Audio Analyzer A2 PRO

A Cost-Effective Solution for R&D and Production Testing

A2 PRO is a dual-channel audio analyzer with optional module slots, integrating analog and digital audio testing. Specifically designed for R&D and production testing, it provides an economical modular solution for electronic audio clients.

Performance Indicators

系统性能	
Residual THD+N (20 kHz BW)	-100dB+2.0μV Typical <-109 dB(1kHZ, 2.5V)

Signal Source Indicators	
Sine Wave Frequency Range	2 Hz to 80.1 kHz
Frequency Accuracy	3 ppm
IMD Test Signals	SMPTE, MOD, DFD
Maximum Output Amplitude (Balanced)	14.40 Vrms
Amplitude Accuracy (1kHz)	±0.05dB
Amplitude Flatness (20Hz-20kHz)	±0.008 dB
Analog Output Configuration	**Balanced & Unbalanced & Common Mode**
Dolby/dts Signal Source	**Yes (Pre-encoded Files)**

Analyzer Indicators	
Maximum Rated Input Voltage	125 Vpk
Maximum Bandwidth	> 90 kHz
IMD Testing Functions	SMPTE, MOD, DFD
Amplitude Accuracy (1kHz)	±0.05 dB
Amplitude Flatness (20Hz—20kHz)	±0.01 dB
Residual Input Noise (20kHz BW)	2.0 μVrms
Independent Harmonic Analysis	d2-d10
Maximum FFT Length	1.2M points
DC Voltage Measurement	Yes

Key Features

- **Fully benchmarked against APx516B audio analyzer from AP**
- **Single module slot for optional BT/HDMI/I2S/PDM/A2B/DIO and other digital interface modules**
- **Typical THD+N of -109dB, ±0.05 dB amplitude accuracy**
- **Comprehensive testing within 3 seconds, no coding required**
- **Supports VB.NET, C#.NET, MATLAB, and complete LabVIEW drivers**
- **Simultaneous analog and digital audio measurements**
- **Supports open-loop measurements**
- **Universal software platform generating various report and image formats for easy sharing**
- **Multi-mode user interface supporting one-click testing, intelligent setup, data graphics judgment, advanced settings, etc.**

Options

Bluetooth R&D Interface Option	AB-BT-DUO
PDM Interface Option	AB-PDM/PDM16
DSIO Interface Option	AB-DSIO
HDMI Interface Option	AB-HDMI2+eARC
Digital Interface Option	AB-DIO
Audio Bus Interface	AB-A2B
Electroacoustic R&D Test Option	AX-SPK-RD
Electroacoustic Production Line Test Option	AX-SPK-PT
Perceptual Audio Test Option	AX-PESQ/POLQA2
Speech Transmission Test Option	AX-STIPA

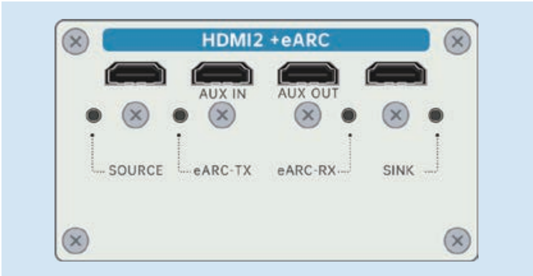
General Specifications

Dimensions (W×D×H)	482mm*434.8mm*95.5mm
Weight	4.9kg±0.2kg
Operating Voltage (AC)	220V,50Hz/100V-240V,50Hz-60Hz

Call to Actions

A10-A2 PRO Digital Input/Output Options

AB-HDMI2+eARC Option



The HDMI2+eARC module allows engineers to measure HDMI audio quality and format compatibility in consumer audio devices, such as surround sound receivers, set-top boxes, streaming media players, TVs, soundbars, Blu-ray disc players, smartphones, and tablets. It provides connections for source and sink devices, along with additional HDMI connections for auxiliary video input and monitor output. Supporting both ARC and eARC, it enhances the ability to test ARC and eARC audio quality and connectivity on compatible receivers, TVs, and streaming hardware.

Main Features

- HDMI 2.1 Source and Sink：Yes
- HDMI 1.4a ARC（Audio Return Channel）：Yes
- HDMI 2.1 eARC（Enhanced Audio Return Channel）：Yes
- HDMI Audio Layout 0 or 1：Yes
- Linear PCM Audio Generation：Up to 8 channels at up to 192kHz (up to 24-bit)
- Lossless Format Generation (e.g., dts-HD Master Audio):48,96 and 192kHz
- Compressed Format Generation (e.g., Dolby Digital, dts Digital Surround)::Yes
- Video Test Signal Generation: HDMI standard resolutions, color depths (including 10-bit), and refresh rates
- HDCP Encryption for both Source and Sink: Fully supported
- Auxiliary interface for third-party video test equipment
- Bit-by-bit verification of digital data reproduction
- Display of audio information boxes and status bits

AB-A2B Option

The AB-A2B interface option is a set of audio interfaces developed based on the emerging A2B audio bus technology, featuring independent Master and Slave interfaces as well as an external power input. Combined with the powerful audio testing software ATC, it efficiently performs various audio tests on A2B components, such as car stereos, car amplifiers, and car microphones.

AB-A2B Option

- Master
  - 8V@500mA bus loading capacity
  - Supports connection of 9 nodes
  - Supports 8-32bit depth signal output (≥ -190dB)
  - Up to 32 channels for upstream
  - Up to 32 channels for downstream
  - Optional external power supply (12V-24V)
- Slave
  - Optional input and output
  - Supports 8-32bit depth signal output (≥ -190dB)
  - Up to 32 channels for upstream
  - Up to 32 channels for downstream
- Sampling Rate: 44100Hz or 48000Hz
- Signal Output
  - 10Hz to 23.9520kHz
  - Accuracy ±0.00002Hz (20Hz ~ 23.9520kHz)
  - Flatness ±0.000001dB (20Hz ~ 23.9520kHz)
- External Interface: Both Master and Slave use Molex HSAutoLink interface

A10-A2 PRO Digital Input/Output Options

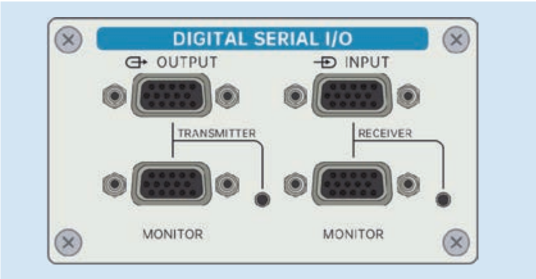
AB-BT-DUO Option

The BT-DUO option is a fully revised and upgraded Bluetooth hardware module with dedicated source and receiver, new Bluetooth chips and the latest firmware, higher RF power, and improved shielding. It offers a range of new audio codecs to meet the diversity of current Bluetooth products, supports new operational functions in profiles, and enables faster Bluetooth connections, allowing engineers to quickly and accurately test Bluetooth products and grasp their device performance.

Main Features

- Bluetooth Device Core Version: V4.2
- Profile Versions：A2DP v1.3， AVRCP v1.4， HFP v1.7，HSP v1.2
- A2DP Audio Codecs：SBC，aptX，aptX-LL， aptX-HD，AAC
- HFP Audio Codecs：CVSD， mSBC
- RF Connection：N-SMA
- RF Input Impedance：50Ω typical
- RF Output Impedance：50Ω typical
- RF Power：Typical maximum +8 dBm
- RF Sensitivity：≤-81 dBm Typical

AB-DSIO Option



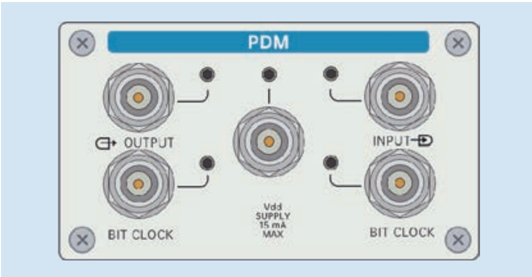
The DSIO option adds multi-channel digital serial interfaces to the audio analyzer, providing direct connections for chip-level interfaces, playing a decisive role in circuit board design and R&D. It is easy to use, provides accurate test results, and can be widely applied to audio quality testing of various digital devices.

Main Features

- Hardware Interface: HD-15 connector
- Channel Numbers：1，2，4，6，8，16
- Formats: I2S, DSP, Left Justified, Right Justified
- Pulse Voltage：1.8V， 2.5V，3.3V
- Data Length：8-32位
- Word Length：8-128位
- Sampling Rate：4kHz-768kHz(1/2channels)；
  4kHz-216kHz(4/6/8channels);
  4kHz-109kHz(TDM16channels)
- Master Clock Rate：4KHz- 56MHz
- Master Clock Source: Transmitter (external or internal), Receiver (external or internal)
- Master Clock: Reversible
- Bit/Clock Direction: Transmitter (IN or OUT), Receiver (IN or OUT)
- Bit Clock Edge Synchronization: Rising or falling
- Jitter: Select ON or OFF
- Multi-channel Configuration: TDM (1/2/4/6/8/16 channels), multiple data connections (1/2/4/6/8 channels)
- Features a simple and flexible configuration interface, allowing saving or loading of test configurations to facilitate user testing and greatly improve test efficiency

A10-A2 PRO Digital Input/Output Options

AB-PDM Option

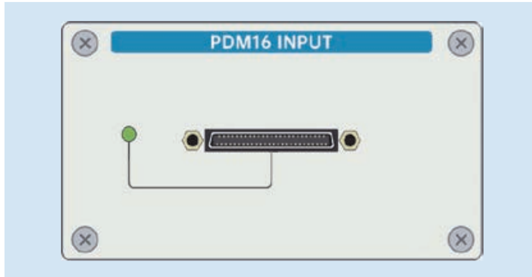


PDM (Pulse Density Modulation) is a 1-bit, high-clock-frequency data stream used in MEMS (Micro-Electro-Mechanical Systems) digital microphones, widely applied in smartphone design. The audio analyzer equipped with the PDM option can directly connect to any device with PDM input or output to achieve comprehensive audio testing required by the mobile telecommunications industry.

Main Features

- Sampling Rate Range：4 kHz ~ 216 kHz
- Clock Range：128 kHz~24.576 MHz
- Interpolation Ratio：×16到×800共33个比率
- Modulator：4阶或5阶
- Interface Logic Level：0.8~3.3V
- Edge Mode: Rising edge/1 channel, Falling edge/1 channel, Stereo (rising and falling edges)/2 channels
- Vdd Output: 0.0~3.6V, maximum 15 mA
- Modulator Maximum Input Level：-0 dBFS
- SNR：127dB(1kHz,20kHzBW，256x oversampling rate, 5th order modulator)
- THD+N：-128 dB (1 kHz，- 9.3 dBFS ，20 kHz BW，256x oversampling rate, 5th order modulator)
- Dynamic Range：137 dB (AES17，CCIR-RMS ，256x oversampling rate, 5th order modulator)
- Smoothness：±0.001 dB (20 Hz ~ 20 kHz)
- Interfaces: Output data, output clock, input data, input clock, external power supply

AB-PDM16 Option



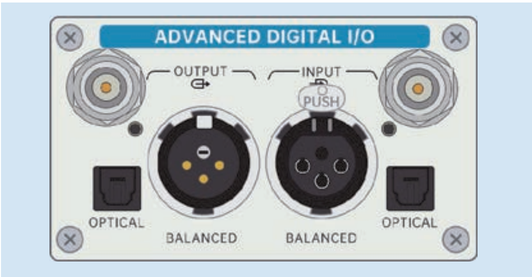
As an optional input module for Ax series audio analyzers, PDM16 consists of an input module installed in the analyzer, a remote interface box, and an extension cable, providing up to 16 channels of sampling-accurate inter-channel phase information, which is critical data for creating MEMS microphone arrays. PDM16 uses synchronous sampling based on a common clock to ensure clear inter-channel phase relationships.

Main Features

- Sampling Rate Range：4 kHz ~ 216 kHz
- Clock Range：128 kHz ~ 24.576MHz
- Oversampling Rate：32, 64, 128, 256
- Edge Mode: Rising edge/odd channels, Falling edge/even channels
- Vdd Output: 0.0-3.6V, maximum 50mA
- Interface Logic Level：0.8-3.3V
- SNR：< 129dB (20 kHz BW, unweighted)
- THD+N：< -130dB (20 kHz BW, unweighted)
- Dynamic Range：< 137dB (AES17, CCIR-RMS)
- Flatness：±0.002dB (20Hz to 20kHz, 32x decimation) ±0.001dB (20Hz to 20kHz, 64x, 128x, 256x, 512x decimation)
- Decimation Ratio: 5 options：32x, 64x, 128x, 256x, 512x
- Inter-channel Phase Alignment: All channels are synchronously sampled from a common clock, fully maintaining phase relationships between channels
- Connector: Connected to DUT via 40-pin IDC 2.5mm (0.1") cable head (on remote interface box)

A10-A2 PRO Digital Input/Output Options

AB-ADIO Option



The advanced digital interface module AB-ADIO provides AES3, AES/EBU balanced digital I/O interfaces on XLR ports, unbalanced SPDIF digital I/O interfaces on BNC ports, and optical digital I/O interfaces on TOSLINK. Additionally, the ADIO module can generate advanced interface impairment signals for complex device testing via AES/SPDIF/TOSLINK. It also includes the advanced master clock module AMC.

Main Features

- Formats: Unbalanced: SPDIF-EIAJ per IEC60958

  Balanced: AES-EBU per AES3-1992

  Optical: TOSLINK® or equivalent
- Sampling Rate: 11.025 kS/s to 200 kS/s (electrical)

  11.025 kS/s to 108 kS/s (optical)
- Sampling Accuracy: ±3 ppm
- Output Amplitude:

  Unbalanced: 0.0 to 2.5 Vpp into 75 Ω
- Balanced:: 0.0 to 8.0 Vpp into 110 Ω

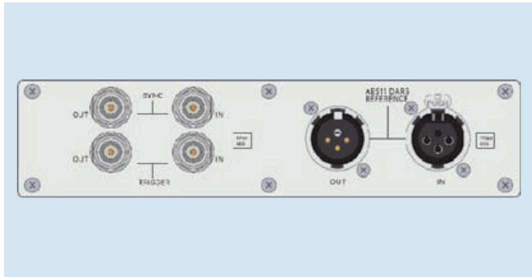
  Amplitude Accuracy: Unbalanced: ±(8% + 20 mV)
- Balanced: ±(10% + 80 mV)

  Input Amplitude:
- Unbalanced: 0 to 2.5 Vpp, ±(5% + 6 mV)

  Balanced: 0 to 8.0 Vpp ±(5% + 25 mV)

  Compatible Instruments: Standard for A10, optional for A5-A9

AB-AMC Option



The advanced master clock module AB-AMC processes input and output clock signals to synchronize the audio analyzer with external devices, while generating and analyzing jitter signals for audio analyzers equipped with ADIO, DSIO, PDM, or HDMI modules.

Main Features

- Synchronous Input

  Signal Types: Square or Sine

  Voltage Range: 0.8 Vpp to 5.0 Vpp (RIN >10 kΩ, AC coupled)

  Frequency Range: 4 kHz to 50 MHz, square; 1 MHz to 50 MHz, sine

  Locking Range: Typically 100 ppm
- Synchronous Output

  Signal: Square

  Amplitude: +0.8 V to +3.6 V, 0.1 V steps

  Frequency Range: 8 kHz to 50 MHz
- Reference Input (AES11/DARS)

  Voltage Range: 2.0 Vpp to 6.0 Vpp

  Sampling Rate Range: 27 kS/s to 216 kS/s

  Locking Range: Typically 100 ppm
- Reference Output (AES11/DARS)

  Amplitude: 5.0 Vpp into 110 Ω, balanced

  Sampling Rate Range: 8 kS/s to 216 kS/s
- Trigger Input

  Voltage Range: -0.5 V to +5.5 V

  Threshold Range: +0.8 to +3.6 V, 0.1 V steps

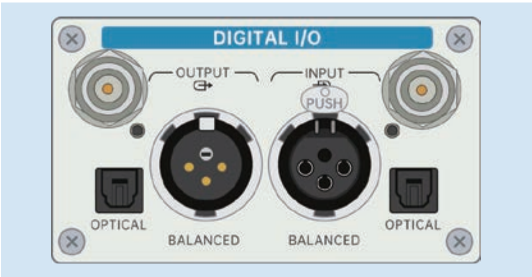
  Minimum Pulse Width: Typically 20 ns
- Trigger Output

  Trigger Source: Analog Sine Generator, Audio Genera-tor, and Jitter Generator

  Amplitude: +0.8 V to +3.6 V, 0.1 V steps
- Compatible Instruments: Standard for A10, optional for A5-A9

A10-A2 PRO Digital Input/Output Options

AB-DIO Option

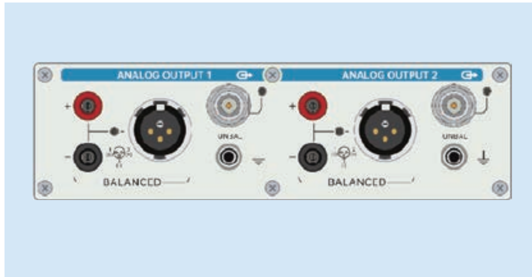


The digital interface module DIO provides balanced digital inputs and outputs compatible with AES3, AES/EBU, and IEC60958-4 on XLR connectors, unbalanced digital inputs and outputs compatible with SPDIF, IEC60958-3, AES3id, and SMPTE 276 M on BNC connectors, as well as optical digital inputs and outputs on Toslink interfaces.

Main Features

- Formats: Unbalanced: SPDIF-EIAJ per IEC60958
  Balanced: AES-EBU per AES3-1992
  Optical: TOSLINK® or equivalent
- Sampling Rate: 16 kS/s to 216 kS/s (electrical)
  16 kS/s to 216 kS/s (optical)
- Sampling Accuracy: ±3 ppm
- Output Amplitude:
  Unbalanced: 0.0 to 2.5 Vpp into 75 Ω
  Balanced:: 0.0 to 8.0 Vpp into 110 Ω
- Amplitude Accuracy:
  Unbalanced: ±(8% + 20 mV)
  Balanced: ±(10% + 80 mV)
- Input Amplitude:
  Unbalanced: 0 to 2.5 Vpp, ±(5% + 6 mV)
  Balanced: 0 to 8.0 Vpp ±(5% + 25 mV)
- Compatible Instruments: Optional for A2 PRO
  standard for other models

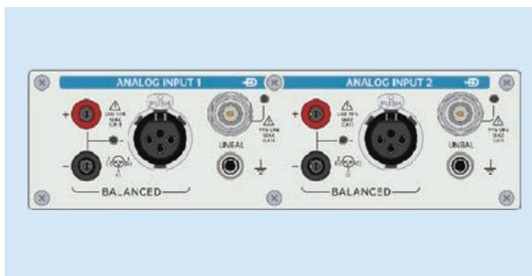
AB-AG52 Option



The high-performance analog signal source option AB-AG52, developed for amplifier designers, generates extremely pure square wave signals with a rise time better than 2 microseconds and improves system THD +N to -110dB (typical). AG52 can generate square wave +sine wave signals as specified in DIM 100, DIM 30, and DIM B, and increases the maximum output level of the balanced interface from 21.21 Vrms to 26.66 Vrms.

*This option is suitable for audio analyzers with analog 2-channel output (A5/A6 options, standard for A7/A10)

AB-BW52 Option



The high-performance analog analyzer option AB-BW52 features ultra-high bandwidth options, providing advanced FFT performance and high-resolution analog input bandwidth of up to 1MHz.

*This option is suitable for audio analyzers with analog 2-channel input (A5/A6 options, standard for A10)

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