

DATA SHEET

TESIRAFORTÉ X 800

MEETING ROOM DSP



TesiraFORTÉ™ X 800 is a meeting room DSP featuring multiple network and analog audio connection points, with 8 channels of Acoustic Echo Cancellation (AEC) assignable across any digital or analog input. Five 1-Gigabit Ethernet ports are provided, four of which are PoE+ powered, and all of which support media and control traffic of various types including AVB, Dante and VoIP.

A USB port also supports 1x1 mono or 2x2 stereo USB audio along with HID synchronization allowing the device to act as a conferencing audio peripheral to systems such as Biamp's Modena family or Unified Communications platforms.

Biamp Launch technology provides the capability for device discovery and tuning to be undertaken without the need for custom programming, and additionally provides the user with a full performance report of the space upon completion.

TesiraFORTÉ X 800 provides extensive audio processing, including but not limited to: AEC technology, signal routing and mixing, equalization, filtering, dynamics, and delay, as well as control, monitoring, and diagnostic tools. Forte X 800 is auto-configurable using Launch but also allows users the option to manually override and completely customize its programming using Tesira software.

FEATURES

- Up to 64x64 AVB Streams of digital audio networking with Biamp AVB devices
- 32 x 32 channels of digital audio networking via the Dante protocol
- AES67-enabled Dante endpoint
- 2 mic/line level inputs, 2 mic/line level outputs
- Five 1 Gigabit Ethernet Ports
- Four ports support PoE+ power (IEEE 802.3.at Class 4, 30W)
- Up to 2x2 channels of configurable USB audio
- 8 AEC channels assignable to any input
- 4-pin GPIO
- Surface mountable with included bracket
- Supports port authentication via IEEE 802.1X
- SIP VoIP interface via Gigabit Ethernet connection
- CE marked, UL listed, and RoHS compliant
- Covered by Biamp Systems' 5-year warranty

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ARCHITECTS & ENGINEERS SPECIFICATION

The Conference Room DSP shall support Ethernet connection for programming and control on a RJ-45 connector. The Conference Room DSP shall have internal DSP processing. The Conference Room DSP shall include 4 channels of General Purpose Input and Output connection (GPIO) for sending or receiving logic signals. The programming of the GPIO ports shall be software configurable. The Conference Room DSP shall include a Universal Serial Bus (USB) connection on a standard USB-B type connector. The Conference Room DSP shall be software configurable to stream up to 2 channels of digital USB Class 1 Audio transmission either into or out of the Conference Room DSP or simultaneous input and output. The Conference Room DSP shall support port authentication via IEEE 802.1X. The Conference Room DSP shall provide 2 balanced input connections for receiving of microphone or line level analog audio signals on screw-down, removable connectors. Any network audio or analog audio connection may be assigned one of eight channels of Acoustic Echo Cancellation (AEC). Acoustic Echo Cancellation (AEC) hardware and firmware, the parameters, routing and operation of microphone or line level analog audio signals on screw-down, removable connectors. Each individual channel shall have its own dedicated connection. The Conference Room DSP shall integrate to Voice Over Internet Protocol (VoIP) systems on a RJ-45 connector and shall support Session Initiation Protocol (SIP) v2.0 or later.

The Conference Room DSP shall be capable of being deployed with zero programming or manual tuning and shall provide a post-commissioning status report via the use of Biamp Launch technology.

The Conference Room DSP shall feature software-configurable signal processing, including but not limited to: signal routing and mixing, equalization, filtering, dynamics, and delay, as well as control, monitoring, and diagnostic tools.

The Conference Room DSP shall provide front panel LED identification of device power, status, alarm, and activity as well as system-wide alarm. The Conference Room DSP shall be surface mountable using the included mounting hardware. The fixed I/O DSP shall be CE marked, UL listed, and shall be compliant with the RoHS directive. Warranty shall be five years. The Conference Room DSP shall be TesiraFORTÉ X 800.

TESIRAFORTÉ X 800 SPECIFICATIONS

Frequency Response: 20Hz to 20kHz, +4dBu output:	+0.25 dB/-0.5 dB	Crosstalk, channel to channel, 1 kHz: 0dB gain, +4dBu input:	< -85dB
THD+N (22Hz to 22kHz): 0dB gain, +4dBu input:	< 0.006%	54dB gain, -50dBu input:	< -75dB
EIN (no weighting, 22Hz to 22kHz):	< -125dBu	Sampling Rate:	48kHz
Dynamic Range (in presence of signal) 22Hz to 22kHz, 0dB gain:	> 108dB	A/D - D/A Converters:	24-bit
Input Impedance (balanced):	8kΩ	Power Consumption: 100-240VAC 50/60Hz:	< 150W
Output Impedance (balanced):	207Ω	USB: Bit Depth:	24-bit
Maximum Input:	+24dBu	Number of Channels:	up to 2x2
Maximum Output (selectable): +24dBu, +18dBu, +12dBu, +6dBu, 0dBu, -31dBu		Sample Rate:	48kHz
Maximum Number of AVB Channels:	128x128	Environment: Ambient Operating Temperature Range:	32-104° F (0-40° C)
Maximum Number of AVB Streams:	64x64	Humidity:	0-98%, non-condensing
Maximum AVB Stream Passthrough:	150	Altitude:	0-6,600 feet (0-2000 Meters) MSL
Maximum Number of Dante Channels:	32x32	Compliance:	FCC Part 15B (USA) Canada ICES-003 (A) / NMB-003 (A) CE marked (Europe) UL und C-UL listed (USA and Canada) RCM (Australia) RoHS Directive (Europe)
Maximum Number of Dante Flows:	32x32		
Input Gain Range (6dB steps):	0-66dB		
Overall Dimensions: Height:	1.47 inches (37.3 mm)		
Width:	8.11 inches (206 mm)		
Depth:	8.11 inches (206 mm)		
Weight:	1.9 lbs (0.86kg)		
Phantom Power:	+48VDC (7mA/input)		

TESIRAFORTÉ X 800 BACK PANEL



OPTIONAL ACCESSORIES

Accessory Pack