page as a Microsoft Excel spreadsheet file (*.xls) or as a comma-separated-value (*.csv) text file.

Copy to clipboard

Click the Copy button to copy the data to the Windows clipboard as tab-delimited text. This data can be pasted into other applications such as Microsoft Word or MS Excel.

XY Graphs

APx500 graphs display two (and occasionally three) parameters of a signal measurement on an XY grid.

In a frequency domain display, these parameters are typically level versus frequency, with level on the Y axis (vertical, left), and frequency on the X axis (horizontal, bottom). In a time domain display, these parameters are typically level versus time, with level on the Y axis and time on the X axis. Other relationships can be graphed: phase angle versus frequency, DUT output level versus generator level (linearity) and so on.

Dual axis graphs

An example of a three parameter Graph is the Rub and Buzz result in the Acoustic Response measurement. Crest Factor is on the Y axis (vertical, left), Peak Ratio is on the second Y axis (vertical, right) and frequency is on the X axis (horizontal, bottom). Read more about Dual Axis graphs starting on page 569.

The XY Graph Panel

The XY Graph Panel contains the XY Graph and the XY Graph toolbar. Below the XY Graph Panel is the Selector filmstrip, where you can choose which result to view, add results, etc.

Result Name

The name of the result is at the top of the Graph. You can change the result name in the Graph Properties dialog, in the Selector or in the Navigator/Sequencer.

Measurement Timestamp

The current date and time are shown at the top right of the Graph. You can enable or disable the timestamp in the Project/Sequence Properties dialog.

Zoom

You can zoom in and out by hovering the mouse pointer in the Graph area and using the scroll wheel on your mouse; if you do not have a scroll wheel, you can hold the **Ctrl** button while dragging the mouse pointer in the Graph area or on an X- or Y-axis scale, creating a marquee area. When you release the mouse, the Graph will zoom to the marquee area. Click **Set Zoom/Pan to Original** in the toolbar or context menu to return the Graph to its original size.

Pan

Click anywhere in the Graph or in an X- or Y-axis scale. The mouse pointer will take the shape of a hand; drag to move the Graph image. Click **Set Zoom/Pan to Original** in the toolbar or context menu to return the Graph to its original position.

X-axis and Y-axis

By default, there is only one active Y-axis (the left axis) on an XY Graph. You can add a second set of data that share the same X-axis but are plotted to a second Y-axis, on the right. Click the chevron icon on the right side of the Graph to define the right Y-axis. Read more about Graphs with two Y-axes starting on page 569.

For an active axis, click the chevron icon (next to an axis title) to open a menu that provides access to **Units** and **Log/Linear** settings, **Autoscale On/Off** and the **Graph Properties** dialog.

Graph toolbar

Note: The following commands are also available by right-clicking in the Graph display and choosing the command from the right-click menu.

Save Graph image to disk

Click this button to save the Graph to disk as an image file. A file browser window opens to enable you to name the file and brows to a folder. Supported file formats are

- .bmp
- .jpg
- .emf
- .png
- .pdf

This image can then be imported into another application.

Copy Graph image to clipboard

Click this button to copy the Graph to the Windows clipboard as a bitmap image. This image can then be pasted into another application.

Print/Export Results for this View

Click this button to print or export the results of the current measurement. If limits are set, the results are compared to the limits.

Zoom in

The Zoom control gives you the ability to enlarge an area of the Graph. Click the Zoom button, and the mouse pointer becomes crosshairs (a cross).

Horizontal and vertical zoom

Drag the crosshairs across the Graph to define a marquee area for enlargement. Release the mouse button and the Graph will zoom to the area defined.

You can also zoom by holding the **Ctrl** button down and dragging a marquee in the Graph.

Horizontal zoom only

Hold the **Ctrl** button while you drag the crosshairs, and the marquee will define an area for horizontal enlargement.

You can also zoom horizontally by hovering the mouse pointer over the X-axis scale and using the scroll control on the mouse; if you do not have a scroll control, you can hold the **Ctrl** button while dragging the mouse pointer left or right on the X-axis scale.

Vertical zoom only

Hold the **Shift** button while you drag the crosshairs, and the marquee will define an area for vertical enlargement.

You can also zoom vertically by hovering the mouse pointer over a Y-axis scale and using the scroll control on the mouse; if you do not have a scroll control, you can hold the **Ctrl** button while dragging the mouse pointer up or down a Y-axis scale.

Zoom out to original

Click this button to return a zoomed display to the original magnification and panning position.

You can also return to the original by double-clicking in the Graph area.

Fit view to data

This control autoscales the graph to the current data, at the moment the **Fit view to data** button is clicked.

(Pan)

To pan (slide the X and Y axes of the Graph without changing the current magnification), click and drag anywhere in the Graph. To pan horizontally, click and drag in the X-axis scale. To pan vertically, click and drag in a Y-axis scale.

Show/Hide Cursors

Click this button to show or hide the Graph cursors and the associated cursor legend panel. See Cursors on page 568 for detailed information.

Show Graph data

Click this button to open a data view window that displays the Graph data. From this window, you can copy the data to the Windows clipboard as text. You can also export the underlying Graph result data as an

Excel spreadsheet (*.xls), a comma separated value text file (*.csv) or a Matlab file (*.mat).

Draw/Edit Limits

This button opens a menu where you can select one of the two following choices:

Draw Limits

Click this button to open the Draw Limits dialog.

Edit Limits

Click this button to open the Edit Limits dialog. See page 577 for information about limits.

Edit Graph Properties

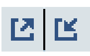
Click this button to open the Edit Graph Properties dialog.

Units

A measurement produces a result whose value can be expressed within a certain family of measurement units appropriate to that parameter, so the list of unit choices available on the meter bar display is determined by the type of measurement.

Choose a measurement unit for your display from the Units drop-down list. Some units are absolute; others calculate values in reference to values set in reference fields in APx500.

Undock / Dock



At the right end of the Graph toolbar is the **Undock** button. This sets the current Graph in a new window that can be moved and resized. In the new window, click the **Dock** button to return the Graph to its docked location.

Additional features

Edit Annotations

You can add text annotations to the Graph display. These annotations are included with the Graph in the Report and when copying or saving the Graph image.

Right-click on the Graph, and select choose **Edit Annotations** from the context menu. This opens the Annotation Toolbox.

To add an annotation, click the button in the Annotation Toolbox, and drag the mouse cursor to create a box on the Graph. Using the Toolbar, choose font size, style and color. Use the **Fill Color** tool to color the box background. Type in the box to enter text for the annotation.

To edit an existing annotation, open the Annotation Toolbox and select the annotation box. Change the font style, size or color in the toolbar. Select and edit the type in the box.

To move the text annotation box, click on the box frame and drag the box to a new position.

To resize the text annotation box, click on a box corner and drag to resize the box.

To delete a text annotation, open the Annotation Toolbox, select the annotation box to be deleted, and click the **Delete Annotation** button.

Autoscale

For the X-axis or the Y-axis, or both, autoscale causes the Graph scale to automatically expand or contract to fit the range of the measurement. These selections have the same functions as the **Auto** checkboxes in the Graph Properties dialog.

Graph Legend

The Graph Legend controls which data are visible on the Graph, and how the data are displayed.

Channel identification conventions

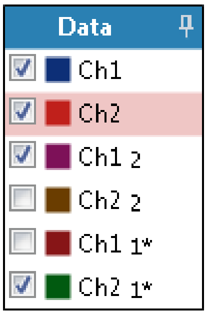
For a single acquisition, the Legend will show an entry for each channel. By default, the channel names take the form of Ch1, Ch2 and so on. If channels have been renamed, the new name is displayed.

If data is appended or imported, the Legend will show entries for each channel of each data set. Appended data are identified by a number in subscript, corresponding to the “Measured n” number automatically assigned to the appended data; imported data are similarly identified by a number corresponding to the “Imported n” number. Imported data channels are also marked with an asterisk (*) to differentiate them from appended measurements.

If the Data Set selector is set to **All Data**, the Legend will show all channels of all data sets.

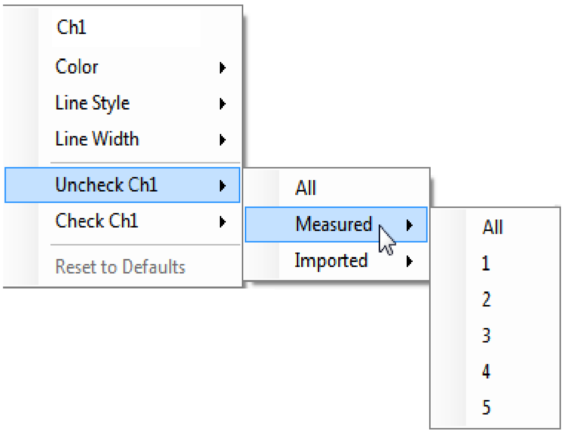
Using the Legend to set trace visibility

If you click on a Legend entry, the entry will be highlighted in the Legend. Using **Ctrl+Click**, you can highlight more than one entry. On the Graph, the traces that correspond to highlighted Legend entries remain full visible, while any other traces are displayed with less intensity (dimmed).



Each Legend entry has a checkbox. If the checkbox is checked, the trace will be visible on the Graph. If the checkbox is not checked, the trace will not be visible on the Graph.

To quickly set visibility on or off for a channel across all measured or imported data sets, right-click on a Legend entry and choose **Check Ch n** or **Uncheck Ch n**, and the choose **All**, or **Measured**, or **Imported**.



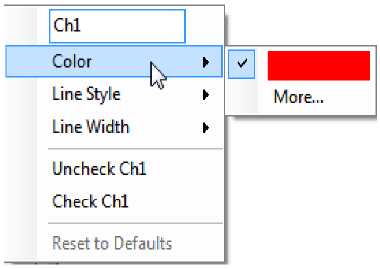
Managing the Legend

With large channel counts and many data sets, the Legend list can grow quite long. When the Legend list grows too long for the display area, scroll arrows will appear at the top and bottom of the Legend display to allow you to move through the Legend entries.

If a channel name is too long for the space allocated to the Legend display, you can see the full channel name in the tool tip that appears when you hover the mouse cursor over the Legend entry. Alternatively, you can drag the bar separating the Graph from the Legend to the left. This will reduce the Graph display area and enlarge the Legend display area.

Using the pin control at the top of the Legend display, you can set the display to show both checked and unchecked entries, or to only show checked entries. When the pin is set to show only checked (visible) entries, hovering over the Legend display with the mouse cursor will temporarily show the entire list.

Using the Legend to set local trace colors and styles



You can set local (measurement result) trace names, colors and styles using the Legend display. To set global (project-wide) names, colors and styles, see "Setting Global Graph Colors and Styles" on page 32.

To set local (measurement result) trace names, colors and styles, right click on a Legend entry to open this context menu.

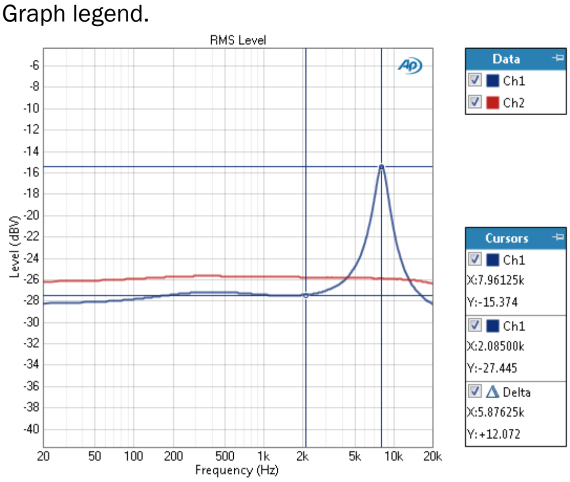
You can change the channel name by typing in the name field. Click on **Color**, **Line Style** or **Line Width** to set the trace style.

Cursors

You can add one or two cursors to an XY Graph display to point to and evaluate points of interest on the traces. An optional Delta result will display the difference between the two cursors scaled in the units chosen for the Graph.

Click the **Show/Hide Cursors** button on the Graph toolbar or on the context menu that appears when you right-click on the Graph. This is a "toggle" switch that will show or hide the cursors and the cursor legend.

When the cursors are displayed, the cursor legend also appears to the right of the Graph, under the Graph legend.



Cursors: attached and unattached

Cursors can be independent of the Graph traces (unattached), or can be attached to any available trace. By default, both cursors are attached to the Channel 1 trace, as shown. When a cursor is unattached, you can move it to any point on the Graph by grabbing either of the cursor graticules or their point of intersection and dragging the cursor. When a cursor is attached to a trace, you can only grab the vertical graticule and drag the cursor in a horizontal direction.

Cursor Legend

Managing the Legend

If a channel name is too long for the space allocated to the Legend display, you can see the full channel name in the tool tip that appears when you hover the mouse pointer over the Legend entry. Alternatively, you can drag the bar separating the Graph from the Legend to the left. This will reduce the Graph display area and enlarge the Legend display area.

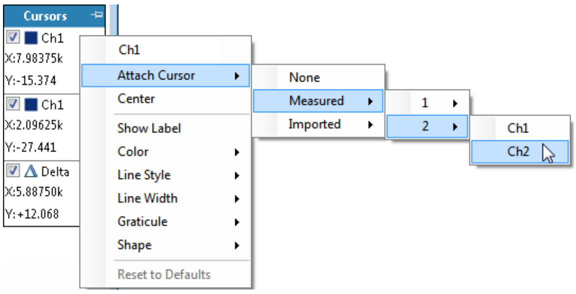
Using the pin control at the top of the Legend display, you can set the display to show both checked and unchecked entries, or to only show checked entries. When the pin is set to show only checked (visible) entries, hovering over the Legend display with the mouse pointer will temporarily show the entire list.

Setting cursor visibility

Each Legend entry has a checkbox. If the checkbox is checked, the cursor (or Delta result) will be visible on the Graph and included in the report. If the checkbox is not checked, the cursor (or Delta result) will not be visible on the Graph nor be included in the report.

Opening the cursor properties context menu

In the cursor legend, right-click in one of the cursor areas. A flyout context menu will open, allowing you to view and set the cursor's properties.



Labeling a cursor

From the cursor's properties context menu, click on the cursor's label (the first item in the menu). The label will become an editable field where you can modify the cursor name. By default, the cursor labels take the form of Cursor1(Ch1) and so on.

Attaching a cursor to a trace

From the cursor's properties context menu, go to Attach Cursor > and select the Graph trace you would like to attach the cursor to, or None. If data is appended or imported, the menu will show entries for each channel of each data set. Acquired and appended data are grouped in "Measured" submenu, and imported data are similarly grouped in the "Imported" submenu.

Centering a cursor

Sometimes you may zoom or pan to a portion of the Graph that does not include the cursor. From the cursor's properties context menu, choose Center to bring the cursor to the center of the current Graph view.

Setting cursor colors and styles

Show label

From the cursor's properties context menu, choose Show Label. This will toggle the cursor label visibility on the Graph display.

Color, Line Style, Line Width

From the cursor's properties context menu, you can set cursor names, colors and styles.

Graticule

The lines which define the cursor's position are the graticules. From the cursor's properties context menu, you can set a cursor's horizontal or vertical graticules to None, Short or Long.

Shape

By default, the intersection of a cursor's graticules is marked with a small circle.

From the cursor's properties context menu, you can choose other shapes to indicate the intersection of the graticules.

Dual Axis Graphs

You can add a second Y-axis to APx XY Graphs, enabling simultaneous viewing of two sets of data that share a common X-axis.

- View the same data with two different units: Vrms and dBV, for example.
- View two different primary results: Level and THD+N, for example.
- View a primary result and a Derived Result on the same graph.

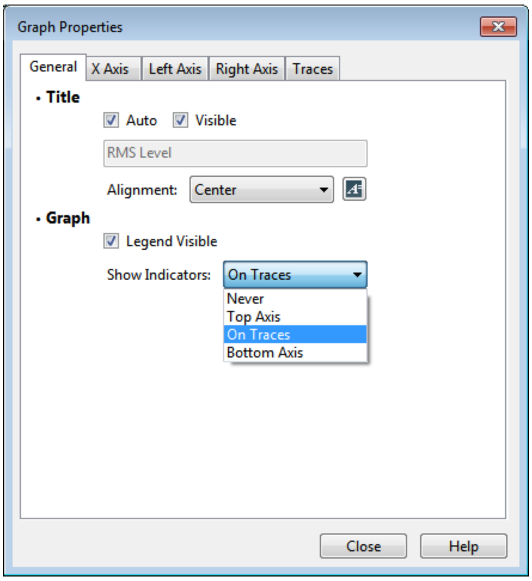
The Acoustic Response / Loudspeaker Test Rub and Buzz result uses this feature to plot the crest factor against the left axis and the peak ratio against the right axis, providing a graphically compelling Rub and Buzz indicator.

To add a second axis to an XY Graph, click the chevron icon (hover the mouse pointer on the right side of the Graph to see the icon) and select **Define Axis**. Choose the result you'd like to plot against the right axis.

XY Graphs: Edit Graph Properties

The Graph Properties dialog allows you to edit the Graph title and X- and Y-Axis titles and ranges.

General tab



Title

The Graph title always carries the name of the measurement (RMS Level, for example). To this you may add more descriptive text in the Title field.

Alignment

Select **Left**, **Center** or **Right** title alignment.

Font...

The **Font...** button opens the Windows font browser. You can modify the title font family, style, size, color and so on from this dialog. Default font is MS Sans Serif 8 point.

Visible

When **Title Visible** is checked the title is applied to the Graph.

Legend Visible

The legend is the boxed display containing channel number checkboxes on the right side of the Graph. When Legend Visible is checked the legend is applied to the Graph.

Show Indicators (some XY Graphs)

Some XY Graphs (see Stepped Frequency Sweep, for example) can show indicators at data points on the trace. This feature is most useful with graphs with relatively few data points.

In graphs with many data points, the markers and labels will overlap and obscure each other. Check **Show Indicators** to add markers.

Select

- **Never**
- **Top Axis**
- **On Traces** (the default)
- **Bottom Axis**

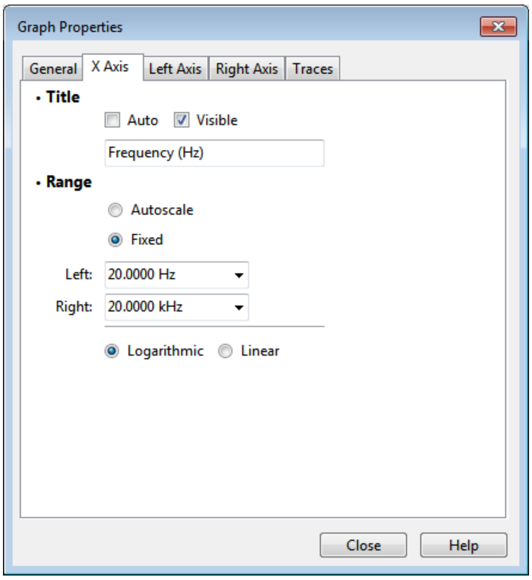
Show Markers (PESQ Averaged and POLQA Averaged only)

PESQ Averaged and POLQA Averaged graphs can show markers with X and Y values for each point plotted.

Select

- **X Labels**
  File number is shown on marker
- **Y Labels**
  MOS for file is shown on marker

X-Axis tab



Title

When **Auto** is not checked, you can enter a title for the **X-Axis** in the **Title** field.

Auto

When **Auto** is checked, the X-Axis label shows the parameter and the units, such as Generator Level (dBV).

Visible

When **Visible** is checked the X-Axis title is applied to the Graph.

Range

Range sets the values at the left and right edge of the Graph. Enter values in the **Left** and **Right** range fields. You can also select units from the range field drop-down menus.

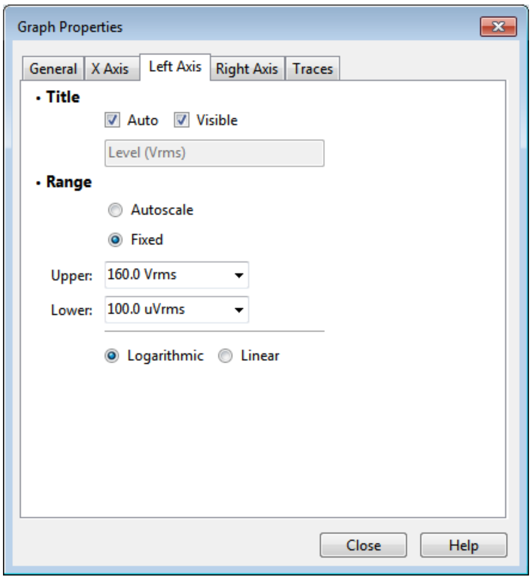
Auto

Auto sets the range around the current signal characteristics.

Logarithmic

When **Logarithmic** is checked, the X-Axis scale follows a logarithmic rule. When unchecked, the X-Axis scale follows a linear rule. The logarithmic scale is not available for certain units.

Left Axis tab



Dual axis graphs will have a Left Axis and a Right Axis tab.

Title

When **Auto** is not checked, you can enter a title for the Y-Axis in the Title field.

Auto

When **Auto** is checked, the Y-Axis label shows the parameter and the units (Level [dBV], for example).

Visible

When **Visible** is checked the Y-Axis title is applied to the Graph.

Range

Range sets the values at the top and bottom edges of the Graph. Enter values in the Upper and Lower range fields. You can also select units from the range field drop-down menus.

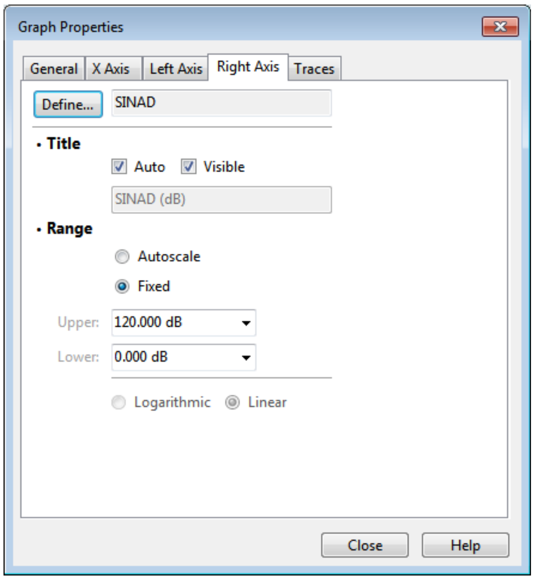
Auto

Auto sets the Y-Axis range around the current signal characteristics.

Logarithmic

When **Logarithmic** is checked, the Y-Axis scale follows a logarithmic rule. When unchecked, the Y-Axis scale follows a linear rule.

Right Axis tab



Dual axis graphs will have a Left Axis and a Right Axis tab.

Title

When **Auto** is not checked, you can enter a title for the Y-Axis in the Title field.

Auto

When **Auto** is checked, the Y-Axis label shows the parameter and the units (Level [dBV], for example).

Visible

When **Visible** is checked the Y-Axis title is applied to the Graph.

Range

Range sets the values at the top and bottom edges of the Graph. Enter values in the Upper and Lower range fields. You can also select units from the range field drop-down menus.

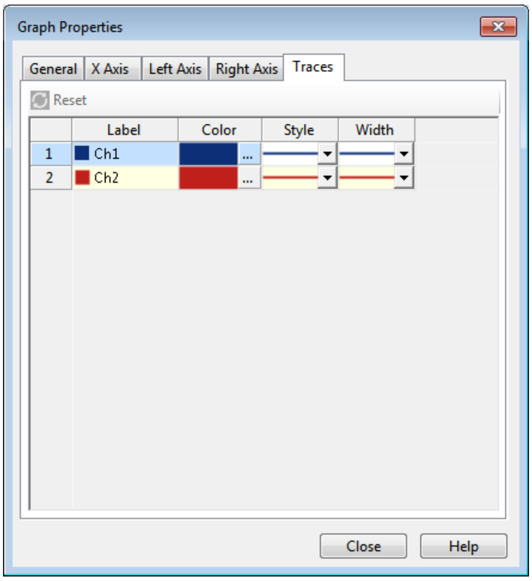
Auto

Auto sets the Y-Axis range around the current signal characteristics.

Logarithmic

When **Logarithmic** is checked, the Y-Axis scale follows a logarithmic rule. When unchecked, the Y-Axis scale follows a linear rule.

Traces tab



Open the **Graph Properties** dialog and go to the **Traces** tab. By default, the settings are derived from the global setting made in the **Labels** dialog, accessed from **Signal Path Setup**, or from the global settings made in the **Colors** tab, accessed from the **Project/ Sequence Properties** dialog. In the **Traces** tab you can edit the local settings for each trace, overriding the global settings.

Label

Enter a new **Label** for the channel in this field. This setting is local and will only affect this result.

Color

Click on the button to open a color picker. Choose a new **Color** for this channel. This setting is local and will only affect this result.

Style

Select a trace **Style** from the drop down menu. This setting is local and will only affect this result. This setting is local and will only affect this result.

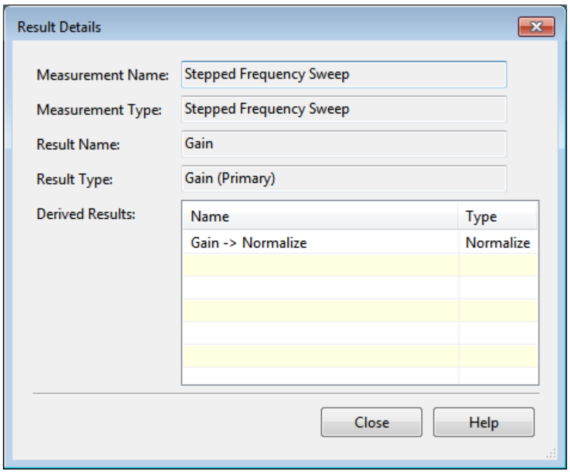
Width

Select a trace **Width** from the drop down menu.

Alternatively, you can right click in a Legend entry to open a context menu to edit the **Label**, **Color, Line Style** or **Line Width** for that channel.

For more information about setting global labels, see page 54. For information about global colors and line styles, see page 32.

The Result Details dialog

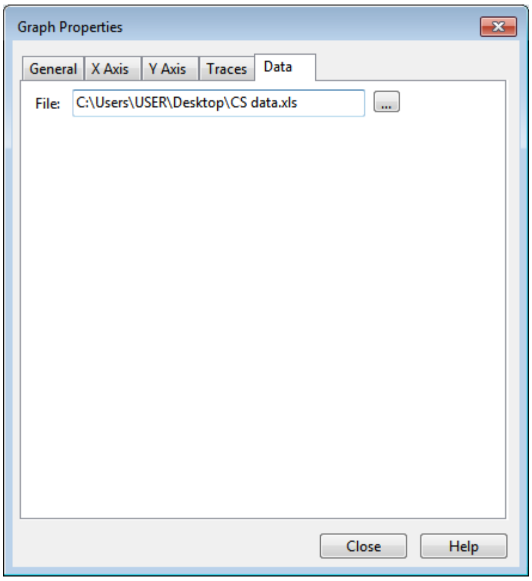


To view result details, select the result and click **Details** in the **Selector** toolbar. This is particularly useful when working with **Derived Results**, as it reveals the relationships between source and derived results. Read more about **Derived Results** in Chapter 95.

The **Result Details** dialog is informational only. It provides a display of a result's key characteristics, including:

- **Measurement Name**
  This field displays the name of the measurement providing the data for the result(s). The measurement name can be in the Navigator or Selector panes.
- **Measurement Type**
  This field displays the type of measurement, which cannot be changed.
- **Result Name**
  This field displays the name of the current result. The result name can be edited in the Navigator or Selector panes.
- **Result Type**
  This field displays the type of result, which cannot be changed.
- **Source**
  This field only appears if the current result is a derived result. It shows the source result for the current result.
- **Derived Results**
  If there are other results derived from the current result, they are shown here.

Data tab (Defined Results only)



This tab only appears for a Defined result.

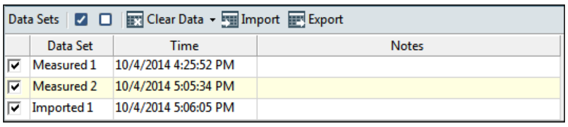
File

Enter the path and filename for the data file to be referenced by the Defined result, or use the Browser button to browse to the file. Read more about Defined Results on page 574.

Managing Data Sets

A **Data Set** is the set of results measured from an acquisition, or imported from a data file.

Data Sets are managed in the **Data Sets** panel, located below the **Selector** in the APx workspace.

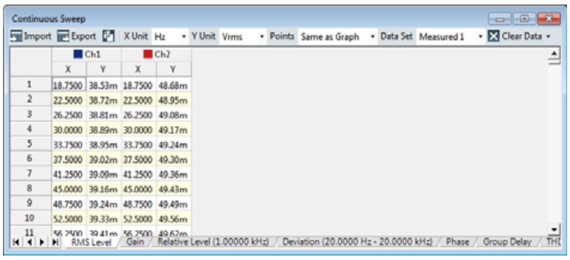


Data Sets panel

The **Data Sets** panel displays all the **Measured** and **Imported** data sets in a grid format.The first column provides visibility checkboxes; the second column displays the name of the data set; the third column displays the date and time of the acquisition or importation; the fourth column is available for notes. The notes are saved with the project.

Click **Hide** to hide the **Data Sets** panel. When hidden, click **Show Data Sets** to restore the panel.

Graph data



The Graph data window shows the values for each data point in the Graph. There are X and Y values for each channel.

Tabbed Pages

For measurements that have more than one view, tabs at the bottom of the data grid give you access to Graph data pages for the other views.

Import 

You can import Graph data from a file on disk to be displayed in the current Graph. The imported data is managed as a new data set, and is designated by the next available Imported Data Set number (n+1).

Data can be imported from a Microsoft Excel spreadsheet, a CSV (comma separated value) text file, or from Audio Precision *.adx or *.atsx legacy data files. If your data comes from some other source, data within the file must be arranged in the same format as a data file exported from APx500.

Measurement views represented in the data file that are compatible with the current measurement are shown in the first set of fields. If more than one result is available for a measurement view, you can select the result to import.

Measurement views in the data file that are not compatible with the current measurement are shown in the second set of fields, and will not be imported.

Export 

Click the **Export** button to save the data for that page as a Microsoft Excel spreadsheet file (*.xls) or as a comma-separated-value (*.csv) text file. For Excel sheets, each tabbed page is saved as a new worksheet page within the file. For CSV files, tabbed pages are listed in sequential order within the file.

Copy to clipboard 

Click the Copy button to copy the data to the Windows clipboard as tab-delimited text. This data can be pasted into other applications such as Microsoft Word or MS Excel.

Data arrangement in the file

The data arrangement is discussed in terms of a spreadsheet grid. A CSV file, if used, should correspond to this arrangement. See the illustration on the previous page.

	A	B	[Note]
1	Title	{empty}	[This can be any text string, or empty]
2	Trace Name	{empty}	[This can be any text string, or empty. It can be centered in merged cells A-B]
3	X-axis legend	Y-axis legend	[This can be any text string, or empty]
4	X-axis unit name	Y-axis unit name	[This can be any text string, or empty]
5	{X-data1}	{Y-data1}	[This can be any value, or NaN]
6	{X-data2}	{Y-data2	[This can be any value, or NaN]
7	{X-data3}	{Y-data3}	[This can be any value, or NaN]
8	{X-data4}	{Y-data4}	[This can be any value, or NaN]
9	{X-data5}	{Y-data5}	[This can be any value, or NaN]

- In APx500, NaN (Not a Number) represents an undefined value resulting from an invalid operation.
- As it is imported, the data is checked to verify that the values in column A monotonically increase. The data are and sorted if necessary.
- Cells in rows 1-4 can be empty
- When an empty row ≥ 5 occurs, it is interpreted as the end of the data.
- When an XY result has one empty X or Y cell in a row ≥ 5, it is interpreted as an error.
- For more Traces 2-16, if required, repeat this pattern in columns C-D, E-F, and so on.

X Units / Y Units

For each page, you can select units that differ from the initial Graph units.

Points

The default number of data points is Same as Graph, the same as the number of Graph data points. This number can be large for continuous sweeps. You can reduce the number of points shown in the data grid by choosing a smaller number from the Points dropdown list, but you will lose resolution in the data shown or exported. The Points control is not available in the stepped sweep data grid dialogs, because Stepped Sweeps normally have a manageable number of points.

Note that Same as Graph reports only the points shown in the current Graph view. Zooming or panning the graph will affect the number of points reported.

Data

For appended or imported data, you can select the data set you would like to view or export.

User Defined Results

Overview

You can define a new result and add it to a measurement. The data for a Defined Result is read from a file on disk at measurement run time, or when the (Refresh) button is clicked).

A Defined Result can be of the XY Graph type, or of the Bar Graph (Meter) type. It can contain any data, related or unrelated to the measurement to which it is attached.

Any units for the data can be arbitrarily assigned, and do not need to correspond to units used in APx500.

Creating and Using a Defined Result

You can attach a Defined Result to most measurements. Use one of these three methods to create a Defined Result.

1. Right-click on a measurement node in the Navigator, and select Define New Result. Choose (X, Y) or Bar.
2. Or, select a measurement and from the Project menu, choose Define New Result. Choose (X, Y) or Bar.
3. Or, click Add in the Selector menu bar. Choose Defined Result, then choose (X, Y) or Bar.

The Defined Result will be added to the measurement, and the Graph Properties dialog will open to the Data tab. In the File field, enter the path and filename for the data file to be referenced by the Defined Result, or us the Browser button to browse to the file.

The Graph Title, X and Y legend and units used in the Defined Result are imported from the file. Graph Properties (scale, alignment, title, trace color, etc.) are set in APx500 in the Graph Properties dialog.

See Data File Formats for Defined Results on the next page.

Application

You may define any result you find useful and add it to a measurement. You may assign any unit names to the data, including unit names not used in APx500.

Use Case 1

Goal: You wish to make an acquisition in Measurement Recorder, and to measure the audio with a fast RMS measurement, evaluating the waveform at a reading rate of 1000 times per second.

1. Save the acquisition from Measurement Recorder

2. Make a fast RMS measurement with an external utility program that outputs its results in a CSV file.

3. Define a New Result attached to Measurement Recorder.

4. Reference that Defined Result to the CSV file.

Use Case 2

Goal: You wish to aggregate results from 3 different APx500 2-channel measurements into one Graph.

1. Run each measurement.

2. Export the data from the desired result from each measurement to CSV files.

3. Aggregate the data in an external program, such as Microsoft Excel.

4. Format the 3 sets of stereo data into a 6-channel format.

5. Define a New Result attached to a measurement of your choice.

6. Reference that Defined Result to the CSV file.

Data File Formats for Defined Results

File types

- Microsoft Excel spreadsheet (*.xls). Only the first sheet in a workbook will be read.

  or

- Comma-separated value (CSV) text file (*.txt).

	A	B
1	Level	
2	Trace 1	
3	X	Y
4	Hz	Vrms
5	20	0.1002349
6	20.447743	0.1002349
7	20.90551	0.1001984
8	21.373525	0.1001984
9	21.852017	0.1001984

```
Level,,,
Trace 1,,,
X,Y,,
Hz,Vrms,,
20,0.100234876,,
20.44774311,0.100234876,,
20.90550991,0.100198422,,
21.3735248,0.100198422,,
21.85201722,0.100198422,,
```

Excel Spreadsheet **CSV**

See a guide for proper arrangement of data in the file on the next page.

94

Limits

Using Limits: an overview

Limits are preset values to which measurement values are compared during testing. In APx500, limit values or limit curves can be attached to any measurement. As you make the measurement, you can see whether or not the measurement values exceed your limits.

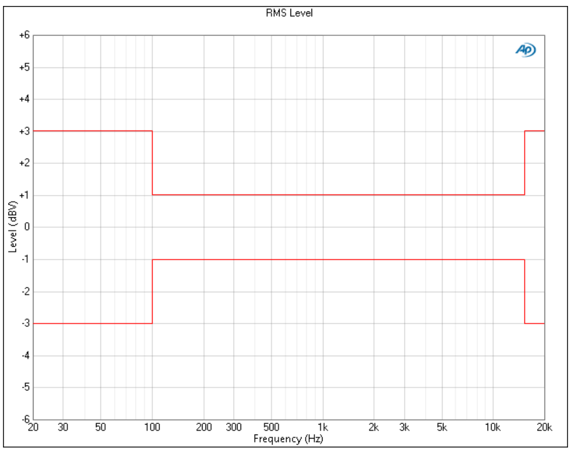
In an APx500 automated sequence, measurements containing values that exceed limit values are marked as “Failed” in the Navigator panel and on the Report. Measurements without limit failures are marked as “Passed.”

Go to page 494 for more information about using pass/fail limits in a sequence.

Setting limits directly

You can set limits directly to a result graph. For example, perhaps it is important that your DUT perform to an industry standard that requires that frequency response be ±3 dB below 100 Hz and above 15 kHz, and ±1 dB at points between.

You can simply draw or enter these limits in an APx500 measurement that provides frequency response results, and run the test or sequence. The limits might appear on your graph as shown:



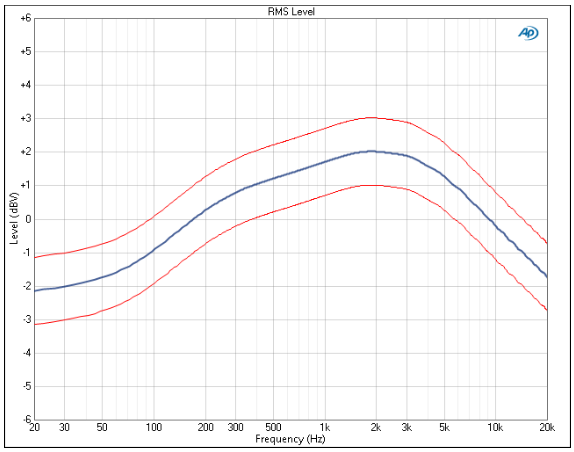
Creating limits from measurement data

Alternatively, you may have a “golden unit,” a DUT that performs very well, and that you would like to use as a standard for your Pass / Fail testing. For an X/Y graph, you can create limits from the measured response of this unit.

For example, let’s use the frequency response of a golden unit.

- Measure the response.
- Create limits from the data. In the Edit Limits dialog, click **Copy Graph Data**.
- Now click **Offset Limit**. Enter the offset value that represents the tolerance you will allow for this test. In the figure below offset was set to ±10 mVrms.

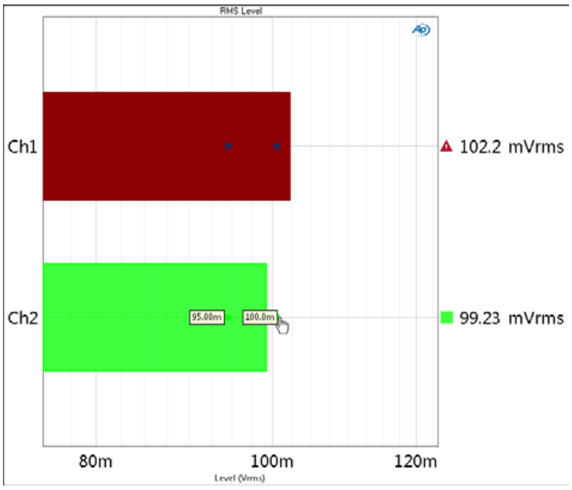
The offset limits follow the DUT response as shown. The red lines are the upper and lower limits; the blue line in the middle is the golden unit data.



Creating offset limits from graph data is discussed in detail on page 581.

Limits for single value results

You can apply upper and lower limits to a APx500 bar graph display. These are used as Pass / Fail criteria when comparing a device to specifications or to a "golden unit."



Pass / Fail markers

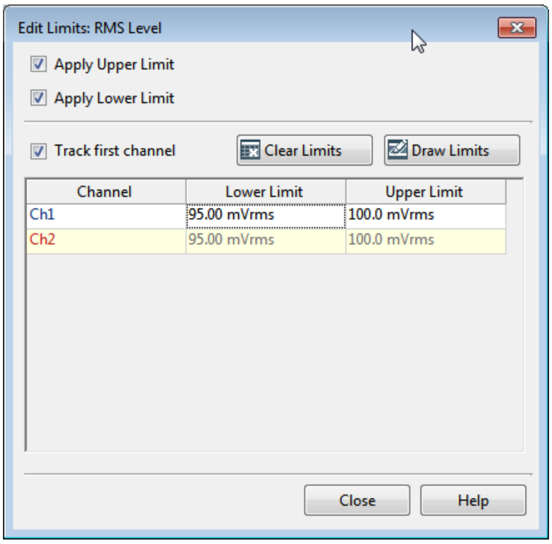
In a single value measurement, limits are shown as arrows on the meter bar display. A measurement that exceeds a limit changes the color of the meter bar to dark red. (The bar above had been blue before exceeding the limit.)

Measurements that exceed limits are marked as Failed ⚠ .

In Sequence Mode, both the sequence report and the Navigator measurement are marked as Passed or Failed when the sequence is run.

Edit limits for single value results

The Edit Limits dialog provides capabilities to create, apply and edit limits.



Track First Channel

If Track First Channel is checked, the limit values for channel 1 are copied to the other channels, and the limit values for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels.

Creating New Limits

New limits are created by entering limit values in the Edit Limits dialog.

Applying limits

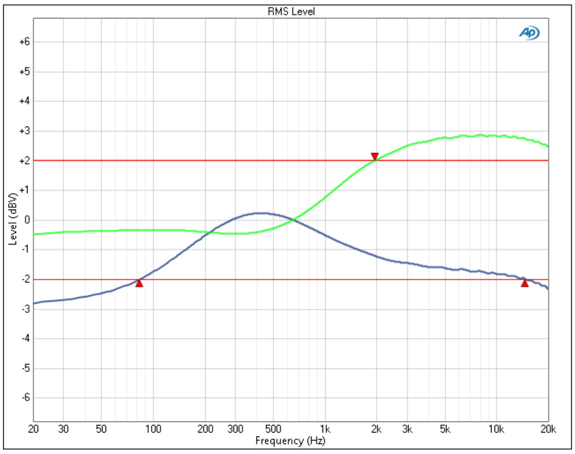
Limit data that is not applied remains attached to a measurement view but is not visible on the meter bars or active in a sequence.

To apply limits check **Apply Upper Limit** or **Apply Lower Limit**. When limits are applied, the limit markers appear on the meter bars and are used during a sequence to enable Pass / Fail marking.

When set, limits are indicated by arrows on the meter bar display. Limits can be moved by grabbing a limit arrow using the mouse pointer and dragging it to a new position.

Limits for XY results

You can apply upper and lower limits to APx500 XY graphs. These are used as Pass / Fail criteria when comparing a device to specifications or to a "golden unit."



Pass / Fail markers

In a sweep measurement, a limit appears as a red line drawn on the graph. A red triangle is drawn at the intersections where a measurement trace crosses a limit line.

When a report is generated, measurements that fall entirely between limits are marked as Passed ✅. Measurements that exceed limits are marked as Failed ⚠.

When a sequence is run, both the sequence report and the Navigator measurement are marked as Passed or Failed.

Dual axis XY graphs

APx graphs can plot two results on the same display, using a left and right Y-axis. The following topics point out the differences in drawing, editing and applying limits when working with one and two Y-axis displays.

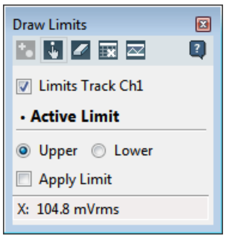
Creating and Editing Limits for sweep measurements

Limits for sweep measurements can be set and viewed on the on graph using the Draw Limits tool. You can also set and view limits in the Limits Grid in the Edit Limits dialog.

Draw Limits for XY graphs (one Y-axis)

Select **Draw Limits** from the Edit Limits menu on the graph toolbar, or right-click on the graph and select the **Draw Limits** item from the menu.

Drawing an upper limit for channel 1



By default, Draw Limits is set to channel 1, upper limit.

- When no limit has been created, the **Draw Limits**  tool is selected.
  Drag the mouse pointer across the graph to create an upper limit curve. After this first entry, the **Add Points** tool is selected.
- Use the **Add Points**  tool to extend the curve. Click the mouse pointer on the graph to add a point.
- Use the **Move Points**  tool to drag an existing point to a new location.
- Use the **Erase Points**  tool to remove a point.
- Click **Delete Limit**  to entirely remove a limit curve.

Drawing a lower limit for channel 1

- Select **Lower** to create a lower limit curve for channel 1.
- Repeat the steps described above.

Track First Channel

With **Track First Channel** set, all channels will have these limit curves attached. To draw different limit curves (or have no limit curves) for other channels, deselect **Track First Channel**.

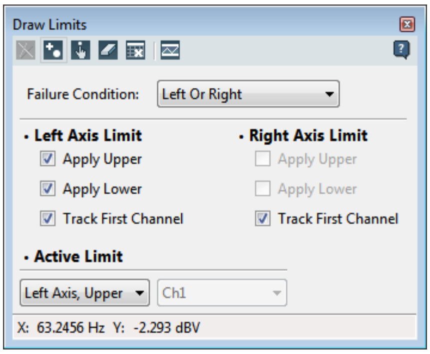
Apply Limits

Limits remain attached to a result until deleted. However, for a limit to be visible and active, it must be applied. Check the appropriate **Apply Limit** boxes.

Editing limits

Click the **Edit Limits** button  to go to the Edit Limits dialog box, where you can import, export, offset and edit limits by keyboard entry.

Draw Limits for Dual Axis graphs



For graphs that have sets of result data, displayed on the left and right axes, there are more settings on the Draw Limits dialog.

Apply Limit

You can choose to apply the upper left, lower left, upper right or lower right limit.

Current Limit

You can choose to draw the upper left, lower left, upper right or lower right limit.

Failure Condition

You can specify whether a limit failure is flagged when a point on either the left or right axis exceeds a limit, or when points on both left and right axes exceed a limit, for the same X value.

Edit Limits for XY results

The Edit Limits dialog provides capabilities to apply, edit, import, export and offset limits. You can also copy current graph data to create new limits. If you have used the Draw Limits tool before opening Edit Limits, the values for the limit points you have drawn will be shown in the limits grid.

The Limits Grid

Most of the Edit Limits dialog is taken up by the Limits Grid. The Limits Grid displays the X and Y values for each current limit point for all channels. Upper and lower limits are available on tabbed grid pages.

Track First Channel

If **Track First Channel** is checked, the X and Y limit values for channel 1 are copied to the other channels, and the limit values for channels greater than 1 cannot be edited. Any changes made to channel 1 are reproduced in the other channels.

Creating New Limits

New limits can be created in three ways:

- Drawing limits with the Draw Limits tool.
- Importing limits from an external limits file.
- Copying the current graph data into the Limits Grid.

Once the Limits Grid has data, you can edit the X or Y values of any point by entering a new value in the corresponding cell. You can also add or delete limit points using the Add Point or Delete Points tools.

Add Point

Each horizontal line in the Limit Grid represents one limit point. Select a point by placing the cursor on a line of data on the Limit Grid. Click Add Point to add a new point below the selected point.

Delete Points

Each horizontal line in the Limit Grid represents one limit point.

Select one or more points by placing the cursor on one or more lines of data on the Limit Grid. Click Delete Points to remove the selected points.

Applying limits

Limit data that is not applied remains attached to a result but is not visible on the graph or active in a sequence.

To apply limits to a graph and sequence, check either or both Upper or Lower checkboxes in Apply Limits. When limits are applied, the limit curves appear on the graph and are used during a sequence to enable Pass / Fail marking.

Importing and Exporting limits

Limit data can be imported from external files. Supported file formats include Microsoft Excel spreadsheets (*.xls) and comma-delimited text files (*.csv), as well as the AP2700 and ATS-2 formats (*.adx) and (*.atsx). Upper limit data and lower limit data are imported from separate files; use the tabs at the bottom to select the corresponding target page. When limit data are imported, all current limit data are first cleared from memory.

When importing data from a file exported from the graph Data Grid, only data corresponding to the current measurement view (THD Level, for example) will be imported. If there is no such data in the file, no limit data will be imported.

Limit data can also be exported to external files in two file formats, *.xls and *.csv. Upper limit data and lower limit data are saved in separate files; select the appropriate limit source page using the tabs at the bottom of the grid. These files can be edited in Microsoft Excel or a text editor and imported back into APx500 if desired.

Copying graph data into the Limits Grid

A simple way to create limits is to copy the current graph data into the Limits Grid, and then to move the data to higher and lower values by entering new values or by using the Offset Limits dialog.

Click Copy Graph Data to bring the current graph data into the Limits Grid. When graph data is copied into the Limits Grid, all current limit data is first cleared from memory.

Data points for graph limits

Graph data usually consists of many more points of data than is useful to define a limit curve. A typical stepped sweep may have 31 points; a continuous sweep may have thousands of points. When copying graph data into the Limits Grid, the number of points copied is reduced to 10, 20, 30, 40 or 50.

Want more points?

Here's a hint if you want very high-resolution limits. Export the Graph Data to an Excel or CSV text file, then import that file into the Limit Editor. Now use Offset Limits to create upper and lower limits that follow the data exactly.

Edit Limits for Dual Axis graphs

Apply Limit

You can choose to apply the upper left, lower left, upper right or lower right limit.

Current Limit

You can choose to draw the upper left, lower left, upper right or lower right limit by selecting a grid page using the tabs at the bottom of the grid.

Failure Condition

You can specify whether a limit failure is flagged when a point on either the left or right axis exceeds a limit, or when points on both left and right axes exceed a limit, for the same X value.

Offset Limits

The Offset Limits dialog provides an easy way to raise or lower an entire limit curve (changing all the Y values) by a specific amount. This is useful if your current limit curve is of the correct shape, but requires a shift to either tighten or loosen your pass / fail criteria.

Using Offset Limits with copied graph data

When used with Copy Graph Data, Offset Limits is also a quick way to create limits that mimic measured data, such as that from a "golden unit."

- In the Edit Limits dialog, select the upper limit grid page with the Selected Limit Upper button. Click Copy Graph Data. From the drop-down list, choose the number of points of data to be copied.
- Choose Offset Limit.
- Enter a value (perhaps +3 Vrms) in the Offset field for Channel 1. Click OK.
- Select the lower limit page. Repeat the previous steps for the lower limit, but enter –3 Vrms. Click OK.

Upper limits are typically offset by positive voltage values; lower limits by negative voltage values. By following the steps in this example, you will have created a pair of limits about 6 Vrms apart surrounding the existing Channel 1 data.

Track First Channel

If the Track First Channel checkbox in the Offset Limits dialog is checked, all channels will be offset by the value entered in the Channel 1 offset field.

Note: When a limit offset is applied, the values in the Limits Grid are permanently changed to incorporate the offset. You may continue to edit the Limit values by changing values in the Limits Grid, by using the Draw Limits tool or by applying new limit offsets.

Centering new data within limits

In some applications, the general conformance of a trace to the shape of a set of limits is of importance, with absolute level less important or irrelevant.

You can automatically center new data within existing limits for the Relative Level results of these measurements:

- Acoustic Response (Chapter 21)
- Bandpass Frequency Sweep (Chapter 22)
- Continuous Sweep (Chapter 28)
- Frequency Response (Chapter 43)
- Multitone Analyzer (Chapter 60)
- Stepped Frequency Sweep (Chapter 76)

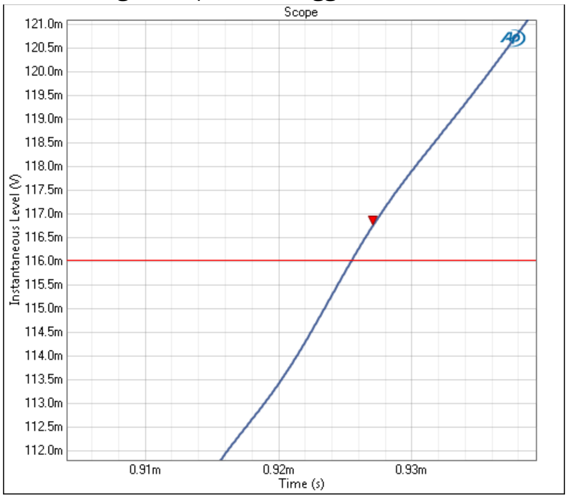
You can automatically center new data within existing limits for the Linearity results of these measurements:

- Bandpass Level Sweep (Chapter 24)
- Stepped Level Sweep (Chapter 77)

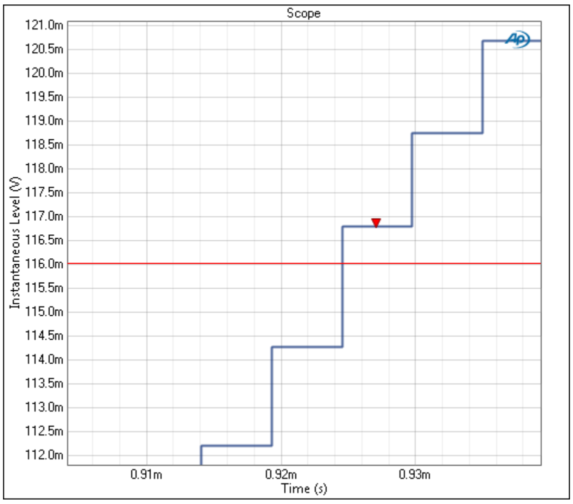
Refer to the measurement chapters for operational details.

Interpolation and limit failure markers.

Limit failure markers (small orange triangles) are placed at the first data point that exceeds a limit. When interpolation is on and the graph is zoomed in to high magnification, the limit failure markers may appear to be "floating" in the graph, not at the intersection of the limit line and the interpolated data trace. Although this appears to be an error, it is actually correct behavior. Turn interpolation OFF, and you will see that the marker is at the first (real) data point that exceeds the limit. There is no data at the intersection, although interpolation suggests there is.



Limit failure marker "floating" with interpolation OFF.



Limit failure marker on data with interpolation ON.

95

Derived Results

Introduction

Derived results accomplish many of the same functions as Computes in other Audio Precision software.

Most measurement results, whether primary, imported, or derived, can have a derived result attached. A result that has a derived result attached is called a *source* result. The derived result is computed from the source result, using one of a set of mathematical functions such as arithmetic mean (average), normalize, invert, etc. See page 559 and Chapter 93 for more information about primary or imported results.

The availability of derived result functions depends upon the type of source result the derived result is attached to. A meter result such as **Level and Gain: RMS Level**, for example, does not support **Normalize** or **Invert**, but an XY result such as **Frequency Response: Level** does.

Not all results support a derived result. Additionally, derived results do not support importation of graph (result) data (see page 559).

Working with Derived Results

Navigating, adding, deleting and renaming derived results is the same as working with primary results. See more information about working with results on page 560.

Of particular interest when working with derived results is the Result Details dialog, which reveals the relationships between source and derived results. See page 572.

Go to page 592 for several illustrated examples that show the software user interface for attaching and configuring derived results.

Derived Results settings and specifications

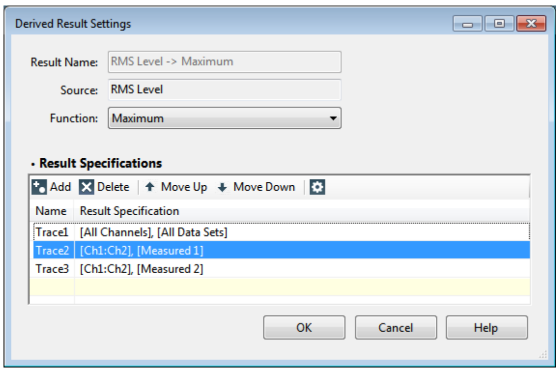
Each derived result has settings to define the computations that are applied to the source result. These can be very simple and straightforward, requiring the entry of just one value in a setting field.

Or, the settings can involve multiple sets of specifications, each of which define different computations across a range of source data.

Single value (meter) source results only support single value derived results. XY source results, however, can have derived results attached that may have an XY output, or a single value output, depending upon the derived result function and the intention of the user.

Compare (when attached to an XY result) and Statistics group derived results require more a detailed result specification, defining the channels and/or data sets to be considered. These are edited in Result Specification dialogs. Click the **Derived Result Settings...** button on the Result Settings bar to open this dialog.

Result settings with multiple specifications



As you can see from this example, several result specifications (up to 16) can be associated with certain derived results. Each of these specifications will generate a bar or a trace in the derived result display, labeled with the name shown in the grid.

Result Name

This is the name of the current derived result. By default, derived results are named in this pattern: Source result -> derived result. You can rename the result.

Source

This field displays is the source result (or results) upon which the current derived result is acting.

Function

For Statistics group derived results, this field displays the currently applied Statistics group function. This function can be changed at any time, without redefining result specifications.

Add

Initially, each derived result has one result specification. Double-click or select **Properties** to open the Result Specification dialog for editing.

The default specification is named **Bar1** or **Trace1**, and occupies the first line of the grid. Click **Add** to add more result specifications, which will take the default names **Bar2, Bar3** or **Trace2, Trace3** and so on. You can rename each specification.

You can also add a result specification from the context menu available by right-clicking in the grid area and selecting **Add Result Specification...**

Delete

When there is more than one result specification in the grid, you can remove a result specification by highlighting it in the grid and clicking **Delete**, or by right-clicking and selecting **Delete Result Specification** from the context menu.

Move Up / Move Down

When there is more than one result specification in the grid, you can move a result specification by highlighting it in the grid and clicking **Move Up** or **Move Down**, or by right-clicking and selecting **Move Up** or **Move Down** from the context menu.

Properties

You can view and edit the result specification properties by double-clicking the specification line in the grid, or by highlighting the line and clicking the **Properties** button, or by right-clicking and selecting **Properties** from the context menu.

Functions available for derived results

- Smooth (page 584)
- Normalize (page 585)
- Invert (page 586)
- Offset (page 586)
- Compare (page 586)
- Specify Data Points (page 587)
- Harmonic Sum (page 588)

Min/Max Statistics group:

- Minimum (page 590)
- Maximum (page 590)
- Geometric Mean (page 590)
- Arithmetic Mean (page 591)
- RMS (root mean square) (page 591)
- Standard Deviation (page 591)
- Max Difference from Geometric Mean (page 591)
- Max Difference from Arithmetic Mean (page 591)
- Data Distribution (page 591).

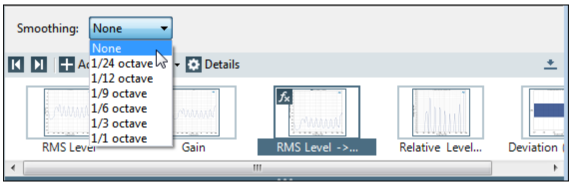
Smooth (frequency domain)

Smooth is a derived results function that is primarily used to smooth frequency-domain XY results. Frequency-domain smoothing is a common technique in loudspeaker response measurement, useful in revealing trends by reducing anomalies in the response curve for viewing. We recommend using smoothing in the Acoustic Response measurement.

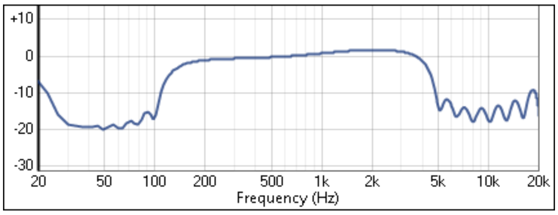
The smoothing algorithm for frequency-domain XY results effectively passes the data through a constant-Q bandpass filter. The bandwidth of this filter is selected in the Smoothing field. The APx500 implementation uses a hybrid FFT bin averaging and interpolation technique to achieve smooth results even at very low bin densities.

When a Smooth derived result is selected, the Smoothing field will be available in the Results Setting Bar. Note that the Deviation result, which does not support derived results, has a dedicated smoothing function in the Acoustic Response measurement.

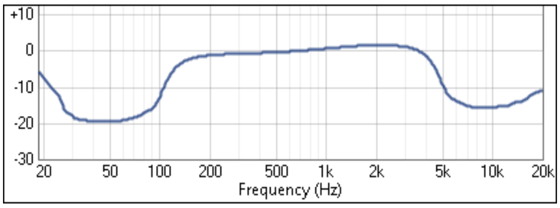
Select the desired value from the dropdown list.



The first trace displays the unsmoothed level results.



The trace below has had 1/3 octave smoothing applied.



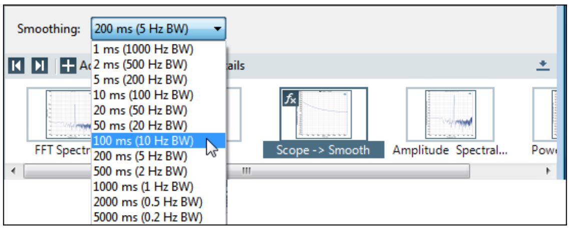
Smooth (time domain)

As mentioned above, Smooth is a derived results function that is primarily used to smooth frequency-domain XY results. However, it can also be attached to time-domain XY results.

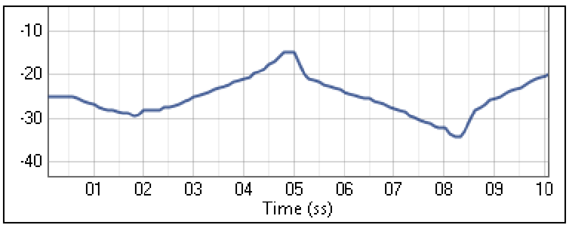
The smoothing algorithm for time-domain XY results effectively passes the data through a single-pole low-pass filter. The time constant and bandwidth of this filter is selected in the Smoothing field.

When a Smooth derived result attached to a time-domain result is selected, the Smoothing field will be available in the Results Setting Bar.

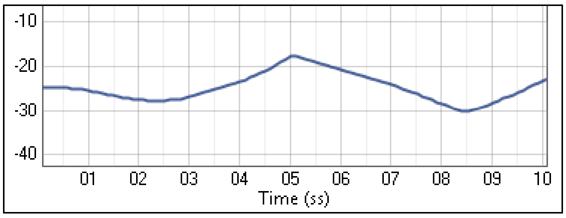
Select the desired value from the dropdown list.



The first trace displays an unsmoothed level result from the Measurement Recorder, showing a DUT whose gain varies over time.



The trace below has had smoothing applied.



Note that the Measurement Recorder has a maximum reading rate of 20 points per second, which provides only 200 data points in a 10 second acquisition. With so few points, the traces do not appear as smooth as a typical frequency domain trace, which may have thousands of data points in an acquisition.

Normalize

Normalize can only be attached to an XY result.

Normalize is a derived results function that shifts an XY trace so that the data point at the X-axis reference is normalized (set to the unit value of the base unit of the unit domain). The X-axis for XY graphs is usually *frequency*, but for some results is *time*.

Normalization is accomplished by first converting the result units to the base unit for the unit domain, then dividing the value of each data point by the value of the data point at the selected X-axis reference.

For Vrms, the unit value (normalization value) is

- 1 Vrms.

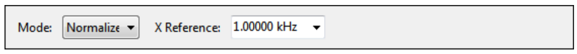
For Vrms, the unit value (normalization value) is

- 1 W.

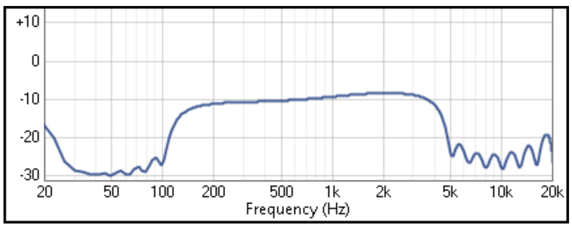
For decibel unit, the unit value (normalization value) is

- 0 dB.

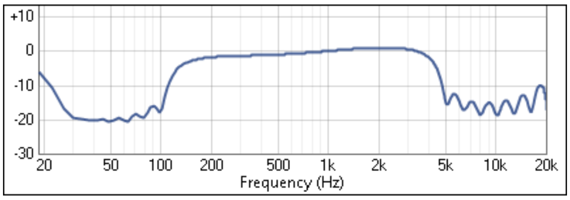
Enter the X-axis reference point in the **X Reference** field.



The first trace shown below is the Level result, before normalization.



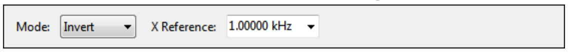
The second trace below has been normalized at 1 kHz.



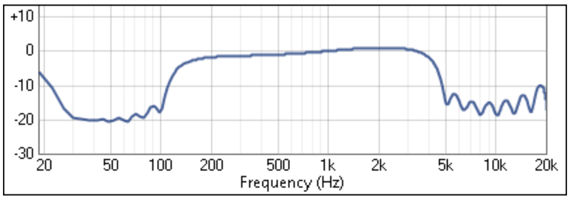
Invert

Invert can only be attached to an XY result.

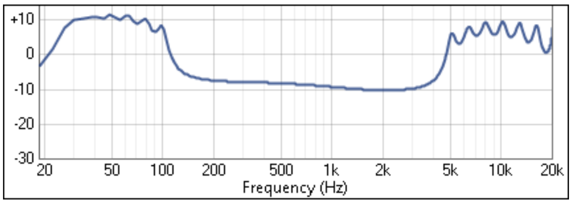
Invert is a derived results function that “flips” an XY trace around a user-defined pivot point. Set the pivot point by choosing an X-axis reference, entered in the **X Reference** field on the Result Settings bar.



The first trace shown below has been normalized at 1 kHz.



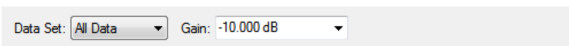
The second trace below is the normalized result above, inverted with an X Reference (pivot point) of 1 kHz.



Offset

Offset is a derived results function that shifts the value(s) of the result data by a specified amount. Offset can be attached to either single value (meter) results or XY source results.

One of two algorithms is used, depending upon the unit domain.



Offset applied to audio level unit domains

For audio levels and ratios of audio levels, Offset is accomplished by application of a **Gain Value**, multiplying or dividing the source result value. The **Gain Value** is specified in the Result Settings bar.

For example, specify a +10 dB offset for an existing result of 1 Vrms by entering 10 dB in the **Gain Value** field. 1 Vrms = 0 dBV, and derived result will be +10 dBV, which is equal to 3.162 Vrms.

Offset applied to other unit domains

For any DC level, Frequency or Phase result, for Acquired Waveform results, or for the Signal Analyzer Scope result, offset is accomplished by adding or subtracting an offset value to the source result. The **Offset Value** is specified in the Result Settings bar.

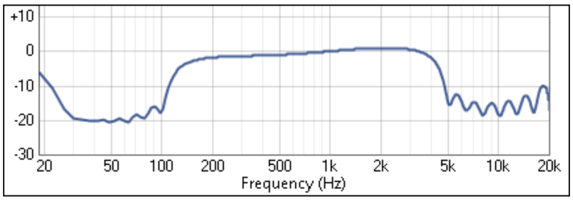
...when attached to meter results

When attached to a meter source result, Offset returns the current value for each channel, shifted by the Offset value and displayed as single value (meter) bar graphs.

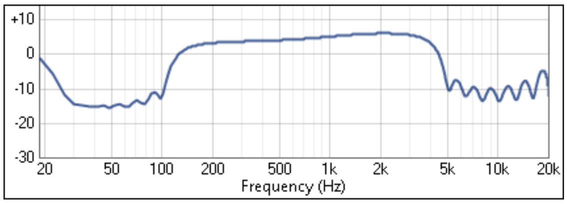
...when attached to XY results

When attached to an XY source result, Offset returns the value for each data point, shifted by the Offset value and displayed as XY graph traces.

The first trace shown below has been normalized at 1 kHz.



The second trace below is the normalized result, offset by +5 dB.

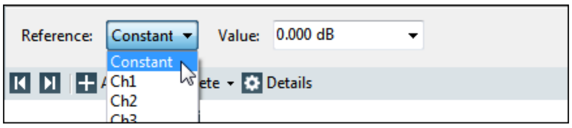


Compare (Ratio or Level)

Compare is a derived results function that compares data to a constant or to the data in another channel or data set. Compare can be attached to either single value (meter) results or XY source results.

For source results in absolute units, such as volts or watts, Compare computes the difference between the values. For source results in relative units, such a decibels (dB), Compare computes the ratio between the values.

attached to a meter result

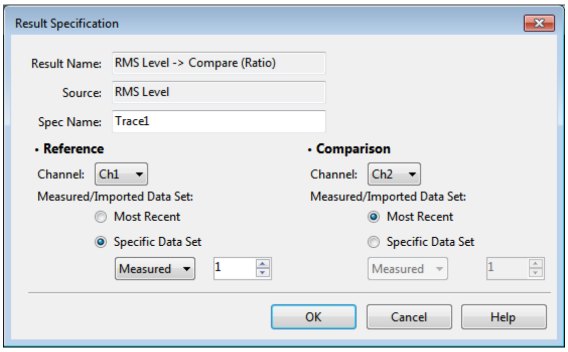


When attached to a meter source result, Compare finds the current value for each channel, compared to the value of the reference. The reference can be set to a constant value, or to the current value in any channel. The comparison values are single value (meter) results, displayed as bar graphs.

attached to an XY result

When attached to an XY source result, Compare finds the current Y value for each data point, compared to the Y value of the reference data point. The comparison data points can be selected from any channel and from any data set; similarly, the reference data points can be selected from any channel and from any data set. The comparisons are displayed as XY graph traces.

These selections are made in the Result Specification.



Typical Result Specification dialog for Compare when attached to an XY source.

In this application, Compare supports up to 16 specifications, managed in a Derived Result Settings dialog (page 583).

A note about channel count in derived results...

In a derived result setting you can select channel numbers beyond the channel count available via the physical interface connectors on your APx analyzer.

This is because all APx500 systems support up to 16 input channels of audio data, when the input configuration is set to File.

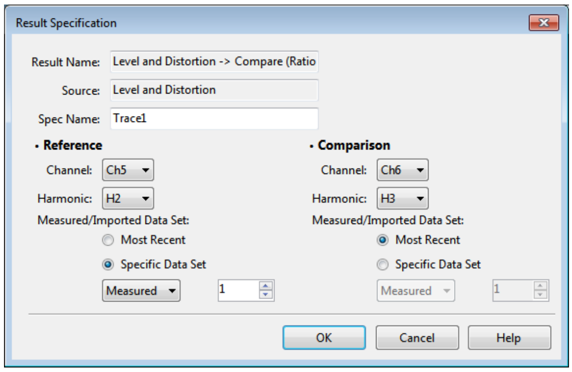
Be sure to select channel numbers that are relevant to your test setup.

A note about specifying non-existent data sets in derived results...

You can specify data sets that have not yet been acquired or imported. This supports use in automated sequences, where the derived result can be defined before the sequence has completed all of its append activities.

Special Compare Specifications for Acoustic Response

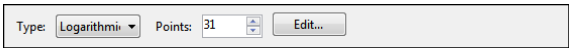
When Compare is attached to an Acoustic Response Level and Distortion result, both Channels and Harmonics can be selected.



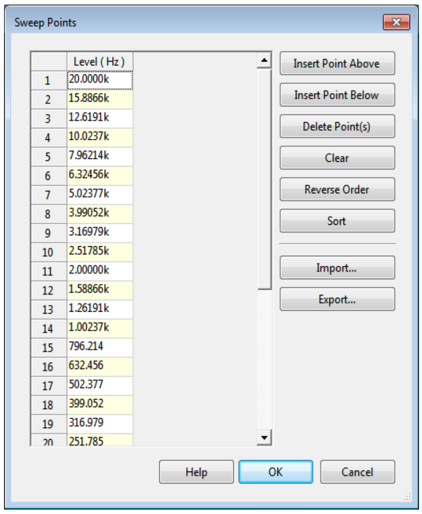
Specify Data Points

Specify Data Points can only be attached to results whose source is a continuous-sweep technology measurement: Acoustic Response, Continuous Sweep, and Frequency Response.

Specify Data Points is a derived results function that allows you to select specific frequencies for reporting results for a continuous-sweep based measurement.



After attaching Specify Data Points to a result, use the controls in the Result Settings bar to select the data points to report. Choose Logarithmic or Linear for regularly-spaced frequencies, and set the number of desired points. For exact control, click Edit to open the Sweep Points dialog and create, import or export a custom Sweep Points table.



Explanation

Measurements based on continuous sweep technology (Acoustic Response, Continuous Sweep, and Frequency Response) gather data points at thousands of frequencies. The exact frequencies used are determined at the time of measurement, and depend upon the sweep settings. The exact frequencies of sweeps appended with identical settings will also vary slightly from one another.

If you are seeking measurements at one or more specific frequencies (at 1.00000 kHz, for example, or at the series of conventional frequencies used in a 1/3-octave sweep), you will find that continuous-sweep based measurements will provide values at frequencies very close to, but not exactly at, your specified frequencies. A typical 20 Hz to 20 kHz logarithmic sweep, for example, might return data near 1 kHz at exactly 0.99750 kHz and 1.00125 kHz.

Specify Data Points allows you to select specific frequencies for reporting results. The value of the nearest neighbor in the source data is reported at the chosen frequency.

Use cases

If you are required to report measurements requirements specify data at exact frequencies, use Specify Data Points to select the required frequencies.

If you are programmatically gathering results across appended sweeps, for example, slight variances in reported frequencies can complicate programming. Use Specify Data Points to ensure that every sweep has valid results at the selected frequencies.

Specify Data Points Single Value

Specify Data Points Single Value can only be attached to results whose source is a continuous-sweep technology measurement: Acoustic Response, Continuous Sweep, and Frequency Response.

Specify Data Points Single Value is a derived results function that allows you to select a specific frequency for reporting results in a continuous-sweep based measurement. Specify Data Points can be applied to measured results or imported results.

After attaching Specify Data Points Single Value to a result, enter the frequency of choice to select the data point to report.

Specify Data Points Single Values finds the desired data point for each channel for the data sets selected in the Result Specification, and displays the value of that data point on a bar graph.

Explanation

Measurements based on continuous sweep technology (Acoustic Response, Continuous Sweep, and Frequency Response) gather data points at thousands of frequencies. The exact frequencies used are determined at the time of measurement, and depend upon the sweep settings. The exact frequencies of sweeps appended with identical settings will also vary slightly from one another.

If you are seeking measurements at one specific frequency (at 1.00000 kHz, for example), you will find that continuous-sweep based measurements will provide values at frequencies very close to, but not exactly at, your specified frequency. A typical 20 Hz to 20 kHz logarithmic sweep, for example, might return data near 1 kHz at exactly 0.99750 kHz and 1.00125 kHz.

Specify Data Points Single Value allows you to select a specific frequency for reporting results. The value of the nearest neighbor in the source data is reported at the chosen frequency.

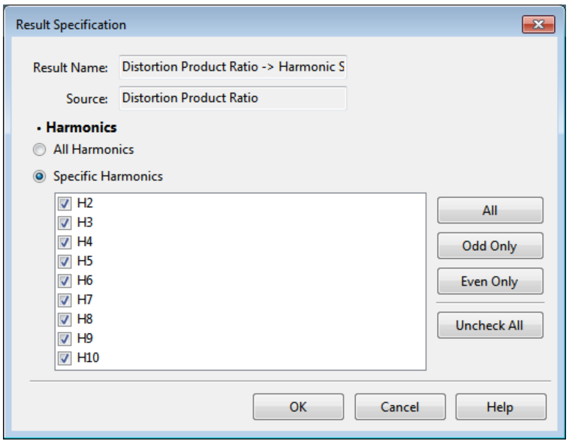
Use cases

If you are required to report measurements that specify data at an exact frequency, use Specify Data Points Single Value to select the required frequency.

If you are programmatically gathering results across appended sweeps, for example, slight variances in reported frequencies can complicate programming. Use Specify Data Points Single Value to ensure that every sweep has a valid result at the selected frequency.

Harmonic Sum (Ratio or Level)

Harmonic Sum Ratio

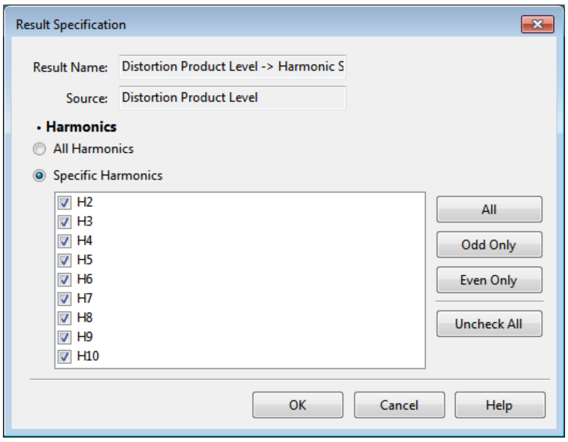


The Harmonic Sum Ratio derived result can only be attached to the THD+N Distortion Product Ratio result.

The derived result provides the sum of the levels of any combination of harmonic products H2 through

H10, divided by the signal level. The result is expressed as a ratio.

Harmonic Sum Level



The Harmonic Sum Level derived result can be attached to the THD+N Distortion Product Level result.

The derived result provides the sum of the levels of any combination of harmonic products H2 through H10.

These derived results are available for THD+N Distortion Product Ratio or Distortion Product Level results only.

Min/Max Statistics group:

The Statistics group is a set of derived results that apply statistical computations to APx500 source results. For a given source result type (single value or XY) and a given derived result output type (single value or XY), functions in the Statistics group share common settings and specifications.

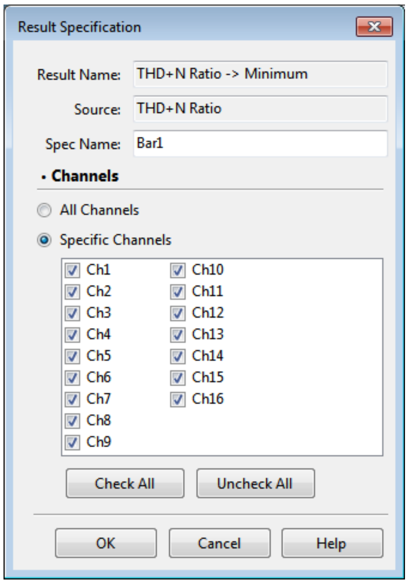
Once you have attached a Statistics group derived result, you can select and view other statistical functions without needing to define a new result. Simply select a different function from the drop-down menu on the Results Settings bar or in the Result Specification dialog.

Statistics group functions

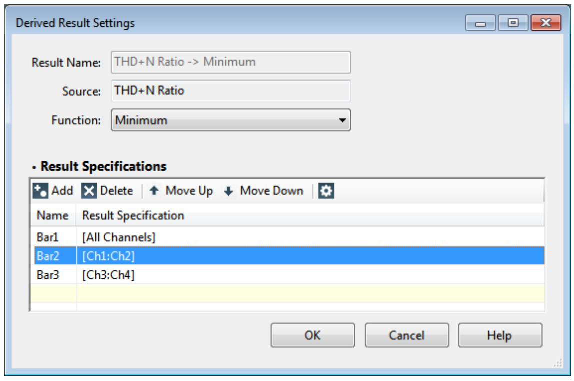
- Minimum (page 590)
- Maximum (page 590)
- Geometric Mean (page 590)
- Arithmetic Mean (page 591)
- RMS (root mean square) (page 591)
- Standard Deviation (page 591)
- Max Difference from Geometric Mean (page 591)
- Max Difference from Arithmetic Mean (page 591)

Statistics Group (single value output) attached to single value (meter) results

When attached to a meter result, a statistics (single value output) derived result gathers information across specified channels, and displays a single value result as a meter bar. Channels are selected in the Result Specification dialog, shown here.



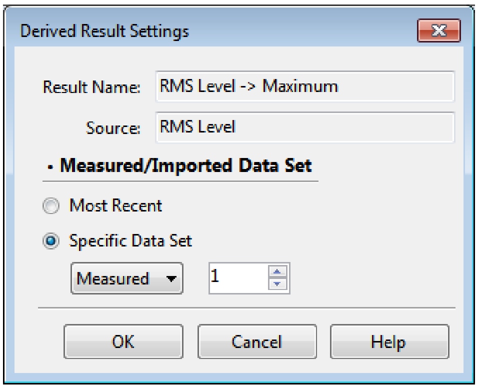
Up to 16 result specifications are supported, managed in the Derived Result Settings dialog shown here.



Statistics Group (single value output) attached to XY results

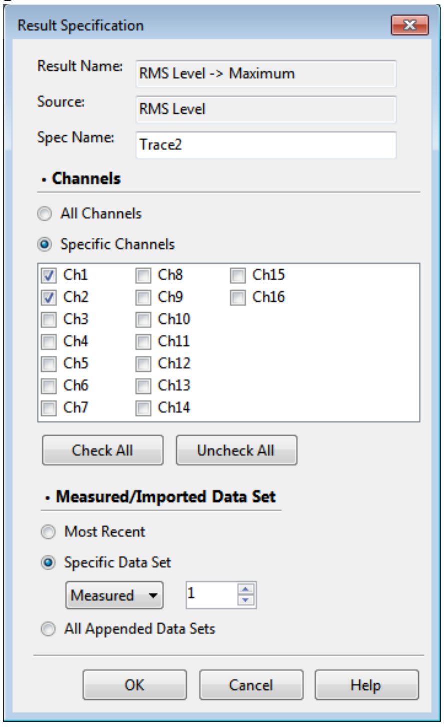
When attached to an XY result, a statistics (single value output) derived result gathers information for each channel for a specified data set, and displays a single value result as a meter bar for each channel.

Data sets are selected in the Result Specification dialog.

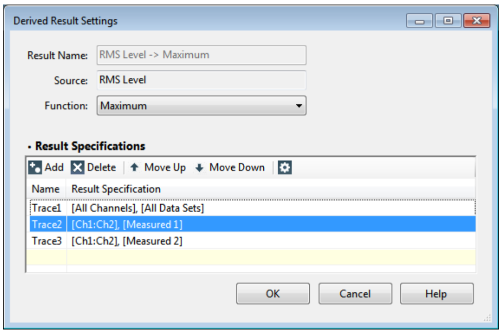


Statistics Group (XY output)

A Statistics group (XY output) derived result can only be attached to an XY source result. Information is gathered across selected channels for selected data sets. The derived result is displayed as an XY graph, with a trace for each result specification. Channels and data sets are selected in the Result Specification dialog, shown here.

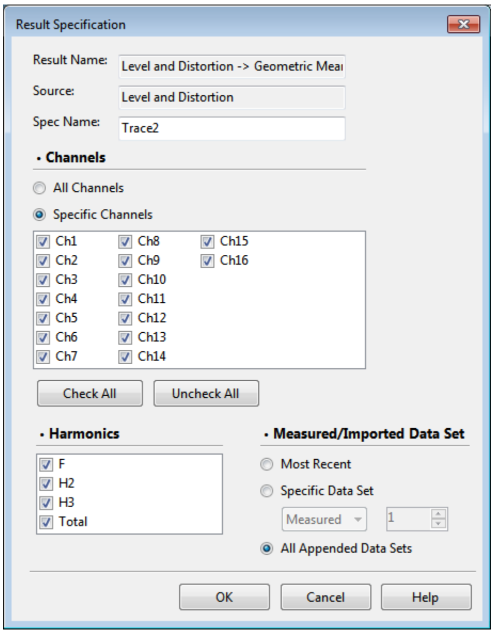


Up to 16 result specifications are supported, managed in the Derived Result Settings dialog, shown here.



Special Statistics Specifications for Acoustic Response

When a Statistics Group derived result is attached to an Acoustic Response Level and Distortion result, both Channels and Harmonics can be selected.



Minimum

Minimum is a Statistics group function that finds the lowest value in the result data it acts upon.

Maximum

Maximum is a Statistics group function that finds the highest value in the result data it acts upon.

Geometric Mean

Geometric Mean is a Statistics group function that finds the value of the geometric mean of the result data it acts upon. The geometric mean is defined as nth root of the sum of the squares of the values of n data points.

Formula:

$$\left(\prod_{i=1}^{n} a_i\right)^{1/n} = \sqrt[n]{a_1 a_2 \cdots a_n}.$$

Arithmetic Mean

Arithmetic Mean is a Statistics group function that finds the arithmetic mean (average) of the result data it acts upon. The arithmetic mean is defined as the sum the values of all data points divided by the number of all data points.

Formulae:

(x1+x2+x3...xn)/n.

or, in summation notation,

$$\bar{x} = \frac{1}{n} \cdot \sum_{i=1}^{n} x_i$$

RMS (root mean square)

RMS is a Statistics group function that finds the root mean square of the result data it acts upon. The root mean square (also called the quadratic mean) is defined as the square root of the sum of the squares of the values of all data points.

Standard Deviation

Standard deviation is a Statistics group function that finds the standard deviation (the degree of variation from the mean) of the result data it acts upon.

Definition of standard deviation

The standard deviation function shows the degree of variation from the arithmetic mean (average).

Standard deviation is the square root of the arithmetic mean of the variance of each point. The variance is the squared deviation (the difference between the point value and the arithmetic mean of all point values, squared). To compute this, first find the arithmetic mean of all points. Then find the difference of each point from the mean, and square each of these results. This is the variance for each point. Then find the arithmetic mean of all the variances. Lastly, find the square root of this mean.

For example, for 9 points (2, 4, 6, 4, 3, 5, 7, 3, 2), the average (arithmetic mean) is the sum of all the values divided by the number of values, or 36/9 = 4.

The deviation of each point from the average (4) is (–2, 0, 2, 0, –1, 1, 3, –1, –2).

The squared deviation, or variance, of each point is (4, 0, 4, 0, 1, 1, 9, 1, 4).

The average of these variances is 24/9 = 2.67.

The square root of this average is the standard deviation = 1.633

Formulae:

Where the standard deviation is represented by (sigma), and the arithmetic mean is represented by μ (mu), and X is the value of each point, and N is the number of points:

$$\sigma = \sqrt{\frac{(x_1 - \mu)^2 + (x_2 - \mu)^2 + \cdots + (x_N - \mu)^2}{N}}$$

or, in summation notation,

$$\sigma = \sqrt{\frac{1}{N} \sum_{i=1}^{N} (x_i - \mu)^2}$$

Max Difference from Geometric Mean

Max Difference from Geometric Mean is a Statistics group function that finds the difference between the value of the data point(s) with the maximum difference from the geometric mean and the value of the geometric mean, in the result data it acts upon.

Max Difference from Arithmetic Mean

Max Difference from Arithmetic Mean is a Statistics group function that finds the difference between the value of the data point(s) with the maximum difference from the arithmetic mean and the value of that mean, in the result data it acts upon.

Data Distribution

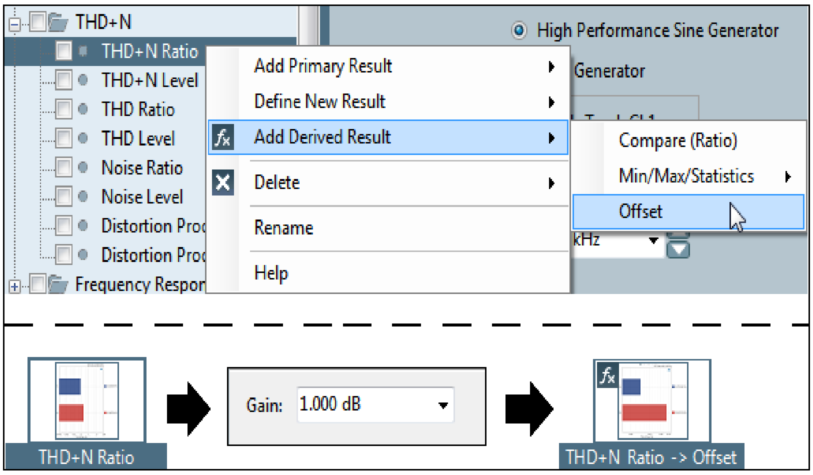
This a statistical result that displays the confidence interval, a measure of the reliability of an estimate, such as a mean. When computing a MOS average, the ITU P.862.1 requires, among other results, a 95% confidence interval result.

Assuming the data distribution is Gaussian, the 95% point corresponds to 1.96 standard deviations from the mean, and the result displayed is 1.96 times the standard deviation of the input data.

The result is expressed as a mean value followed by a +/- value for the deviation.

Example 1. Attach Offset to THD+N Ratio

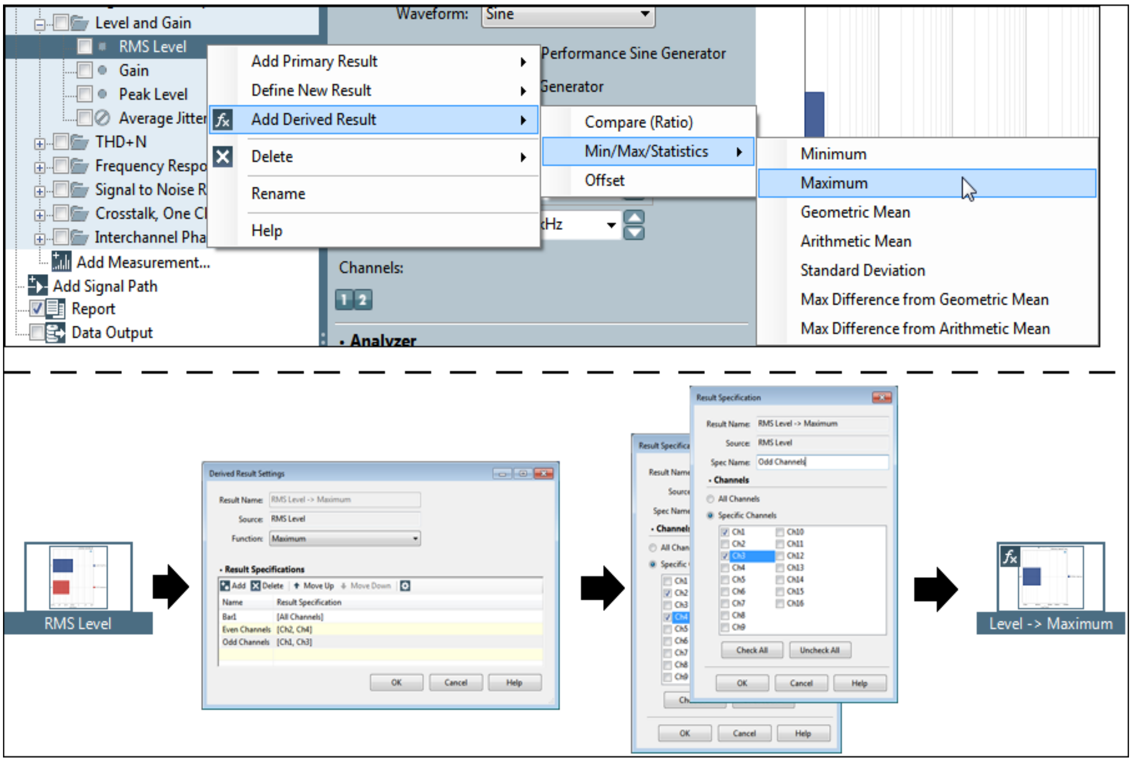
Attaching a Derived Result Offset Function to a THD+N Ratio result.



Attaching the Derived Result to the source result places a Gain field on the Result Settings bar, where you can specify the offset gain. We have entered 10.00 %. The Derived Result is of the single value type, with meter bars showing the offset for each channel.

Example 2. Attach Maximum (single value) to RMS Level

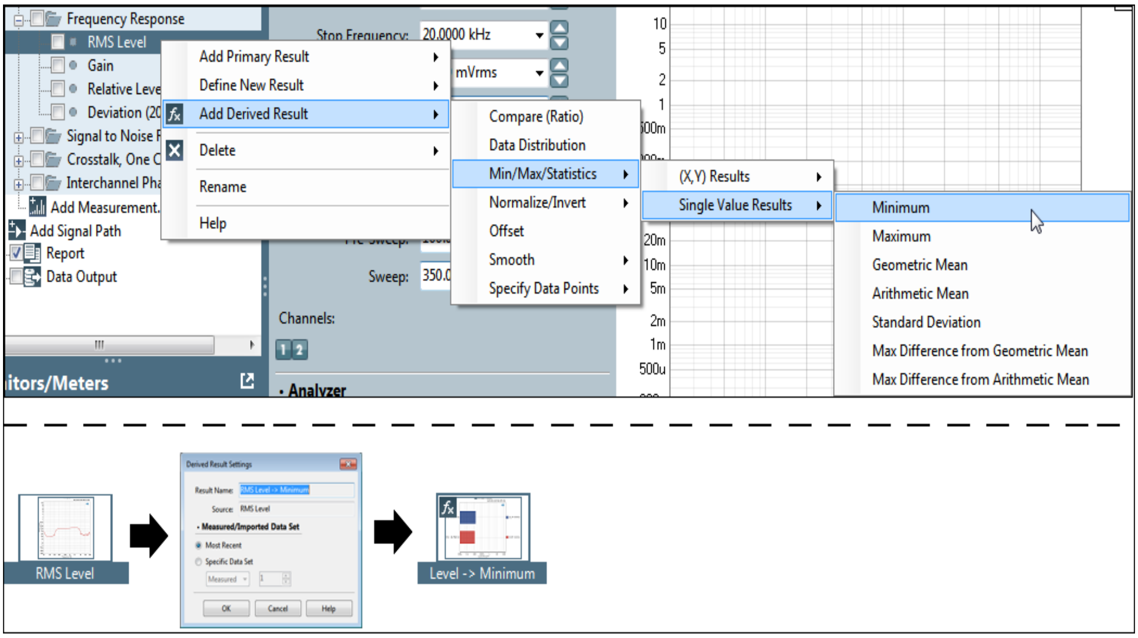
Attaching a Derived Result Maximum Function to an RMS Level result.



Attaching the Derived Result to the source result opens a Result Settings dialog, where you can specify one or more Result Specifications. We have made two specifications: one that returns the maximum across the even channels, and one that returns the maximum across the odd channels. The Derived Result is of the single value type, with one meter bar for each specification.

Example 3. Attach Minimum (single value) to Frequency Response Level

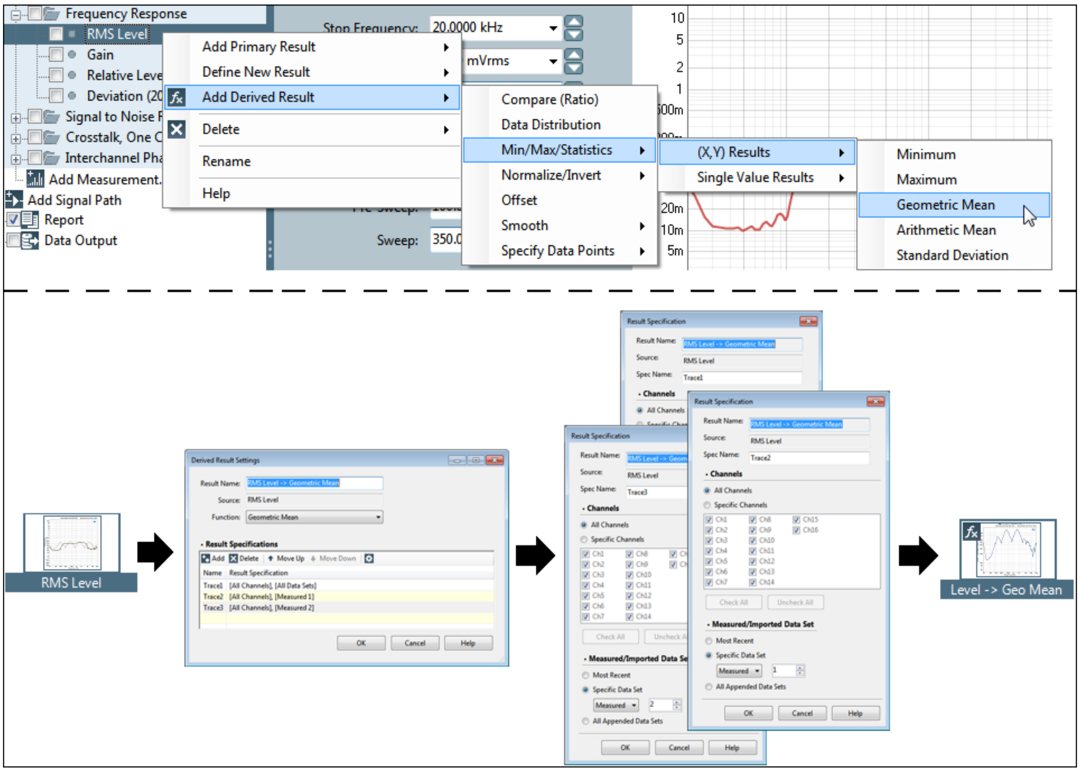
Attaching a Derived Result Minimum Function to a Frequency Response Level result.



Attaching the Derived Result to the source result opens a Result Settings dialog, where you can specify the Data Set to be considered. We have kept the default, “Most Recent”. The Derived Result is of the single value type, with one meter bar for each channel.

Example 4. Attach Geometric Mean (XY) to Frequency Response Level

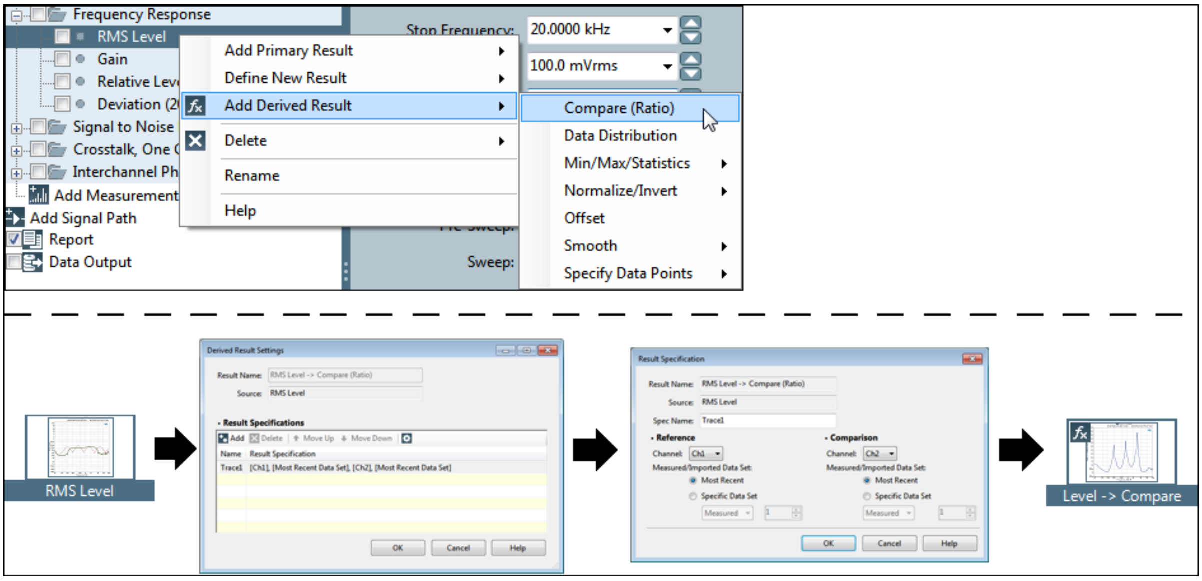
Attaching a Derived Result Geometric Mean (XY) Function to a Frequency Response Level result.



Attaching the Derived Result to the source result opens a Result Settings dialog, where you can specify one or more Result Specifications. We have made three specifications: one that returns the geometric mean across channels 1 and 2, for Data Set 1; one that returns the geometric mean across channels 1 and 2, for Data Set 2, and that returns the geometric mean across all channels for all Data Sets. The Derived Result is of the XY type, with one graph trace for each specification.

Example 5. Attach Compare (XY) to Frequency Response Level

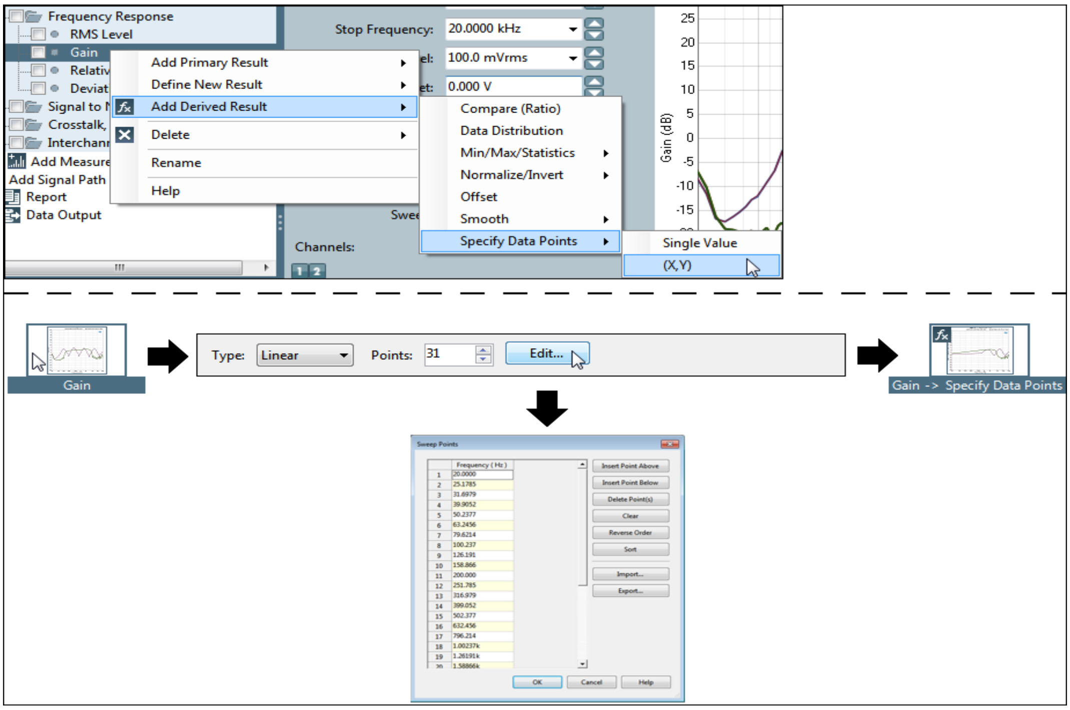
Attaching a Derived Result Compare (XY) Function to a Frequency Response Level result.



Attaching the Derived Result to the source result opens a Result Settings dialog, where you can specify one or more Result Specifications. We have only one specification, the default, which compares the data in channel 1 with the data in channel 2, for the most recent data set. The Derived Result is of the XY type, with one graph trace for each specification.

Example 6. Attach Specify Data Points (XY) to Frequency Response Gain

Attaching a Derived Result Specify Data Points Function to a Frequency Response Gain result.



Attaching the Derived Result to the source result places Type and Points fields on the Result Settings bar, where you can specify Linear or Logarithmic and the number of points, or you can open a Sweep Points dialog to add or delete specific points. We have kept the defaults. The Derived Result is of the XY type, with all traces redrawn with the new data points.

96

Aux Control

Overview

Aux Control (sometimes referred to as GPIO or General Purpose Input/Output) provides the capability to communicate with external devices, transmitting and receiving control commands via general-purpose 8-bit digital ports, available on 9-pin D-Sub connectors on the APx500 series instrument rear panel.

Aux Control Out

Aux Control Out commands can be set manually in the APx500 software, or can be set automatically in response to a specified condition in an APx500 sequence. Applications include control of mechanical devices such as turntables, mains power, indicator lights, annunciators, and so on.

Aux Control In

Aux Control In commands can be read visually in the APx500 software, or can be used to trigger actions in an APx500 sequence. Applications include input of operator controls (such as a foot switch) and reading of device states.

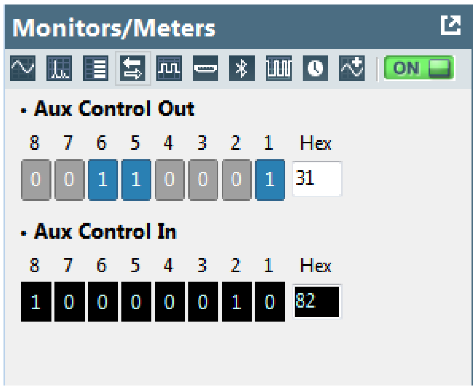
Reading and Setting Aux Control

Viewing and manually setting Aux Control has been placed in a Monitor view, which allows you to work with Aux Control without leaving the current Measurement view.

Go to the monitors and click the **Aux Control** button.

The Aux Control Monitor displays the current state of the Aux Control Out and Aux Control In bits. Bit states are shown both in a binary display and as the hex equivalent.

Additionally, Aux Control Out bits can be set from this monitor window, either by clicking on the binary display to toggle the state of a bit, or by entering the hex equivalent in the Hex: field.

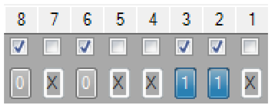


Using Aux Control with Sequences

You can read or set Aux Control states from within a sequence step. See pages 484 and 488.

You can start a sequence with an Aux Control In command, and set the Aux Control out state at the beginning of the sequence. See page 481. You can also set the Aux Control out state at the end of a sequence, in response to pass, fail or sequence cancellation conditions. See page 481.

You can also cause sequence dialogs, such as Pass or Fail, to be closed with an Aux Control In command. See page 482.



The Aux Control interface used in the Sequence has three states for each bit: 1, 0, and “X”. X is an “ignore” or “don’t care” setting.

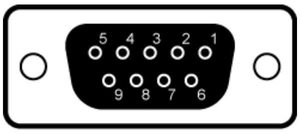
For an Aux Control In interface, X means that for that bit, either a 1 or a 0 will satisfy the pattern.

For an Aux Control Out interface, X means that the current state of the bit will be left unchanged.

Aux Control Physical Layer

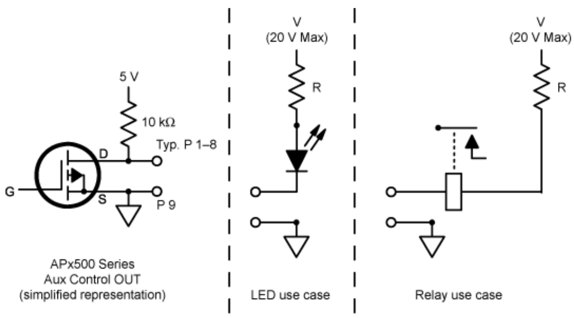
Aux Control Out

Aux Control Out uses open-drain relay drivers, capable of sinking 250 mA per pin. The On resistance is nominally 1.3 Ω. The open drain is pulled high to 5 V with a 10 kΩ resistor.



Aux Control Out DE-9F connector

Pin 9 is the common (ground) connection; pins 1–8 are numbered to correspond with the Aux Control bits 1–8.

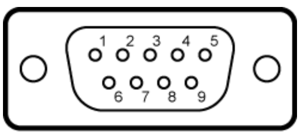


LEDs are typically connected using TTL driver circuits, but can be connected directly as shown above. V and R depend upon the LED selected and other circuit considerations.

Relays are typically connected using TTL driver circuits, but can be connected directly as shown above. If the relay package does not provide an internal suppression diode, be sure to protect the circuit by connecting one across the relay coil. V and R depend upon the relay selected and other circuit considerations. Maximum current per Aux Control Out pin is 250 mA.

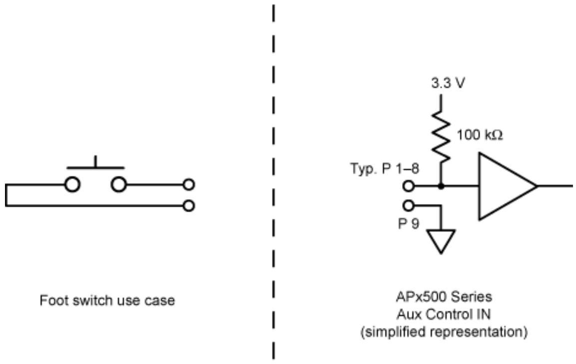
Aux Control In

The input port is a TTL-compatible interface with 5 V tolerant inputs. Vinh (high) is 2.0 V and Vinl (low) is 0.8 V. The inputs are pulled high with a 100 kΩ resistor to 3.3 V. Maximum continuous input voltage is 5.2 V.



Aux Control In DE-9M connector

Pin 9 is the common (ground) connection; pins 1–8 are numbered to correspond with the Aux Control bits 1–8.



Input switches and sensors are often connected using TTL driver circuits, but can be connected directly as shown above.

Switchers

Introduction

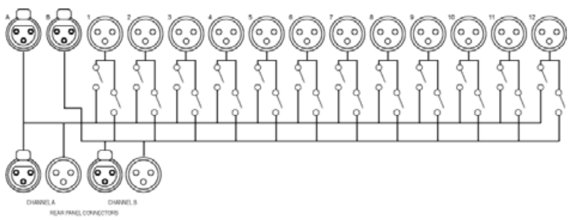
Audio Precision manufactures optional switchers that can be used to connect or disconnect many signal channels (up to 192 input and 192 output channels) to the instrument inputs or outputs, under software control.

This chapter discusses the APx500 switcher control interface and provide an overview of switcher features and applications.

Switchers can also be configured from within a sequence by adding a Switcher Configuration step. See page 491.

See the SWR-2755 User’s Manual (provided with the switcher and available on the Audio Precision Web site) for more information.

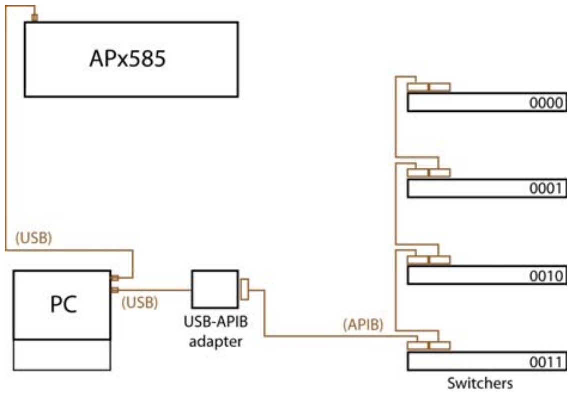
Balanced output switcher SWR-2755M



Block diagram of SWR-2755M switcher

Connecting the switcher to the control computer

Audio Precision switchers use the Audio Precision Interface Bus (APIB) for switcher control. To use a switcher with APx500, you must have at least 2 USB 2.0 ports on the control computer. One of the ports must be connected to the APx500 instrument; the other port must be connected to an Audio Precision USB-APIB adapter, which can then be connected to one or more switchers using APIB. The USB-APIB is an accessory interface available from Audio Precision.



Switcher control interface bus diagram

Audio Precision USB-APIB adapter

Setting switcher addresses

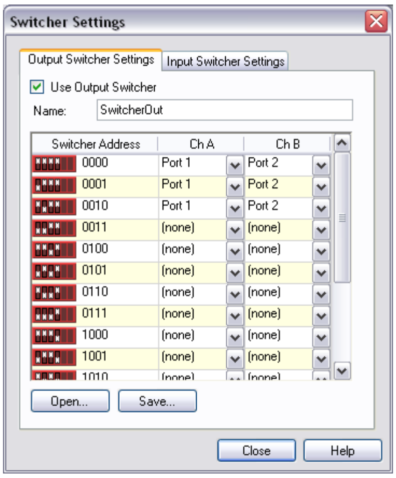
APx500 can control up to 16 input and 16 output switchers, for a total of 192 switched input channels and 192 switched output channels. Each switcher is identified by setting a DIP switch on the SWR-2755 hardware to a binary address, and entering the matching binary address in the software.

Switcher address settings

Each switcher must be assigned a unique binary address. On the switcher rear panel is a DIP switch with 6 switches; the four switches to the left set the binary address for that unit. To address a switcher from the APx500 software, the binary address must match on the switcher hardware and the switcher address controls in the software.

Note that the DIP switch orientation on the hardware is reversed from the digit order in ordinary binary notation, and that 0=UP and 1=DOWN: binary number 0111 corresponds to switch settings of DOWN-DOWN-DOWN-UP. For clarity, both DIP switch position and binary number are shown in the control panel.

From Signal Path Setup, click **Switcher Settings**. Choose the **Output Switchers Settings** tab or the **Input Switchers Settings** tab to view a Switcher panel.



Switcher Address

This is the binary address for each switcher hardware unit. On the switcher rear panel is a DIP switch with 6 switches; the four switches to the left set the binary address for that unit. To address a switcher from the APx500 software, the binary address must match on the switcher hardware and the switcher address controls in the software.

Switcher Channels A and B

Each switcher has 2 channels called Channel A and Channel B, that are used to connect to 2 APx 500 series outputs (for an output switcher) or inputs (for an input switcher). These are connected to the switcher ports by internal relays, under the APx500 software control.

Switcher Ports

Each switcher has 12 ports that are used to connect to DUT channels according to your test requirements. An input switcher's ports are designed to connect to a DUT's output channels, and an output switcher's ports are designed to connect to a DUT's input channels. For each APx500 signal path, you can make a port selection for Channel A and Channel B for each addressed switcher.

All But ChA

The Channel B port selection list includes All But ChA as a choice. This selection connects switcher Channel B to all ports except the current port connected to Channel A.

Saving switcher settings

APx500 switcher settings are always saved with the project file, and are included in the project when the file is re-opened.

Additionally, output and input switcher settings can be saved as individual files from the Signal Path Switchers panels. Output switcher settings files have the filetype *.swo, and input switcher settings files have the filetype *.swi.

Click the **Save** button to save the current switcher settings as a separate file. Click the **Open** button to open previously saved switcher files.

Creating a signal path for each step

Each Navigator signal path sets one state for each of the switchers connected, set in the Advanced Settings panel for that signal path. For example, the input switcher at address 0000 could have its ChA connector switched to Port 1, and its ChB connector switched to Port 2.

To switch to two different DUT channels, create a second Navigator signal path using the same 0000 switcher address, but set ChA to Port 3 and ChB to Port 4. Make the necessary audio connections to match. When you move from Signal Path 1 to Signal Path 2, the switcher will change to Ports 3 and 4.

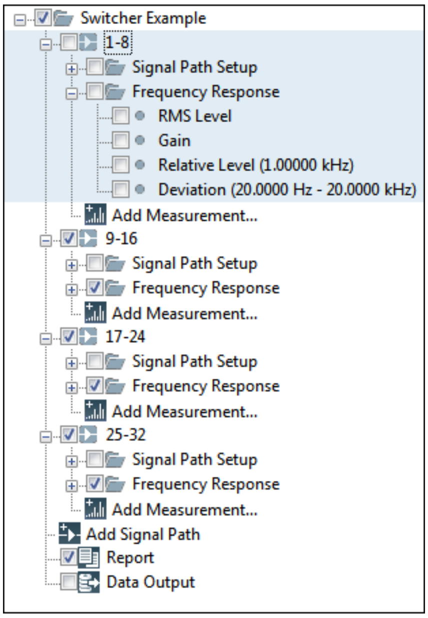
Switcher example

The block diagram on the next page shows an APx585 testing a 32x8 mixer using four output switchers. Generator pairs 1–2 are connected to Channels A and B on switcher 0000, 3–4 to A and B on switcher 0001 and so on.

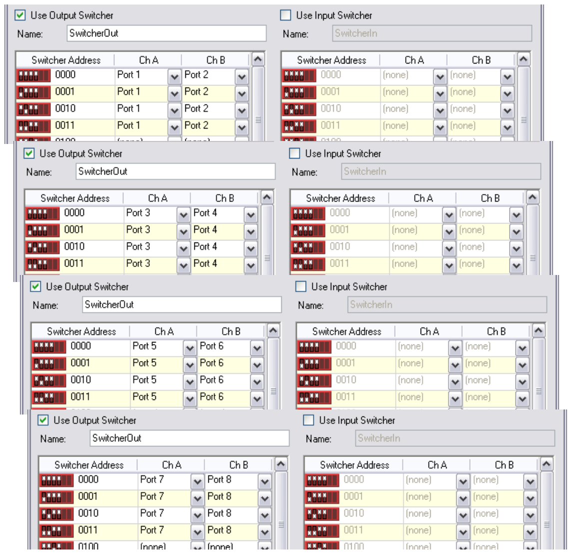
The switcher ports are connected to various mixer inputs. Ports 1–2, 3–4, 5–6 and 7–8 from 0000, for example, carry Generator signal from 1–2, and are wired to mixer inputs 1–2, 9–10, 17–18, and 25–26.

The eight outputs from the mixer are connected directly to the analyzer inputs. If there were more outputs to be tested, input switchers could be added to the setup.

The four Navigator signal paths shown here shift the switchers through the four states necessary to test all the mixer channels. The four switcher address configurations show the switcher states specified in each of the four signal paths.



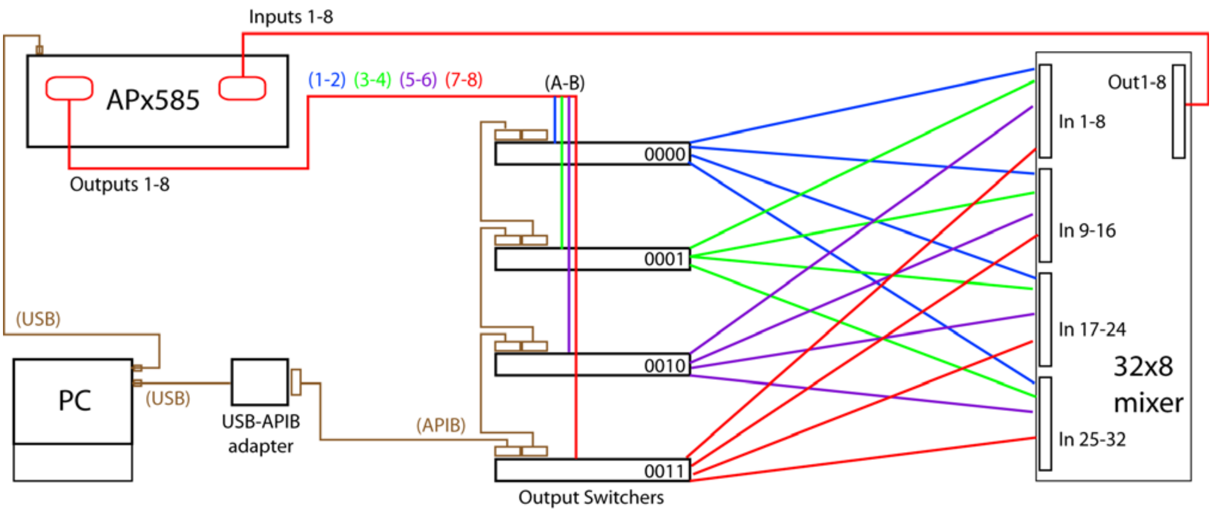
Four Signal Paths in the Navigator



Four sets of switcher settings

When this sequence is run, each signal path performs its measurements on eight channels at a time, and then switches to the next signal path and the next eight channels.

All results are shown in the sequence report.



Block Diagram

98

Units of Measurement

Unit Domains

Many different units of measurement are used to describe audio signals and audio system components and characteristics. Values can be expressed in absolute terms such as the volt, the watt, the ohm, the hertz and the second. Values can also be displayed as the ratio of one value to another using units like the decibel or percent. In addition, APx500 provides relative units which express the relationship of a value to a user-supplied reference, or to another APx500 setting.

Measurements using Audio Precision analyzers fall into one of eight *unit domains*. Each unit domain has a *base unit*, from which the equivalent units are calculated.

Depending on the current APx500 view and the parameter you are setting or measuring, different units choices in different unit domains will be available. When you request a value in different units, APx500 recalculates the value.

Unit domains used in APx500:

- Analog Level
- Digital Level
- Ratio
- Ratio, cross-domain
- Relative
- Frequency
- Phase
- Time
- Acoustic
- No Unit (Impulse Response only)

Units: Analog Level

V—volts (instantaneous voltage)

This unit is used in the APx500 DC Level measurement, and in the time domain views, including Signal Monitors: Scope, Signal Analyzer: Scope; in Acoustic Response: Impulse Response and Acquired Waveform, Continuous Sweep: Impulse Response and Acquired Waveform, Multitone: Acquired Waveform, Multitone: Noise Density and Multitone: DC Level measurements.

Vrms—volts (rms)

This unit is used in APx500 generator settings and analyzer measurements. This setting accurately calibrates the generator output in Vrms when a sine wave is being generated. When using other waveforms (such IMD twin-tone stimulus signals) with Vrms, the signal has the same peak-to-peak voltage as a sine wave at that setting. For the analog outputs, Vrms is an open-circuit value and the actual output voltage will be less with a load. In analyzer measurements, Vrms readings are always made using an rms detector and are correct for any waveform.

Vp—volts (peak)

This unit is used in APx500 generator settings. This calibration is arithmetically scaled to the Vrms value by the square root of 2, approximately 1.414. Since non-sinusoidal waveforms are calibrated to have the same peak-to-peak value as a sine wave, the Vp unit should be correct for most non-sinusoidal waveforms. For the analog outputs, Vp is an open-circuit value and the actual output voltage will be less with a load.

Vpp—volts (peak-to-peak)

This unit is used in APx500 generator settings. This calibration is arithmetically scaled to the Vrms value by 2 times the square root of 2, approximately 2.828. Since non-sinusoidal waveforms are calibrated to have the same peak-to-peak value as a sine wave, the Vpp unit should be correct for most non-sinusoidal waveforms. For the analog outputs, Vpp is an open-circuit value and the actual output voltage will be less with a load.

dBV—decibels relative to 1.000 Vrms

This is usually the correct choice of units for audio levels in consumer audio equipment and systems. The dBV is a voltage unit and, like the dBu, is independent of circuit impedance. For the analog outputs, dBV is an open-circuit value and the actual output voltage will be less with a load. See More about decibels on page 606.

dBu—decibels relative to 0.7746 Vrms

This is usually the correct choice of units for audio levels in professional audio equipment and systems. The dBu is a voltage unit and, unlike the dBm, is independent of circuit impedance. For the analog outputs, dBu is an open-circuit value and the actual output voltage will be less with a load. See More about decibels on page 606 and More about dBm vs. dBu on page 607.

dBm—decibels relative to 1 mW

dBm is rarely the correct choice of units for audio measurements. dBm units are often Improperly chosen when dBu units should be used. See dBm versus dBu on page 607.

Like watts, the dBm is a unit of power used in APx500 generator settings and analyzer measurements.

dBm as generator output units

The APx generator outputs have a source impedance of 100 Ω for the balanced connections and 50 Ω for the unbalanced connections. When using dBm as a generator unit, the internal generator level is set to the value that produces the desired power in the load, calculated for the balanced (100 Ω source impedance) outputs. For a correct calculation, you must enter the value of the load resistance in the Reference Levels > Advanced Settings dialog. 600 Ω is the most common circuit impedance for dBm measurements.

As mentioned, this calculation is made correctly for the balanced outputs. The unbalanced outputs present a source impedance of 50 Ω, which is not considered in the dBm units calculation. When using the unbalanced outputs with dBm generator units, the power in the load will not match the value specified in the generator Level field.

Note: units of power (dBm and watts) in generator output settings are features included in Audio Precision instruments as a nod to tradition, but are rarely useful. We do not recommend the use of dBm or watts as generator units.

dBm as analyzer input units

For the analyzer inputs, dBm is accurately measured and calculated for both the balanced and unbalanced inputs. For a correct calculation, you must enter the load impedance at the output of your DUT in the Reference Levels > Advanced Settings dialog. 600 Ω is the most common circuit impedance value for dBm measurements.

W—watts

Like dBm, the watt is a unit of power used in APx500 generator settings and analyzer measurements.

watts as generator output units

The APx generator outputs have a source impedance of 100 Ω for the balanced connections and 50 Ω for the unbalanced connections. When using watts as a generator unit, the internal generator level is set to the value that produces the desired power in the load, calculated for the balanced (100 Ω source impedance) outputs. For a correct calculation, you must enter the value of the load resistance in the Reference Levels > Advanced Settings dialog. 8 Ω is the default load for watts measurements.

As mentioned, this calculation is made correctly for the balanced outputs. The unbalanced outputs present a source impedance of 50 Ω, which is not considered in the watts units calculation. When using the unbalanced outputs with watts generator units, the power in the load will not match the value specified in the generator Level field.

Note: units of power (dBm and watts) in generator output settings are included features in Audio Precision instruments as a nod to tradition, but are rarely useful. We do not recommend the use of dBm or watts as generator units.

watts as analyzer input units

For the analyzer inputs, watts are accurately measured and calculated for both the balanced and unbalanced inputs. For a correct calculation, you must enter the load impedance at the output of your DUT in the Reference Levels > Advanced Settings dialog. 8 Ω is the default load for watts measurements.

ohm (Ω)

The ohm is the unit of electrical resistance (R). Its relationship to voltage (V) and current (I) in a circuit is stated by ohm's law, V=IR.

Units: Digital Level

FS—digital full scale

Digital full scale is defined in terms of a sine wave whose peak just reaches the maximum positive code. The rms value of such a sine wave is 1 FS. See More about Digital Level units on page 607.

Other Audio Precision analyzers have used the term FFS (Fraction of Full Scale) for this unit of digital level. 1 FFS = 1 FS.

%FS—per cent of full scale

%FS is the ratio of the rms value of the signal to 1 FS, expressed as a decimal fraction and multiplied by 100. A sine wave half the value of full scale would be indicated as 50 %FS.

dBFS—decibels relative to full scale

This is a logarithmic measurement defined as 20 log10(A/B), where A is the rms level of the signal to be expressed and B is level of a full scale sine wave (1 FS). For example, a signal at 0.1 FS equals –20 dBFS. dBFS is probably the most common unit of measurement used for digital audio amplitudes. 0 dBFS is the full-scale reference.

D—instantaneous digital level

D, the instantaneous digital level, is defined in terms of an instantaneous signal whose peaks just reach the maximum digital codes. The peak value of such a signal that reaches the maximum positive code is +1 D; the peak value of such a signal that reaches the maximum negative code is –1 D.

This unit is used in the APx500 DC Level measurement, and in the time domain views, including Signal Monitors: Scope, Signal Analyzer: Scope; in Acoustic Response: Impulse Response and Acquired Waveform, Continuous Sweep: Impulse Response and Acquired Waveform, Multitone: Acquired Waveform, Multitone: Noise Density and Multitone: DC Level measurements.

hex—hexadecimal

The hex unit express the instantaneous numerical value of the embedded audio sample in base 16 (hexadecimal) notation, and is used for Constant Value waveform levels, and in the APx500 DC Level measurement, and in the time domain views, including Signal Monitors: Scope, Signal Analyzer: Scope; in Acoustic Response: Impulse Response and Acquired Waveform, Continuous Sweep: Impulse Response and Acquired Waveform, Multitone: Acquired Waveform, Multitone: Noise Density and Multitone: DC Level measurements.

Units: Ratio

X/Y

The X/Y unit is the simple ratio between two numbers.

%—per cent

The per cent unit expresses the ratio of two numbers multiplied by 100. For example, a ratio of 0.7 is expressed as 70 %.

ppm—parts per million

Parts per million expresses the ratio of two numbers, multiplied by 1,000,000. For example, a ratio of 0.00005 equals 50 ppm.

dB—decibel

The decibel is always a ratio of two amplitudes, expressed logarithmically. For voltage amplitudes, decibels are computed by dB=20 log10(V1/V2). See More about decibels on page 606.

Specific ratio units, such as dBu or %Hz, are discussed in more detail in the relevant sections on amplitude or frequency units.

Units: Ratio, cross-domain

FS/Vrms

For cross-domain measurements, the FS/Vrms unit is the ratio between output and input levels, for devices (such as an ADC) that have an analog input and a digital output.

Vrms/FS

For cross-domain measurements, the Vrms/FS unit is the ratio between output and input levels, for devices (such as a DAC) that have a digital input and an analog output.

D/V

For cross-domain impulse response measurements, the D/V unit is the relationship between output and input for devices (such as an ADC) that have an analog input and a digital output.

V/D

For cross-domain impulse response measurements, the V/D unit is the relationship between output and input for devices (such as a DAC) that have a digital input and an analog output.

See Analog Level units earlier in this chapter for more information about V and Vrms. See Digital Level units for more information about FS and D.

Units: Relative

dBrG

dBrG is a user memory, storing decibels relative to a user-defined output (generator) reference level setting. There are two independent dBrG memories: one for an analog output level and one for a digital output level. See More about decibels on page 606.

dBrA and dBrB

dBrA and dBrB are user memories, storing decibels relative to user-defined input (analyzer) references. See More about decibels on page 606.

You can manually enter dBrA and dBrB values in the fields in the Reference Levels view, or you can set them from measured input levels using the Set buttons in the same view. There are two independent pairs of dBr memories: one pair for analog input levels and one pair for digital input levels.

dBSPL1 and dBSPL2

dBSPL1 and dBSPL2 are user memory references just like dBrA and dBrB, with nomenclature to support Sound Pressure Level work.

Units: Frequency

Hz—hertz

This is the basic frequency unit, expressing the number of cycles of a signal in one second.

The other (following) frequency units used in APx500 measurements are relative units, expressing the relationship of the measured frequency to a reference frequency you have entered into the References Levels > Advanced Settings dialog.

dHz—delta hertz

dHz is the difference in frequency between the measured frequency and the reference frequency; dHz=F-R.

F/R

This is a frequency ratio unit, obtained by dividing the measured frequency F by the reference frequency R.

%Hz—per cent hertz

%Hz is obtained by dividing the measured frequency by the reference frequency and multiplying the result by 100; %Hz=100(F/R). For example, a measured frequency of 950 Hz would be expressed as 95.0 %Hz of a reference frequency of 1 kHz.

Units: Phase

deg—degree

The degree is the basic unit of phase measurement, representing 1/360 of a complete revolution of phase.

rad—radian

One radian (rad) is the angle subtended at the center of a circle by an arc of circumference equal in length to the radius of the circle.

2π rad = 360°

1 rad = 360° / 2π = 57.2958° (approx).

Units: Time

s—seconds

Seconds are the basic time unit, abbreviated s. At values common in audio, the display may be in ms (milliseconds), µs (microseconds) or ns (nanoseconds).

Units: Acoustic

Pa—pascal

The pascal (Pa) is a measure of force, defined as 1 newton per square meter. Atmospheric pressure is measured in pascals.

dBSPL—decibels sound pressure level

Sound pressure level is often expressed in decibels relative to a pressure measured in pascals. 0 dBSPL is defined as 20 µPa, considered the threshold of human hearing.
1 Pa = 94 dBSPL.

Units: Unitless x/y (Impulse Response only)

When the input and output domains of a DUT are the same (analog to analog, for example), the impulse response is a unitless property of the device. We express this in APx500

as “x/y”.

See More about Impulse Response on page 218.

More about decibels

The decibel (dB), in one form or another, is arguably the most widely-used unit of measurement in the field of audio. The ear’s response to both sound amplitude and frequency is usually best examined in logarithmic terms, and the dB, being a logarithmic unit of measurement, is often the right choice.

It is important to remember that the dB is always a ratio of two values, and for a meaningful expression both of the values must be known. While “8 dB” means nothing, “8 dB below the input signal” gives us a reference and is meaningful. Whenever using the simple dB unit, the expression must identify the reference value.

A large collection of units of measurement (we’ve seen it referred to as “the dB zoo”) uses the decibel ratio with the reference stated as part of the definition of the unit. Examples include dBV (decibels in relation to 1 Vrms); dBu (decibels in relation to 0.7746 Vrms); dBFS (decibels in relation to digital full scale); dBSPL (decibels of sound pressure in relation to 20 micropascals of pressure) and so on.

The formula for decibels in voltage levels is

dB=20 log10(V1/V2)

where V1 and V2 are the two related voltages.

The formula for decibels in power levels is

dB=10 log10(P1/P2)

where P1 and P2 are the two related powers.

More about RMS Level

Rms (root mean square), is the preferred form of ac signal detection which measures amplitude in terms of its equivalent power content, regardless of signal wave shape.

The rms voltage level (Vrms) is the most common level measurement for audio signals, although this level is often stated in a variety of related units such as dBV or dBu. Rms values are typically lower than peak values for the same signal.

More about dBm versus dBu

Although still in use, the dBm unit of measurement is a legacy from earlier days of audio technology when measuring the audio power transferred in impedance-matched circuits was important. The dBm is a unit of power (referenced to 1 mW), and the audio circuits of the day often had an impedance of 600 Ω. The voltage drop across a 600 Ω resistor dissipating 1 mW is 0.7746 V, and this voltage (which reads 0 dBm on a dBm meter across 600 Ω came to be regarded as "zero level" in professional audio circles.

However, most audio circuits do not have a 600 Ω impedance. Microphone circuits may be 150 Ω, and headphone circuits may be 8 Ω. This is even more the case in modern design, where circuit outputs often have very low impedances (from perhaps 10 Ω to around 100 Ω) and input circuits often are "bridging," with impedances of 10,000 Ω or even 100,000 Ω. A milliwatt dissipated in any of these circuits will not read 0.7746 V, and the dBm meter (which is simply a voltmeter calibrated for a 600 Ω circuit) will be in error. The circuit impedance must always be taken into account in power measurements.

Because of this, the dBm can be misleading. It has fallen out of use (except for specific applications where power transfer is an important consideration and the circuit impedances are known) and its place in professional audio has largely been taken by the dBu, where the "u" can be taken to mean "unloaded." The dBu is a valid level measurement in audio circuits of any impedance.

The value of 0 dBu, 0.7746 Vrms, is the same voltage as 0 dBm in a 600 Ω circuit; in 600 Ω, 0 dBu = 0 dBm. However, at other levels the voltage divider formed by the source and termination impedances will cause the values of dBu and dBm to diverge. In other impedances, the values will not be equivalent even at the zero level.

Unless you know the circuit impedance and have a clear reason to use the dBm reference, use dBu.

More about digital level units

All of the units for digital domain audio amplitudes refer to digital full scale. In pulse-code-modulation (PCM) audio, the amplitude of each audio sample is represented as a number. The maximum and minimum values of these numbers vary with the digital word length, but they are precisely defined. A signal which attempts to exceed these values can drive a digital system into overload; in a popular expression, the signal "runs out of numbers."

Word Length	***Positive Full-Scale Value***	***Negative Full-Scale Value***
16 bit	7FFF Hex	8000 Hex
24 bit	7FFFFF Hex	800000 Hex
32 bit	7FFFFFFF Hex	80000000 Hex

In AES17 the Audio Engineering Society defines digital full scale in terms of a sine wave whose positive peak just reaches the maximum positive digital code. The rms value of this sine wave is the full-scale reference, defined as 1 FS or 0 dBFS. Note that other waveforms (noise or music, for example) have different crest factors than a sine wave (crest factor is the ratio of a signal's peak amplitude to its rms amplitude) and will have different rms values at the amplitude at which their peaks just reach the maxima.

A consequence of defining full scale as an rms value is that high-level signals that have a lower crest factor than a sine wave can display rms values greater than 1 FS or 0 dBFS, even though the digital maximum and minimum codes are not exceeded. For example, a square wave whose top is at the maximum positive digital code will have an rms value of 1.414 FS or +3.01 dBFS.

Glossary

A-weighting filter
a specific noise-weighting filter (ANSI S1.4, IEC Recommendation 179) used to produce noise measurements that correlate well with human observations.

ac mains
the utility-provided ac power source from which most electronics equipment is operated. Also power line, ac line and line voltage.

acceptance limit
In pass/fail testing, the poorest measured performance specification which a manufacturer is willing to accept and ship in a product.

ADC
Analog to digital converter, also A/D converter and A-to-D converter. A device for converting an analog input signal into a series of digital values representing the instantaneous amplitude of the signal at regular sampling intervals.

AES
Audio Engineering Society, with headquarters in New York City.

AES3, AES/EBU
See **digital audio signal, bi-phase coded.**

AES3, AES/EBU
a digital interface standard for professional audio equipment interconnection. Also AES/EBU, IEC60958. See **digital audio signal, bi-phase coded**.

alias
a nonlinear signal product falling within the audio band, caused by the presence of an out-of-band (above 1/2 the sampling rate) signal during the original A/D conversion. Alias signals fall at sum and difference frequencies of the original signal frequency (or its harmonics) and the sampling rate (or its harmonics).

analog recording, processing:
Analog techniques in which a signal is represented, recorded, processed, or transmitted as a continuously variable quantity.

ANSI
American National Standards Institute, a U.S.-based standards organization.

anti-alias filter:
a low-pass filter preceding an A/D converter to prevent signals at frequencies greater than one-half the sampling rate from reaching the converter. Signals above one-half the sampling rate cannot be unambiguously converted and would appear in the digital output as signals at incorrect frequencies.

arbitrary waveform
see generator waveform file.

average responding
a form of ac signal detection indicative of the average absolute value of a waveform, typical of analog meter movements.

balanced line
an audio transmission line where the signal is applied differentially between two conductors, each of which has equal impedances to ground or common.

band-pass filter
a filter which passes a specific frequency band essentially without attenuation while attenuating frequencies both below and above the specified band.

band-reject filter
a filter which attenuates the frequencies within a specified frequency range while passing essentially without attenuation the frequencies below and above that range. If the rejection range is narrow, a band reject filter is also often called a notch filter.

bin, FFT

the basic frequency resolution-determining division of an FFT-computed spectrum. The FFT process cannot resolve signals of different frequency falling into the same bin. The bin width of an FFT can be computed from (sample rate)/(number of amplitude samples) in the digitized waveform used as input to the FFT process. Bin and line are used interchangeably.

bit

binary digit, which may have only two possible states (ON/OFF, HIGH/LOW, 1/0, etc.)

bit depth

The number of bits in an audio coding system. In high-performance linear PCM, bit depth is typically 16, 20 or 24. Telecom and compressed coded audio may use fewer bits, with depths of 14, 13, 12 or 8. Also called “word length” or “word width.”

bits of resolution

the number of bits of the binary word by which signals are represented in a digital recording or transmission system. Each bit adds approximately 6 dB to the theoretical dynamic range available. Thus, a 16-bit digital system is capable of approximately 96 dB dynamic range, etc.

bridging

a relatively high impedance input (usually balanced) which may be connected across a lower impedance circuit without significantly affecting the levels in the lower impedance circuit.

byte

an 8-bit digital word.

CCIR

Comité Consultatif International des Radiocommunications (International Radio Consultative Committee), an international standards-setting organization with headquarters in Geneva, Switzerland.

CCIR 468

a CCIR specification which includes, among other things, standards for weighted and unweighted noise measurements. The weighted standard specifies the CCIR weighting filter and a quasi-peak detector (see weighting filter). The unweighted standard specifies a 22 Hz to 22 kHz bandwidth limiting filter and an rms detector.

CCIR-2k

Also CCIR-ARM. A noise measurement technique developed by Dolby Laboratories, which uses the weighting filter shape specified in CCIR Recommendation 468 but with the unity-gain frequency at 2 kHz rather than 1 kHz, and an average-responding detector.

clipping

the action of a system in flattening and squaring off signal peaks when driven with a signal whose peak amplitude is beyond its linear signal-handling capability.

CMRR

Common mode rejection ratio.

CMT

Common mode testing.

CMTST

common mode test; also, the name of the APx output mode for fixed common mode testing.

consumer

In audio, often used to designate digital audio format. See digital audio signal, bi-phase coded.

continuous sweep

APx500 term for log chirp measurement. In the continuous sweep measurement view a broadband stimulus is generated, acquired and processed providing a family of measurement results.

crest factor

the ratio of a signal's peak amplitude to its rms amplitude.

crosstalk

unwanted signal coupling from one channel of a multi-channel transmission or recording system to another.

DAC

Digital to analog converter, also D/A converter and D-to-A converter. A device which converts a stream of digital numbers, each representing the amplitude of a signal at a particular sampling time, into a corresponding analog signal.

dB

decibel.

dB/octave

dB per octave. A standard means of referring to the ultimate rejection slope (attenuation versus frequency) of a band-limiting filter. Each pole of a band-limiting filter produces an ultimate rejection slope of 6 dB/octave (20 dB/decade). Thus, a 3-pole filter will have a rejection slope of 18 dB/octave (60 dB/decade).

dBFS

decibels with respect to digital full scale. The full scale amplitude (0 dBFS value) is the rms value of a sine wave whose positive peak just reaches positive full scale.

dBm

dB relative to a reference value of 1.000 milliwatts. dBm is a power unit and requires knowledge of power levels (voltage and current, or voltage and impedance, or current and impedance) rather than merely voltage.

dBr

relative dB; dB relative to an arbitrary reference value. The reference value must be stated for this to be a meaningful unit.

dBu

dB relative to a reference of 0.7746 V.

dBV

dB relative to a reference value of 1.000 V.

decade

the interval between two frequencies with a ratio of exactly 10:1, such as the range from 20 Hz to 200 Hz, or from 1 kHz to 10 kHz.

decibel

decibel, a ratio unit for expressing signal amplitudes. If the amplitudes are expressed in voltage, dB = 20 log10 (V1/V2). If the amplitudes are expressed in power, dB = 10 log10 (P1/P2).

detector

the precision ac to dc conversion section of a measurement instrument, located following all ac signal processing and prior to the indicating portion. Detectors are classified according to which parameter of the input ac signal the output dc value linearly follows-true rms, average, peak, etc.

difference product

an intermodulation distortion signal at the frequency which is the difference between two applied signal frequencies. Sometimes referred to as difference tone. For example, test signals of 13 kHz and 14 kHz will produce a 1 kHz difference product when applied to a device which has asymmetrical nonlinearity.

digital audio signal, bi-phase coded

In the consumer and professional audio field, digital audio is typically carried from point to point as bi-phase coded signal, commonly referred to as AES3, AES/EBU or S/PDIF. There are electrical and bitstream protocol differences among the variations of bi-phase coded digital audio, but the various signals are largely compatible.

a. protocols

The “professional” and “consumer” protocols are indicated in the digital signal’s status bits. With some exceptions, the protocols are compatible; even if status bit information is lost, the audio is usually passed. Although either protocol can be carried on any of the electrical or optical formats, systems using the professional formats typically use the professional protocol, and those using consumer formats use the consumer protocol. In APx500 series instruments, by default the digital audio protocol is set to “consumer.” When you are using the electrical digital output, you can optionally set the Professional checkbox in the Signal Path Setup view. This sets the electrical signal to the professional level, and also sets the protocol and status bits to “professional.”

b. electrical formats

AES3, AES/EBU, IEC60958-4 all refer to an electrically balanced professional format, with a 2–7 volt pp signal in a 110 ohm impedance.

AES-3id and SMPTE276M refer to an electrically unbalanced professional format, with a 1 volt pp signal in a 75 ohm impedance.

S/PDIF and IEC60958-3 refer to an electrically unbalanced consumer format, with a 0.5 volt pp signal in high impedance.

In APx500 series instruments, the digital audio output is unbalanced and is by default set to 0.5 volts pp, a “consumer” setting compatible with S/PDIF and IEC60958-3. If the Professional checkbox is set in the Signal Path Setup view, the level is set to 1 volt, compatible with AES-3id and SMPTE276M. The digital input is unbalanced and by default set to high impedance, compatible with S/PDIF and IEC60958-3. If the 75 Ohm Termination checkbox is set in the Signal Path Setup view, the impedance is set to 75 ohms, compatible with AES-3id and SMPTE276M.

c. optical formats

Digital optical formats have been left to manufacturers to define. Ours is compatible with Toshiba’s Toslink format, as are most digital audio optical inputs and outputs.

digital recording or processing

a technique in which the original signal is periodically sampled and the amplitude value at each sampling instant is converted into a number represented by a binary word.

DIN

Deutsches Institute für Normung, a German standards organization.

dither

dither is low amplitude noise added prior to a signal quantization in order to reduce distortion, improve linearity, and extend the available dynamic range downwards below that of an undithered system of the same number of bits. Level is typically one-half to one LSB in amplitude.

DSP

digital signal processor. A specialized microprocessor designed for highly efficient processing (filtering, FFT, etc.) of digitized analog waveforms. Also, digital signal processing, however implemented.

DUT
a common abbreviation in the test and measurement field for “device under test.”

dynamic range
the difference, usually expressed in dB, between the highest and lowest amplitude portions of a signal, or between the highest amplitude signal which a device can linearly handle and the noise level of the device.

EDID
EDID (Extended Display Identification Data). Used in the HDMI interface, EDID is information stored in downstream display devices and communicated upstream to source devices.

EQ
equalization, or an equalizer.

equalizer
circuitry or equipment which produces a varying (usually adjustable) amplitude as a function of frequency.

EUT
a abbreviation in the test and measurement field for “equipment under test.” Audio Precision uses the term DUT for “device under test.”

even order distortion
Even order distortion products are produced by nonlinearities mathematically described by even value exponents. These nonlinearities produce an asymmetrical shape in the output versus input transfer characteristic of the device. Examples include 2nd harmonic distortion and difference frequency intermodulation distortion.

FFT
Fast Fourier Transform, a technique to compute the amplitude versus frequency and phase versus frequency information from a set of amplitude versus time samples of a signal.

flat
constant gain or attenuation across a frequency band, unfiltered.

frequency domain
a means of representing a signal as a plot of amplitude (normally on the vertical axis) versus frequency (normally on the horizontal axis). Spectrum analyzers represent signals in the frequency domain.

fundamental
the lowest frequency component (normally also the highest amplitude) of a periodic signal.

fundamental rejection
the amount, usually expressed in dB, by which a THD+N analyzer rejects the fundamental component of the input signal. The lowest measurable distortion of a THD+N analyzer is limited by fundamental rejection, along with several other attributes.

gain
gain is the ratio of output signal level to input signal level.

generator waveform file
an audio file on disk that is loaded into the APx generator as the test signal; an arbitrary waveform.

graphic equalizer
an equalizer in which the gain at each portion of the spectrum is controlled by a separate fader. The faders are typically vertically-mounted slide controls, so that the positions of the control knobs forms an approximation to the frequency response of the equalizer.

ground loop
an inadvertent signal path formed when interconnecting the chassis of two or more pieces of equipment, each possessing a safety ground. Ground loops can cause hum-related interference.

group delay
the relative time delay between different spectral portions of a signal.

harmonic
a spectral component at an exact integer multiple of a fundamental frequency.

HDCP
HDCP (High-bandwidth Digital Content Protection) can be used to protect content transmitted on an HDMI interface.

HDMI
High Definition Multimedia Interface, designed to carry high-bandwidth digital streams providing an audio/video interface that includes content protection and a bi-directional channel for interaction with connected electronic devices.

high-pass filter
a filter which passes all frequencies above a specified value essentially without attenuation, while attenuating frequencies below that value.

hum
interference at power mains-related frequencies. Hum directly at the mains frequency and odd harmonics is characteristic of magnetic coupling into the affected system. Hum at the second harmonic of the mains frequency is typically caused by inad-

equate filtering of power supplies with full-wave rectifiers.

IEC
: International Electrotechnical Commission, a body responsible for preparing and publishing international standards for the electrical and electronics fields. The IEC is based in Geneva, Switzerland.

IEC60958
: See digital audio signal, bi-phase coded.

IMD
: intermodulation distortion.

jitter
: the undesirable cycle-to-cycle variation in the period of a reference clock, such as are used in digital audio converters. Jitter can cause modulation sidebands and noise if converters operate from a jittered clock. Excessive jitter in an interface can cause digitally-interfaced equipment to malfunction.

jump discontinuity
: for audio waveforms, a jump discontinuity is a near-instantaneous change in level, usually created by a waveform edit.

level
: the magnitude or amplitude of a signal, which may be expressed in a wide variety of units such as volts, dBm, dBu, watts, etc.

limiting
: the action of a limiter in preventing output levels above a specified value. Also, the action of a conventional amplifier when driven beyond its linear range. In this latter context, “limiting” and “clipping” are sometimes used interchangeably except that clipping may imply a more abrupt characteristic than limiting.

limits testing
: testing in which measurements are compared to acceptable values so that a pass/fail decision may be made. Also called pass / fail testing.

line level
: a relatively high amplitude range suitable for transmission of audio signals. Line level is typically in the 0 dBu to +8 dBu range.

low-pass filter
: a filter which passes all frequencies below a specified frequency essentially without attenuation, while attenuating frequencies above that value.

LSB
: least significant bit. The bit in a binary word representing the smallest possible value change.

matching impedances
: Matching assuring that the impedances of sources and loads connected together in a system are equal, so as to assure maximum power transfer.

notch filter
: a band reject filter with a narrow rejection band, often used to eliminate the fundamental frequency for THD+N measurements or to reject a specific spectral component such as power mains hum or a feedback frequency in a public address system.

Nyquist rate
: 1/2 the sample rate in a digital system, the frequency above which signals cannot be unambiguously coded.

octave
: the interval between a 2:1 range of frequencies, such as 400 Hz to 800 Hz or 5 kHz to 10 kHz.

odd order distortion
: distortion products produced by nonlinearities mathematically described with odd order exponents. These nonlinearities cause a symmetrical shape of the output versus input transfer characteristic of a device. An example would be third harmonic distortion, or the 13 kHz product produced by 14 kHz and 15 kHz signals (2*F1 - F2).

pass/fail
: testing in which measurements are compared to acceptable values so that a pass/fail decision may be made. Also called limits testing.

passband
: the frequency band of a filter in which signals are essentially unattenuated.

PCM
: see Pulse code modulation.

PDM
: see Pulse density modulation.

peak
: the maximum instantaneous excursion of a signal.

peak level
: Peak voltage (Vp) is the maximum instantaneous excursion of a signal. Peak-to-peak voltage (Vpp) is the maximum amplitude difference between positive-going and negative-going peaks of a signal. Peak voltage level is another common level measurement for audio signals. Peak values are typically higher readings than rms values for the same signal.

peak to peak
: the maximum amplitude difference between positive-going and negative-going peaks of a signal.

perceptual coding
low-bit-rate coding of a digital signal according to an understanding of human perception of sound, so that the most (perceptually) important data in the signal are coded with the greatest accuracy.

pink noise
noise whose spectral power distribution is such that the power per octave, per decade, or in any other equal-percentage section is the same anywhere across the spectrum. For example, pink noise has the same power in the octave between 50 Hz and 100 Hz as in the octave between 10 kHz and 20 kHz.

power gain
the ratio of output signal power to input signal power of an amplifier.

PPM
(1) in test and measurement, parts per million (usually in lowercase, ppm). (2) in professional audio and, broadcasting, peak program level meter.

preamplifier
amplifier stage or stages used to bring low-level signals, such as those from microphones and phonograph pickup cartridges, up to a higher, standard level suitable for signal routing, mixing, monitoring, etc.

processor
an amplifier designed to modify characteristics of a signal such as frequency response or dynamic range. Processors in common use in broadcasting and professional audio include equalizers, compressors, limiters, reverberation units, modulation processors, noise reduction units, noise gates, etc.

professional
In audio, often used to designate digital audio format. See digital audio signal, bi-phase coded.

psophometric filter
a filter whose response is based on the frequency response of the human hearing system. Most noise weighting filters are psophometric filters.

pulse code modulation
PCM. A form of data transmission in which amplitude samples of an analog signal are represented by digital numbers.

pulse density modulation
PDM. A system for representing a sampled signal as a stream of single bits using delta sigma conversion (sometimes called sigma delta conversion). PDM is the technology used in many oversampling ADCs and DACs, and is the basis of the Sony/Philips commercial digital format and disc trade-named DSD and SACD, respectively.

pulse width modulation
PWM. A form of data transmission in which amplitude samples of an analog signal are represented by the duty factor of a pulse train. Often used in switching (Class D) amplifiers.

PWM
see Pulse width modulation.

Q factor
a measurement of the selectivity or sharpness of a bandpass or bandreject filter. For a bandpass filter, the Q is equal to the ratio of the center frequency to the bandwidth at the –3dB points. For example, a bandpass filter with a Q of 5, tuned to a center frequency of 1,000 Hz, will have a 3 dB bandwidth of 200 Hz.

random noise
noise whose amplitude-vs.-time distribution is mathematically random and unpredictable, never repeating. The spectrum of a random noise signal is continuous with power at all frequencies, rather than power only at certain points as with pseudo-random noise.

rectangular probability density function dither
RPDF or rectangular dither. Dither which has equal probability of occurrence at any amplitude value between plus one-half LSB and minus one-half LSB deviation from the nominal value. Thus, a graph of probability versus digital value is a rectangle.

residual distortion
the irreducible minimum distortion of an audio generator and analyzer. Residual distortion is thus the “floor” below which the instrument is not useful; measurements above but approaching within 6 to 10 dB of the residual distortion value are less accurate.

residual noise
the irreducible noise in a measurement instrument which sets a floor for amplitude measurements.

resolution
the smallest change in a measured parameter to which a measurement instrument can respond.

ripple
(1) undesired ac variations on a dc power supply output. (2) variations in frequency response in the passband of a filter.

RMS
root mean square, the preferred form of ac signal detection which measures amplitude in terms of its equivalent power content, regardless of signal waveshape.

rms level

Rms (root mean square), is the preferred form of ac signal detection which measures amplitude in terms of its equivalent power content, regardless of signal waveshape.The rms voltage level (Vrms) is the most common level measurement for audio signals, although this level is often stated in a variety of relative units such as dBV or dBu. Rms values are typically lower than peak values for the same signal.

rolloff frequency

the frequency considered the transition between the attenuated and non-attenuated portions of the frequency response of a high-pass or low-pass filter. The filter has a specified attenuation, usually either 3.01 dB or the magnitude of the ripple, at the rolloff frequency.

RPDF

See rectangular probability density function dither.

RSS

root sum square; a method for combining the power of a number of signals by squaring each, summing (adding) these squared values, and finally extracting the square root of the sum.

S/N ratio, Signal-to-noise ratio

the difference in level between a reference output signal (typically at the normal or maximum operating level of the device) and the device output with no signal applied. Signal-to-noise ratio is normally stated in dB. The device input conditions for the noise measurement must be specified, such as “input short-circuited” or with a specific value of resistance connected at the device input instead of a signal.

S/PDIF

See digital audio signal, bi-phase coded.

sample rate, sample frequency

the frequency at which the signal is sampled in a digital system. The sample rate must exceed twice the highest analog frequency to be converted. Commonly-used sample rates are 48 kHz, 44.1 kHz, and 32 kHz.

second order product

distortion products produced by a term with the exponent “2” (i.e., a squaring function) in the device transfer function.

separation

the isolation (usually stated in dB) between channels of a stereo or multichannel device. Crosstalk.

shaped dither

dither in which the power-versus-frequency distribution has been modified to place most of the power at frequencies where the human hearing system is not as sensitive.

shielded cable

a construction technique for cables in which a metallic outer conductor completely surrounds one or more signal-carrying conductors. This outer shield is normally connected to ground (earth) at one or both ends.

short circuit

zero-Ohm connection.

sigma-delta

an A/D converter technique characterized by a relatively low-resolution quantizer operated at very high speeds, digitally low-pass filtered to obtain the final digital signal.

single value

APx500 single value measurement views return one value per channel, e.g. “1.1 Vrms.”

single-ended

unbalanced.

skirt, filter

the portion of response curve of a bandpass, high-pass, or low-pass filter where the attenuation increases at a rapid rate.

slew rate

the rate of change of a signal amplitude, typically expressed in Volts per microsecond.

slew rate limiting

the condition when an amplifier output is unable to follow the rate of change of the input signal.

SMPTE

Society of Motion Picture and Television Engineers. A professional and standards-setting organization.

SMPTE276M

See digital audio signal, bi-phase coded.

S/PDIF

See digital audio signal, bi-phase coded.

SPDIF

Sony Philips Digital Interface; a digital interface for consumer audio equipment. Sometimes also referred to as the EIAJ interface. The SPDIF is similar to the professional AES/EBU interface, but is normally an unbalanced coaxial signal of lower amplitude. Most of the status byte definitions are different between SPDIF and AES/EBU.

spectrum analyzer

a measurement instrument which measures and displays a signal as a frequency domain representation of amplitude (usually vertical scale) versus frequency (horizontal scale). Spectrum analyzers may be designed with a variety of technologies including real-time analyzers (bandpass filter bank), heterodyne or scanning analyzers with a local oscillator, mixer, and selectivity at a fixed intermediate frequency, scanning tunable filters, and FFT analyzers.

status bits

Metadata carried in professional and consumer digital audio bitstreams. See digital audio signal, bi-phase coded.

stepped frequency sweep

In APx500 a stepped frequency sweep generates a sine wave at a series of frequencies, and plots the results against the frequency steps.

stepped level sweep

In APx500 a stepped level sweep generates a sine wave at a series of levels, and plots the results against the level steps.

stopband

the frequency band across which a filter has at least some minimum specified value of attenuation.

sum product

an intermodulation distortion product at a frequency equal to the sum of two input frequencies.

sweep

A sweep shows one parameter as a function of a second parameter. The second parameter is moved (swept) across a range of values. Results are displayed as XY graphs or as a table of values. In audio measurement, many sweeps display level or distortion as a function of swept frequency. Amplifier testing often requires level sweeps, displaying output level or distortion as a function of swept input level.

symmetrical

balanced

terminating

connecting the specified load resistance or impedance to a device.

termination

a specific resistance or impedance value which must be connected to the output or input of a device under test for certain parameters to be measured.

THD

THD stands for Total Harmonic Distortion, a measure of all the harmonic distortion products in the DUT's output, with the fundamental stimulus tone removed, and without consideration of noise. THD measurements are usually made with band-limiting or weighting filters. True THD measurements can be valuable but are rare (although easily done in the Audio Precision APx analyzers), since with most techniques it is difficult to measure the distortion products without also measuring the noise. See THD+N.

THD+N

THD+N stands for Total Harmonic Distortion plus Noise, a measure of everything in the DUT's output, with the fundamental stimulus tone removed. Noise and other interfering signal such as hum, buzz and aliased high-frequencies are reported in the result. THD+N measurements are usually made with band-limiting or weighting filters. THD+N measurements are regarded as a key benchmark of a system's performance; the measurement is easy to perform and widely understood, and it shows at a glance not only distortion performance but also indicates a system's immunity (or lack of it) to noise and other interfering signals.

third octave, 1/3 octave

a bandwidth of 1/3 octave, or a frequency ratio of 1.2599:1. Three successive frequency changes by this ratio result in a total frequency change of 2:1 (one octave).

third order

distortion products produced by a cube (exponent of 3) term in a devices's nonlinear transfer function.

time domain

a means of representing a signal as a graph of amplitude (usually on the vertical axis) versus time (on the horizontal axis). An oscilloscope produces a time domain representation of a signal.

Toslink

See digital audio signal, bi-phase coded.

TPDF

See triangular probability density function dither.

transducer gain

the ratio of output power level that an amplifier will deliver to a load of specified resistance, to the power level that the amplifier's driving source will deliver to a specified resistance equal to the nominal input resistance of the amplifier.

triangular probability density function dither

TPDF dither or triangular dither. Dither in which a graph of the probability of occurrence of an amplitude rises linearly from zero at values plus-or-minus one least significant bit above or below the nominal value, to unity at the nominal value. The graph thus forms a triangle of unity value at the horizontal center (nominal digital value on the horizontal digital value scale), falling to zero at one LSB to the left or right on the horizontal scale.

trigger
: an event which causes another event or action, often initiating a signal generation or acquisition.

unbalanced
: an audio connection in which the desired signal is present as a voltage with respect to ground or common, rather than as a differential signal across a pair of balanced conductors.

voltage gain
: the ratio of output signal voltage to input signal voltage of an amplifier.

VU meter
: a volume unit meter, used to indicate program levels in broadcasting, recording, and similar applications.

weighting filter
: a filter with varying attenuation as a function of frequency so as to produce a measurement where the various spectral components affect the measurement in a specified fashion. Most commonly-used weighting filters are attempts to correspond to the varying response of the human hearing system in order to produce measurements (usually of noise) which correlate well with human observations.

white noise
: noise whose spectral power distribution is such that there is equal power per Hz, anywhere in the spectrum. For example, white noise will have the same power in the 30 Hz bandwidth between 70 Hz and 100 Hz as in the 30 Hz bandwidth between 10,000 Hz and 10,030 Hz.

window
: an amplitude-vs.-time function used to multiply the corresponding samples of a digitized waveform before computing an FFT. Window functions go to zero at the two ends of the record. This provides better selectivity, reducing the signal spreading when non-windowed FFTs are computed on signals which are not exactly synchronous in the signal buffer length being transformed.

word length, word width
: See bit depth.

XLR connector
: a high-quality balanced connector designed for audio applications. Sometimes also called a Cannon connector.

Index

Symbols

A

B

C

D

E

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