DATA SHEET
TESIRAFORTÉ® AI FIXED AUDIO DSP

The TesiraFORTÉ® AI is a fixed audio DSP with 12 analog inputs and 8 analog outputs and includes up to 8 channels of configurable USB audio. USB audio allows TesiraFORTÉ to interface directly with USB audio hosts, as well as to take full advantage of today's most sophisticated conferencing solutions. TesiraFORTÉ AI also provides extensive audio processing, including but not limited to: signal routing and mixing, equalization, filtering, dynamics, and delay, as well as control, monitoring, and diagnostic tools; all configured through the Tesira configuration software. TesiraFORTÉ AI is best-suited for small- to medium-sized rooms that require high-quality audio solutions using voice lift and mix-minus, such as conference rooms or council chambers.

BENEFITS

- Includes default configuration file, allowing for plug-and-play usage
- Highly scalable and cost-effective solution that can grow over time with the needs of the customer
- SpeechSense™ technology enhances speech processing
- Integrates directly with soft codecs and other USB audio hosts

FEATURES

- 12 mic/line level inputs, 8 mic/line level outputs
- Gigabit Ethernet port
- Up to 8 channels of configurable USB audio
- RS-232 serial port
- 4-pin GPIO
- 2-line OLED display with capacitive-touch navigation
- Rack mountable (1RU)
- System configuration and control via Ethernet
- Internal universal power supply
- Signal processing via intuitive software allows configuration and control for signal routing, mixing, equalization, filtering, delay and much more
- CE marked, UL listed, and RoHS compliant
- Covered by Biamp Systems' 5-year warranty

ARCHITECTS & ENGINEERS SPECIFICATION

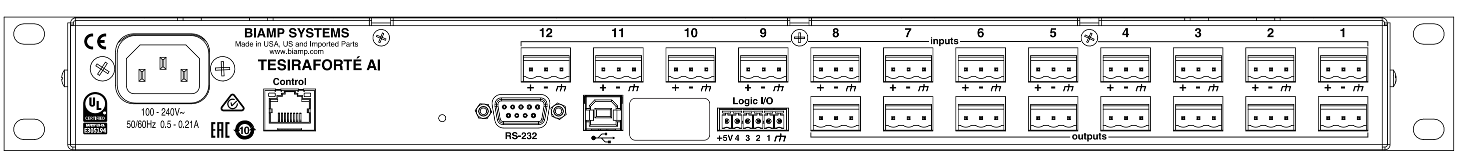
The fixed audio DSP shall be designed exclusively for use with Tesira® systems. The audio DSP shall support Ethernet connection for programming and control on a RJ-45 connector. The audio DSP shall have internal DSP processing. The audio DSP shall include 4 channels of General Purpose Input and Output connection (GPIO) for sending or receiving logic signals. The programming of the GPIO ports shall be software configurable. The audio DSP shall include a RS-232 connection for control data transmission into or out of the audio DSP and such operation shall be software programmable. The audio DSP shall include a Universal Serial Bus (USB) connection on a standard USB-B type connector. The audio DSP shall be software configurable to stream up to 8 channels of digital USB Class 1 Audio transmission either into or out of the audio DSP or simultaneous input and output. The audio DSP shall provide 12 balanced input connections for receiving of microphone or line level analog audio signals on screw-down, removable connectors. The audio DSP shall provide 8 balanced output channels for the transmission of microphone or line level analog audio signals on screw-down, removable connectors. Each individual channel shall have its own dedicated connection. The audio DSP shall provide front panel OLED identification of device power, status, alarm, and activity as well as system-wide alarm. The audio DSP shall be rack mountable (1RU) and 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 audio DSP shall be CE marked, UL listed, and shall be compliant with the RoHS directive. Warranty shall be five years. The fixed audio DSP shall be TesiraFORTÉ AI.

TESIRAFORTÉ AI SPECIFICATIONS

Frequency Response:	
20Hz to 20kHz, +4dBu output:	+0.25 dB/-0.5 dB
THD+N (22Hz to 22kHz):	
0dB gain, +4dBu input:	< 0.006%
54dB gain, -50dBu input:	< 0.040%
EIN (no weighting, 22Hz to 22kHz):	< -125dBu
Dynamic Range (in presence of signal)	
22Hz to 22kHz, 0dB gain:	> 108dB
Input Impedance (balanced):	8kΩ
Output Impedance (balanced):	207Ω
Maximum Input:	+24dBu
Maximum Output (selectable):	+24dBu, +18dBu, +12dBu, +6dBu, 0dBu, -31dBu
Input Gain Range (6dB steps):	0-66dB
Overall Dimensions:	
Height:	1.75 inches (44 mm)
Width:	19.0 inches (483 mm)
Depth:	10.5 inches (267 mm)
Weight:	8 lbs (3.63 kg)
Environment:	
Ambient Operating Temperature Range:	32-104° F (0–40° C)
Humidity:	0–98%, non-condensing
Altitude:	0-6,600 feet (0–2000 Meters) MSL

Phantom Power:	+48VDC (7mA/input)
Crosstalk, channel to channel, 1 kHz:	
0dB gain, +4dBu input:	< -85dB
54dB gain, -50dBu input:	< -75dB
Sampling Rate:	48kHz
A/D - D/A Converters:	24-bit
Power Consumption:	
100-240VAC 50/60Hz:	< 35W
USB:	
Bit Depth:	16- or 24-bit
Number of Channels:	up to 8
Sample Rate:	48kHz
Compliance:	FCC Part 15B (USA) CE marked (Europe) UL und C-UL listed (USA and Canada) RCM (Australia) RoHS Directive (Europe)

TESIRAFORTÉ AI BACK PANEL



DATA SHEET
TESIRAFORTÉ® AVB AI FIXED AUDIO DSP

The TesiraFORTÉ® AVB AI is a fixed audio DSP with 12 analog inputs and 8 analog outputs and includes up to 8 channels of configurable USB audio. USB audio allows TesiraFORTÉ to interface directly with USB audio hosts, as well as to take full advantage of today's most sophisticated conferencing solutions. TesiraFORTÉ AVB AI utilizes Audio Video Bridging (AVB) for digital audio networking, and can be used as a standalone device or combined with other TesiraFORTÉ devices and Tesira servers, expanders, and controllers. TesiraFORTÉ AVB AI also provides extensive audio processing, including but not limited to: signal routing and mixing, equalization, filtering, dynamics, and delay, as well as control, monitoring, and diagnostic tools; all configured through the Tesira configuration software. TesiraFORTÉ AVB AI is best-suited for small- to medium-sized rooms that require high-quality audio solutions using voice lift and mix-minus, such as conference rooms or council chambers.

BENEFITS

- AVB allows audio networking via IEEE open standards protocol
- Includes default configuration file, allowing for plug-and-play usage
- Highly scalable and cost-effective solution that can grow over time with the needs of the customer
- SpeechSense™ technology enhances speech processing
- Integrates directly with soft codecs and other USB audio hosts

FEATURES

- 128 x 128 channels of AVB
- 12 mic/line level inputs, 8 mic/line level outputs
- Gigabit Ethernet port
- Up to 8 channels of configurable USB audio
- RS-232 serial port
- 4-pin GPIO
- 2-line OLED display with capacitive-touch navigation
- Rack mountable (1RU)
- System configuration and control via Ethernet
- Internal universal power supply
- Fully compatible with Tesira AVB servers, endpoints, expanders, and controllers
- Signal processing via intuitive software allows configuration and control for signal routing, mixing, equalization, filtering, delay and much more
- CE marked, UL listed, and RoHS compliant
- Covered by Biamp Systems' 5-year warranty

ARCHITECTS & ENGINEERS SPECIFICATION

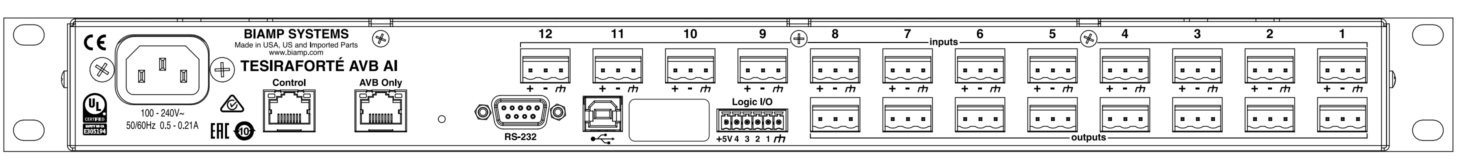
The fixed audio DSP shall be designed exclusively for use with Tesira® systems. The audio DSP shall support Audio Video Bridging (AVB) digital audio networking that shall allow up to 128 x 128 channels. The AVB networking connection shall be implemented on a RJ-45 connector. The audio DSP shall support Ethernet connection for programming and control on a RJ-45 connector. The audio DSP shall have internal DSP processing. The audio 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 audio DSP shall include a RS-232 connection for control data transmission into or out of the audio DSP and such operation shall be software programmable. The audio DSP shall include a Universal Serial Bus (USB) connection on a standard USB-B type connector. The audio DSP shall be software configurable to stream up to 8 channels of digital USB Class 1 Audio transmission either into or out of the audio DSP or simultaneous input and output. The audio DSP shall provide 12 balanced input connections for receiving of microphone or line level analog audio signals on screw-down, removable connectors. The audio DSP shall provide 8 balanced output channels for the transmission of microphone or line level analog audio signals on screw-down, removable connectors. Each individual channel shall have its own dedicated connection. The audio DSP shall provide front panel OLED identification of device power, status, alarm, and activity as well as system-wide alarm. The audio DSP shall be rack mountable (1RU) and 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 audio DSP shall control and proxy all Tesira expander-class devices and Tesira control devices. The audio DSP shall be CE marked, UL listed, and shall be compliant with the RoHS directive. Warranty shall be five years. The fixed audio DSP shall be TesiraFORTÉ AVB AI.

TESIRAFORTÉ AVB AI SPECIFICATIONS

Frequency Response:	
20Hz to 20kHz, +4dBu output:	+0.25 dB/-0.5 dB
THD+N (22Hz to 22kHz):	
0dB gain, +4dBu input:	< 0.006%
54dB gain, -50dBu input:	< 0.040%
EIN (no weighting, 22Hz to 22kHz):	< -125dBu
Dynamic Range (in presence of signal)	
22Hz to 22kHz, 0dB gain:	> 108dB
Input Impedance (balanced):	8kΩ
Output Impedance (balanced):	207Ω
Maximum Input:	+24dBu
Maximum Output (selectable):	+24dBu, +18dBu, +12dBu, +6dBu, 0dBu, -31dBu
Input Gain Range (6dB steps):	0-66dB
Overall Dimensions:	
Height:	1.75 inches (44 mm)
Width:	19.0 inches (483 mm)
Depth:	10.5 inches (267 mm)
Weight:	8 lbs (3.63 kg)
Environment:	
Ambient Operating Temperature Range:	32-104° F (0–40° C)
Humidity:	0–98%, non-condensing
Altitude:	0-6,600 feet (0–2000 Meters) MSL

Phantom Power:	+48VDC (7mA/input)
Crosstalk, channel to channel, 1 kHz:	
0dB gain, +4dBu input:	< -85dB
54dB gain, -50dBu input:	< -75dB
Sampling Rate:	48kHz
A/D - D/A Converters:	24-bit
Power Consumption:	
100-240VAC 50/60Hz:	< 35W
USB:	
Bit Depth:	16- or 24-bit
Number of Channels:	up to 8
Sample Rate:	48kHz
Compliance:	FCC Part 15B (USA) CE marked (Europe) UL und C-UL listed (USA and Canada) RCM (Australia) RoHS Directive (Europe)

TESIRAFORTÉ AVB AI BACK PANEL



biamp.

A: 9300 S.W. Gemini Drive Beaverton, OR 97008 USA T: +1 503.641.7287 W: www.biamp.com

DATA SHEET
TESIRAFORTÉ® DAN AI FIXED AUDIO DSP

The TesiraFORTÉ® DAN AI is a fixed audio DSP with 32 bi-directional channels of Dante™ digital audio, 12 analog inputs, 8 analog outputs, and includes up to 8 channels of configurable USB audio. USB audio allows TesiraFORTÉ to interface directly with USB audio hosts, as well as to take full advantage of today's most sophisticated conferencing solutions. TesiraFORTÉ DAN AI provides extensive audio processing, including but not limited to: signal routing and mixing, equalization, filtering, dynamics, and delay; as well as control, monitoring, and diagnostic tools; all configured through the Tesira software. TesiraFORTÉ DAN AI is best-suited for small- to medium-sized rooms that require high-quality audio solutions using voice lift and mix-minus, such as conference rooms or council chambers.

BENEFITS

- Includes default configuration file, allowing for plug-and-play usage
- Highly scalable and cost-effective solution that can grow over time with the needs of the customer
- SpeechSense™ technology enhances speech processing
- Integrates directly with soft codecs and other USB audio hosts

FEATURES

- 32 x 32 channels of digital audio networking via the Dante protocol
- 12 mic/line level inputs, 8 mic/line level outputs
- 2 Gigabit Ethernet ports: Dante digital audio and Tesira control
- Up to 8 channels of configurable USB audio
- RS-232 serial port
- 4-pin GPIO
- 2-line OLED display with capacitive-touch navigation
- Rack mountable (1RU)
- System configuration and control via Ethernet
- Internal universal power supply
- Signal processing via intuitive software allows configuration and control for signal routing, mixing, equalization, filtering, delay and much more
- CE marked, UL listed, and RoHS compliant
- Covered by Biamp Systems' 5-year warranty

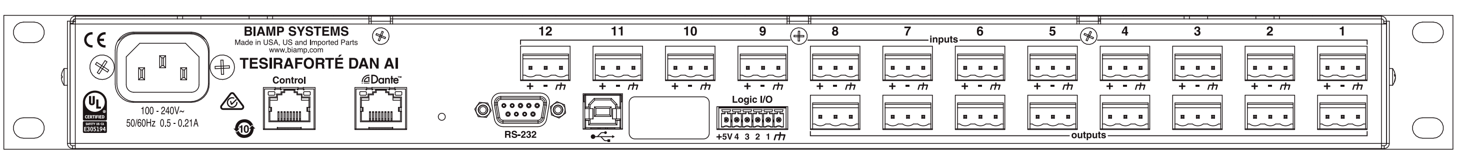
ARCHITECTS & ENGINEERS SPECIFICATION

The fixed audio DSP shall be designed exclusively for use with Tesira® systems. The audio DSP shall support Dante™ digital audio networking that shall allow up to 32 x 32 channels. The Dante networking connection shall be implemented on a RJ-45 connector. The audio DSP shall support Ethernet connection for programming and control on a RJ-45 connector. The audio DSP shall have internal DSP processing. The audio 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 audio DSP shall include a RS-232 connection for control data transmission into or out of the audio DSP and such operation shall be software programmable. The audio DSP shall include a Universal Serial Bus (USB) connection on a standard USB-B type connector. The audio DSP shall be software configurable to stream up to 8 channels of digital USB Class 1 Audio transmission either into or out of the audio DSP or simultaneous input and output. The audio DSP shall provide 12 balanced input connections for receiving of microphone or line level analog audio signals on screw-down, removable connectors. The audio DSP shall provide 8 balanced output channels for the transmission of microphone or line level analog audio signals on screw-down, removable connectors. Each individual channel shall have its own dedicated connection. The audio DSP shall provide front panel OLED identification of device power, status, alarm, and activity as well as system-wide alarm. The audio DSP shall be rack mountable (1RU) and 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 audio DSP shall be CE marked, UL listed, and shall be compliant with the RoHS directive. Warranty shall be five years. The fixed audio DSP shall be TesiraFORTÉ DAN AI.

TESIRAFORTÉ DAN AI SPECIFICATIONS

Frequency Response:	
20Hz to 20kHz, +4dBu output:	+0.25 dB/-0.5 dB
THD+N (22Hz to 22kHz):	
0dB gain, +4dBu input:	< 0.006%
54dB gain, -50dBu input:	< 0.040%
EIN (no weighting, 22Hz to 22kHz):	< -125dBu
Dynamic Range (in presence of signal)	
22Hz to 22kHz, 0dB gain:	> 108dB
Input Impedance (balanced):	8kΩ
Output Impedance (balanced):	207Ω
Maximum Input:	+24dBu
Maximum Output (selectable):	+24dBu, +18dBu, +12dBu, +6dBu, 0dBu, -31dBu
Input Gain Range (6dB steps):	0-66dB
Overall Dimensions:	
Height:	1.75 inches (44 mm)
Width:	19.0 inches (483 mm)
Depth:	10.5 inches (267 mm)
Weight:	8 lbs (3.63 kg)
Environment:	
Ambient Operating Temperature Range:	32-104° F (0–40° C)
Humidity:	0–98%, non-condensing
Altitude:	0-6,600 feet (0–2000 Meters) MSL
Phantom Power:	+48VDC (7mA/input)
Crosstalk, channel to channel, 1 kHz:	
0dB gain, +4dBu input:	< -85dB
54dB gain, -50dBu input:	< -75dB
Sampling Rate:	48kHz
A/D - D/A Converters:	24-bit
Power Consumption:	
100-240VAC 50/60Hz:	< 35W
USB:	
Bit Depth:	16- or 24-bit
Number of Channels:	up to 8
Sample Rate:	48kHz
Compliance:	FCC Part 15B (USA) CE marked (Europe) UL und C-UL listed (USA and Canada) RCM (Australia) RoHS Directive (Europe)

TESIRAFORTÉ DAN AI BACK PANEL



DATA SHEET
TESIRAFORTÉ® CI FIXED AUDIO DSP

The TesiraFORTÉ® CI is a fixed audio DSP with 12 analog inputs and 8 analog outputs and includes Acoustic Echo Cancellation (AEC) technology on all 12 inputs. It also includes up to 8 channels of configurable USB audio. USB audio allows TesiraFORTÉ to interface directly with USB audio hosts, as well as to take full advantage of today's most sophisticated conferencing solutions. TesiraFORTÉ CI also 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; all configured through the Tesira configuration software. TesiraFORTÉ CI is best suited for small- to medium-sized rooms that require high-quality audio solutions using AEC, voice lift, and mix-minus, such as conference rooms or distance learning environments.

BENEFITS

- Includes default configuration file, allowing for plug-and-play usage
- Highly scalable and cost-effective solution that can grow over time with the needs of the customer
- Acoustic Echo Cancellation (AEC) technology on all 12 inputs
- SpeechSense™ technology enhances speech processing
- Integrates directly with soft codecs and other USB audio hosts

FEATURES

- 12 mic/line level inputs with AEC, 8 mic/line level outputs
- Gigabit Ethernet port
- Up to 8 channels of configurable USB audio
- RS-232 serial port
- 4-pin GPIO
- 2-line OLED display with capacitive-touch navigation
- Rack mountable (1RU)
- System configuration and control via Ethernet
- Internal universal power supply
- Signal processing via intuitive software allows configuration and control for signal routing, mixing, equalization, filtering, delay and much more
- CE marked, UL listed, and RoHS compliant
- Covered by Biamp Systems' 5-year warranty

A: 9300 S.W. Gemini Drive Beaverton, OR 97008 USA T: +1 503.641.7287 W: www.biamp.com

ARCHITECTS & ENGINEERS SPECIFICATION

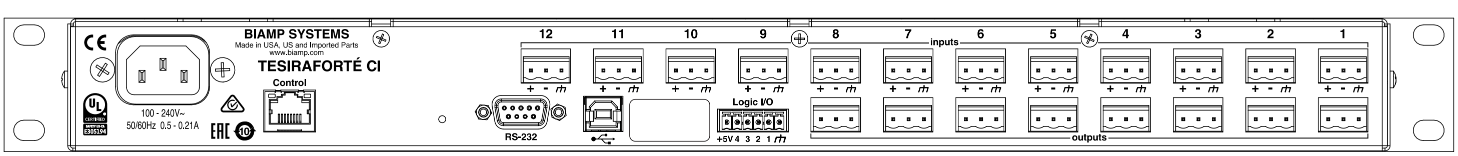
The fixed audio DSP shall be designed exclusively for use with Tesira® systems. The audio DSP shall support Ethernet connection for programming and control on a RJ-45 connector. The audio DSP shall have internal DSP processing. The audio DSP shall include 4 channels of General Purpose Input and Output connection (GPIO) for sending or receiving logic signals. The programming of the GPIO ports shall be software configurable. The audio DSP shall include a RS-232 connection for control data transmission into or out of the audio DSP and such operation shall be software programmable. The audio DSP shall include a Universal Serial Bus (USB) connection on a standard USB-B type connector. The audio DSP shall be software configurable to stream up to 8 channels of digital USB Class 1 Audio transmission either into or out of the audio DSP or simultaneous input and output. The audio DSP shall provide 12 balanced input connections for receiving of microphone or line level analog audio signals on screw-down, removable connectors. The input connections shall include Acoustic Echo Cancellation (AEC) hardware and firmware, the parameters, routing and operation of which shall be software programmable. The audio DSP shall provide 8 balanced output channels for the transmission of microphone or line level analog audio signals on screw-down, removable connectors. Each individual channel shall have its own dedicated connection. The audio DSP shall provide front panel OLED identification of device power, status, alarm, and activity as well as system-wide alarm. The audio DSP shall be rack mountable (1RU) and 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 audio DSP shall be CE marked, UL listed, and shall be compliant with the RoHS directive. Warranty shall be five years. The fixed audio DSP shall be TesiraFORTÉ CI.

TESIRAFORTÉ CI SPECIFICATIONS

Frequency Response:	
20Hz to 20kHz, +4dBu output:	+0.25 dB/-0.5 dB
THD+N (22Hz to 22kHz):	
0dB gain, +4dBu input:	< 0.006%
54dB gain, -50dBu input:	< 0.040%
EIN (no weighting, 22Hz to 22kHz):	< -125dBu
Dynamic Range (in presence of signal)	
22Hz to 22kHz, 0dB gain:	> 108dB
Input Impedance (balanced):	8kΩ
Output Impedance (balanced):	207Ω
Maximum Input:	+24dBu
Maximum Output (selectable):	+24dBu, +18dBu, +12dBu, +6dBu, 0dBu, -31dBu
Input Gain Range (6dB steps):	0-66dB
Overall Dimensions:	
Height:	1.75 inches (44 mm)
Width:	19.0 inches (483 mm)
Depth:	10.5 inches (267 mm)
Weight:	8 lbs (3.63 kg)
Environment:	
Ambient Operating Temperature Range:	32-104° F (0–40° C)
Humidity:	0–98%, non-condensing
Altitude:	0-6,600 feet (0–2000 Meters) MSL

Phantom Power:	+48VDC (7mA/input)
Crosstalk, channel to channel, 1 kHz:	
0dB gain, +4dBu input:	< -85dB
54dB gain, -50dBu input:	< -75dB
Sampling Rate:	48kHz
A/D - D/A Converters:	24-bit
Power Consumption:	
100-240VAC 50/60Hz:	< 35W
USB:	
Bit Depth:	16- or 24-bit
Number of Channels:	up to 8
Sample Rate:	48kHz
Compliance:	FCC Part 15B (USA) CE marked (Europe) UL und C-UL listed (USA and Canada) RCM (Australia) RoHS Directive (Europe)

TESIRAFORTÉ CI BACK PANEL



DATA SHEET
TESIRAFORTÉ® AVB CI FIXED AUDIO DSP

The TesiraFORTÉ® AVB CI is a fixed audio DSP with 12 analog inputs and 8 analog outputs and includes Acoustic Echo Cancellation (AEC) technology on all 12 inputs. It also includes up to 8 channels of configurable USB audio. USB audio allows TesiraFORTÉ to interface directly with USB audio hosts, as well as to take full advantage of today's most sophisticated conferencing solutions. TesiraFORTÉ AVB CI utilizes Audio Video Bridging (AVB) for digital audio networking, and can be used as a standalone device or combined with other TesiraFORTÉ devices and Tesira servers, expanders, and controllers. TesiraFORTÉ AVB CI also 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; all configured through the Tesira configuration software. TesiraFORTÉ AVB CI is best suited for small- to medium-sized rooms that require high-quality audio solutions using AEC, voice lift, and mix-minus, such as conference rooms or distance learning environments.

BENEFITS

- AVB allows audio networking via IEEE open standards protocol
- Includes default configuration file, allowing for plug-and-play usage
- Highly scalable and cost-effective solution that can grow over time with the needs of the customer
- Acoustic Echo Cancellation (AEC) technology on all 12 inputs
- SpeechSense™ technology enhances speech processing
- Integrates directly with soft codecs and other USB audio hosts

FEATURES

- 128 x 128 channels of AVB
- 12 mic/line level inputs with AEC, 8 mic/line level outputs
- Gigabit Ethernet port
- Up to 8 channels of configurable USB audio
- RS-232 serial port
- 4-pin GPIO
- 2-line OLED display with capacitive-touch navigation
- Rack mountable (1RU)
- System configuration and control via Ethernet
- Internal universal power supply
- Fully compatible with Tesira AVB servers, endpoints, expanders, and controllers
- Signal processing via intuitive software allows configuration and control for signal routing, mixing, equalization, filtering, delay and much more
- CE marked, UL listed, and RoHS compliant
- Covered by Biamp Systems' 5-year warranty

ARCHITECTS & ENGINEERS SPECIFICATION

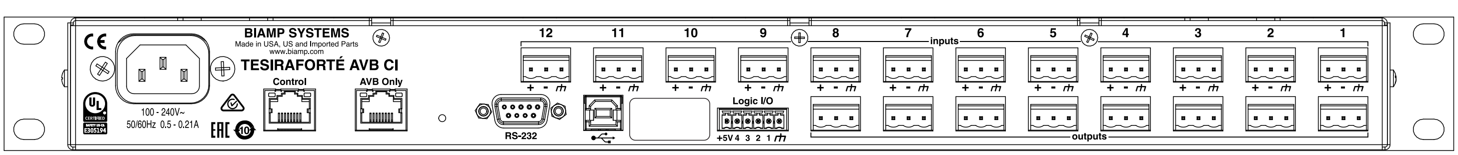
The fixed audio DSP shall be designed exclusively for use with Tesira® systems. The audio DSP shall support Audio Video Bridging (AVB) digital audio networking that shall allow up to 128 x 128 channels. The AVB networking connection shall be implemented on a RJ-45 connector. The audio DSP shall support Ethernet connection for programming and control on a RJ-45 connector. The audio DSP shall have internal DSP processing. The audio 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 audio DSP shall include a RS-232 connection for control data transmission into or out of the audio DSP and such operation shall be software programmable. The audio DSP shall include a Universal Serial Bus (USB) connection on a standard USB-B type connector. The audio DSP shall be software configurable to stream up to 8 channels of digital USB Class 1 Audio transmission either into or out of the audio DSP or simultaneous input and output. The audio DSP shall provide 12 balanced input connections for receiving of microphone or line level analog audio signals on screw-down, removable connectors. The input connections shall include Acoustic Echo Cancellation (AEC) hardware and firmware, the parameters, routing and operation of which shall be software programmable. The audio DSP shall provide 8 balanced output channels for the transmission of microphone or line level analog audio signals on screw-down, removable connectors. Each individual channel shall have its own dedicated connection. The audio DSP shall provide front panel OLED identification of device power, status, alarm, and activity as well as system-wide alarm. The audio DSP shall be rack mountable (1RU) and 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 audio DSP shall control and proxy all Tesira expander-class devices and Tesira control devices. The audio DSP shall be CE marked, UL listed, and shall be compliant with the RoHS directive. Warranty shall be five years. The fixed audio DSP shall be TesiraFORTÉ AVB CI.

TESIRAFORTÉ AVB CI SPECIFICATIONS

Frequency Response:	
20Hz to 20kHz, +4dBu output:	+0.25 dB/-0.5 dB
THD+N (22Hz to 22kHz):	
0dB gain, +4dBu input:	< 0.006%
54dB gain, -50dBu input:	< 0.040%
EIN (no weighting, 22Hz to 22kHz):	< -125dBu
Dynamic Range (in presence of signal)	
22Hz to 22kHz, 0dB gain:	> 108dB
Input Impedance (balanced):	8kΩ
Output Impedance (balanced):	207Ω
Maximum Input:	+24dBu
Maximum Output (selectable):	+24dBu, +18dBu, +12dBu, +6dBu, 0dBu, -31dBu
Input Gain Range (6dB steps):	0-66dB
Overall Dimensions:	
Height:	1.75 inches (44 mm)
Width:	19.0 inches (483 mm)
Depth:	10.5 inches (267 mm)
Weight:	8 lbs (3.63 kg)
Environment:	
Ambient Operating Temperature Range:	32-104° F (0–40° C)
Humidity:	0–98%, non-condensing
Altitude:	0-6,600 feet (0–2000 Meters) MSL

Phantom Power:	+48VDC (7mA/input)
Crosstalk, channel to channel, 1 kHz:	
0dB gain, +4dBu input:	< -85dB
54dB gain, -50dBu input:	< -75dB
Sampling Rate:	48kHz
A/D - D/A Converters:	24-bit
Power Consumption:	
100-240VAC 50/60Hz:	< 35W
USB:	
Bit Depth:	16- or 24-bit
Number of Channels:	up to 8
Sample Rate:	48kHz
Compliance:	FCC Part 15B (USA) CE marked (Europe) UL und C-UL listed (USA and Canada) RCM (Australia) RoHS Directive (Europe)

TESIRAFORTÉ AVB CI BACK PANEL



DATA SHEET
TESIRAFORTÉ® DAN CI FIXED AUDIO DSP

The TesiraFORTÉ DAN CI is a fixed audio DSP with 32 bi-directional channels of Dante™ digital audio, 12 analog inputs with Acoustic Echo Cancellation (AEC), and 8 analog outputs. It also includes up to 8 channels of configurable USB audio. USB audio allows TesiraFORTÉ to interface directly with USB audio hosts, as well as to take advantage of modern conferencing solutions. TesiraFORTÉ DAN CI provides extensive audio processing, including but not limited to: signal routing and mixing, equalization, filtering, dynamics, and delay; as well as control, monitoring, and diagnostic tools; all configured through the Tesira configuration software. TesiraFORTÉ DAN CI is best-suited for small- to medium-sized rooms that require high-quality audio solutions using AEC, voice lift, and mix-minus, such as conference rooms or distance learning environments.

BENEFITS

- Includes default configuration file, allowing for plug-and-play usage
- Highly scalable and cost-effective solution that can grow over time with the needs of the customer
- Acoustic Echo Cancellation (AEC) technology on all 12 inputs
- SpeechSense™ technology enhances speech processing
- Integrates directly with soft codecs and other USB audio hosts

FEATURES

- 32 x 32 channels of digital audio networking via the Dante protocol
- 12 mic/line level inputs with AEC, 8 mic/line level outputs
- 2 Gigabit Ethernet ports: Dante digital audio and Tesira control
- Up to 8 channels of configurable USB audio
- RS-232 serial port
- 4-pin GPIO
- 2-line OLED display with capacitive-touch navigation
- Rack mountable (1RU)
- System configuration and control via Ethernet
- Internal universal power supply
- Signal processing via intuitive software allows configuration and control for signal routing, mixing, equalization, filtering, delay and much more
- CE marked, UL listed, and RoHS compliant
- Covered by Biamp Systems' 5-year warranty

ARCHITECTS & ENGINEERS SPECIFICATION

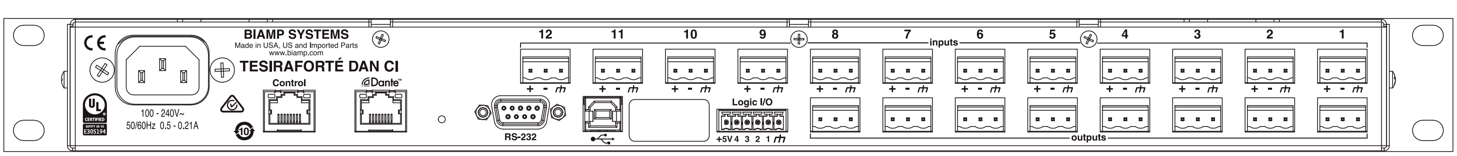
The fixed audio DSP shall be designed exclusively for use with Tesira® systems. The audio DSP shall support Dante™ digital audio networking that shall allow up to 32 x 32 channels. The Dante networking connection shall be implemented on a RJ-45 connector. The audio DSP shall support Ethernet connection for programming and control on a RJ-45 connector. The audio DSP shall have internal DSP processing. The audio 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 audio DSP shall include a RS-232 connection for control data transmission into or out of the audio DSP and such operation shall be software programmable. The audio DSP shall include a Universal Serial Bus (USB) connection on a standard USB-B type connector. The audio DSP shall be software configurable to stream up to 8 channels of digital USB Class 1 Audio transmission either into or out of the audio DSP or simultaneous input and output. The audio DSP shall provide 12 balanced input connections for receiving of microphone or line level analog audio signals on screw-down, removable connectors. The input connections shall include Acoustic Echo Cancellation (AEC) hardware and firmware, the parameters, routing and operation of which shall be software programmable. The audio DSP shall provide 8 balanced output channels for the transmission of microphone or line level analog audio signals on screw-down, removable connectors. Each individual channel shall have its own dedicated connection. The audio DSP shall provide front panel OLED identification of device power, status, alarm, and activity as well as system-wide alarm. The audio DSP shall be rack mountable (1RU) and 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 audio DSP shall be CE marked, UL listed, and shall be compliant with the RoHS directive. Warranty shall be five years. The fixed audio DSP shall be TesiraFORTÉ DAN CI.

TESIRAFORTÉ DAN CI SPECIFICATIONS

Frequency Response:	
20Hz to 20kHz, +4dBu output:	+0.25 dB/-0.5 dB
THD+N (22Hz to 22kHz):	
0dB gain, +4dBu input:	< 0.006%
54dB gain, -50dBu input:	< 0.040%
EIN (no weighting, 22Hz to 22kHz):	< -125dBu
Dynamic Range (in presence of signal)	
22Hz to 22kHz, 0dB gain:	> 108dB
Input Impedance (balanced):	8kΩ
Output Impedance (balanced):	207Ω
Maximum Input:	+24dBu
Maximum Output (selectable):	+24dBu, +18dBu, +12dBu, +6dBu, 0dBu, -31dBu
Input Gain Range (6dB steps):	0-66dB
Overall Dimensions:	
Height:	1.75 inches (44 mm)
Width:	19.0 inches (483 mm)
Depth:	10.5 inches (267 mm)
Weight:	8 lbs (3.63 kg)
Environment:	
Ambient Operating Temperature Range:	32-104° F (0–40° C)
Humidity:	0–98%, non-condensing
Altitude:	0-6,600 feet (0–2000 Meters) MSL

Phantom Power:	+48VDC (7mA/input)
Crosstalk, channel to channel, 1 kHz:	
0dB gain, +4dBu input:	< -85dB
54dB gain, -50dBu input:	< -75dB
Sampling Rate:	48kHz
A/D - D/A Converters:	24-bit
Power Consumption:	
100-240VAC 50/60Hz:	< 35W
USB:	
Bit Depth:	16- or 24-bit
Number of Channels:	up to 8
Sample Rate:	48kHz
Compliance:	FCC Part 15B (USA) CE marked (Europe) UL und C-UL listed (USA and Canada) RCM (Australia) RoHS Directive (Europe)

TESIRAFORTÉ DAN CI BACK PANEL



biamp.

A: 9300 S.W. Gemini Drive Beaverton, OR 97008 USA T: +1 503.641.7287 W: www.biamp.com

DATA SHEET
TESIRAFORTÉ® VT FIXED AUDIO DSP

The TesiraFORTÉ® VT is a fixed audio DSP with 12 analog inputs and 8 analog outputs and includes Acoustic Echo Cancellation (AEC) technology on all 12 inputs. It also includes up to 8 channels of configurable USB audio, a 2-channel VoIP interface, and a standard FXO telephone interface. USB audio allows TesiraFORTÉ to interface directly with USB audio hosts, as well as to take full advantage of today's most sophisticated conferencing solutions. TesiraFORTÉ VT 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; all configured through the Tesira configuration software. TesiraFORTÉ VT is best-suited for rooms that require high-quality audio solutions using VoIP, voice lift, mix-minus, and AEC, such as conference rooms or distance learning environments.

BENEFITS

- Integrates VoIP, POTS, and USB audio in one product allowing integrators to choose the type of audio conferencing that works best for their installation
- Includes default configuration file allowing for plug-and-play usage
- Highly scalable and cost-effective solution that can grow over time with the needs of the customer
- SpeechSense™ technology to enhance speech processing
- Integrates directly with soft codecs and other USB audio hosts

FEATURES

- 12 mic/line level inputs with AEC, 8 mic/line level outputs
- Gigabit Ethernet port
- Up to 8 channels of configurable USB audio
- RS-232 serial port
- 4-pin GPIO
- 2-line OLED display with capacitive-touch navigation
- Rack mountable (1RU)
- System configuration and control via Ethernet
- Internal universal power supply
- SIP VoIP interface via a RJ-45 connector
- Standard FXO telephone interface via RJ-11 connector
- Signal processing via intuitive software allows configuration and control for signal routing, mixing, equalization, filtering, delay and much more
- CE marked, UL listed, and RoHS compliant
- Covered by Biamp Systems' 5-year warranty

ARCHITECTS & ENGINEERS SPECIFICATION

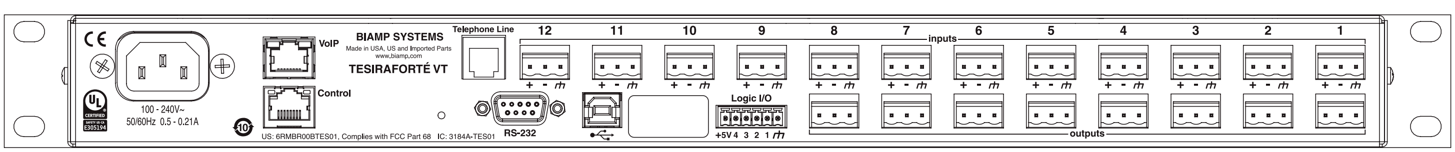
The fixed audio DSP shall be designed exclusively for use with Tesira® systems. The audio DSP shall support Ethernet connection for programming and control on a RJ-45 connector. The audio DSP shall have internal DSP processing. The audio DSP shall include 4 channels of General Purpose Input and Output connection (GPIO) for sending or receiving logic signals. The programming of the GPIO ports shall be software configurable. The audio DSP shall include a RS-232 connection for control data transmission into or out of the audio DSP and such operation shall be software programmable. The audio DSP shall include a Universal Serial Bus (USB) connection on a standard USB-B type connector. The audio DSP shall be software configurable to stream up to 8 channels of digital USB Class 1 Audio transmission either into or out of the audio DSP or simultaneous input and output. The audio DSP shall provide 4 balanced input connections for receiving of microphone or line level analog audio signals on screw-down, removable connectors. The input connections shall include Acoustic Echo Cancellation (AEC) hardware and firmware, the parameters, routing and operation of which shall be software programmable. The audio DSP shall provide 4 balanced output channels for the transmission of microphone or line level analog audio signals on screw-down, removable connectors. Each individual channel shall have its own dedicated connection. The audio DSP shall integrate to Voice Over Internet Protocol (VoIP) systems on a RJ-45 connector for two lines of VoIP communication and shall support Session Initiation Protocol (SIP) v2.0 or later. The audio DSP shall integrate to standard telephony communications on a RJ-11 connector for a single line of telephone communication. The audio DSP shall provide front panel OLED identification of device power, status, alarm, and activity as well as system-wide alarm. The audio DSP shall be rack mountable (1RU) and 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 audio DSP shall be CE marked, UL listed, and shall be compliant with the RoHS directive. Warranty shall be five years. The fixed audio DSP shall be TesiraFORTÉ VT.

TESIRAFORTÉ VT SPECIFICATIONS

Frequency Response:	
20Hz to 20kHz, +4dBu output:	+0.25 dB/-0.5 dB
THD+N (22Hz to 22kHz):	
0dB gain, +4dBu input:	< 0.006%
54dB gain, -50dBu input:	< 0.040%
EIN (no weighting, 22Hz to 22kHz):	< -125dBu
Dynamic Range (in presence of signal)	
22Hz to 22kHz, 0dB gain:	> 108dB
Input Impedance (balanced):	8kΩ
Output Impedance (balanced):	207Ω
Maximum Input:	+24dBu
Maximum Output (selectable):	+24dBu, +18dBu, +12dBu, +6dBu, 0dBu, -31dBu
Input Gain Range (6dB steps):	0-66dB
Overall Dimensions:	
Height:	1.75 inches (44 mm)
Width:	19.0 inches (483 mm)
Depth:	10.5 inches (267 mm)
Weight:	8 lbs (3.63 kg)
Environment:	
Ambient Operating Temperature Range:	32-104° F (0–40° C)
Humidity:	0–98%, non-condensing
Altitude:	0-6,600 feet (0–2000 Meters) MSL

Phantom Power:	+48VDC (7mA/input)
Crosstalk, channel to channel, 1 kHz:	
0dB gain, +4dBu input:	< -85dB
54dB gain, -50dBu input:	< -75dB
Sampling Rate:	48kHz
A/D - D/A Converters:	24-bit
Power Consumption:	
100-240VAC 50/60Hz:	< 35W
USB:	
Bit Depth:	16- or 24-bit
Number of Channels:	up to 8
Sample Rate:	48kHz
Compliance:	
	FCC Part 15B (USA)
	FCC Part 68 (USA)
	Industry Canada CS-03 (Canada)
	CE marked (Europe)
	UL und C-UL listed (USA and Canada)
	RCM (Australia)
	RoHS Directive (Europe)

TESIRAFORTÉ VT BACK PANEL



DATA SHEET
TESIRAFORTÉ® AVB VT FIXED AUDIO DSP

The TesiraFORTÉ® AVB VT is a fixed audio DSP with 12 analog inputs and 8 analog outputs and includes Acoustic Echo Cancellation (AEC) technology on all 12 inputs.It includes up to 8 channels of configurable USB audio, a 2-channel VoIP interface, and a standard FXO telephone interface. USB audio allows TesiraFORTÉ to interface directly with USB audio hosts, as well as to take full advantage of today's most sophisticated conferencing solutions. TesiraFORTÉ AVB VT utilizes Audio Video Bridging (AVB) for digital audio networking, and can be used as standalone device or combined with other TesiraFORTÉ AVB devices and Tesira servers, expanders, endpoints, and controllers. TesiraFORTÉ AVB VT also 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; all configured through the Tesira configuration software. TesiraFORTÉ AVB VT is best-suited for rooms that require high-quality audio solutions using VoIP, voice lift, mix-minus, and AEC, such as conference rooms or distance learning environments.

BENEFITS

- Integrates VoIP, POTS, and USB audio in one product allowing integrators to choose the type of audio conferencing that works best for their installation
- AVB allows audio networking via IEEE open standards protocol
- Includes default configuration file allowing for plug-and-play usage
- Highly scalable and cost-effective solution that can grow over time with the needs of the customer
- SpeechSense™ technology to enhance speech processing
- Integrates directly with soft codecs and other USB audio hosts

FEATURES

- 128 x 128 channels of AVB
- 12 mic/line level inputs with AEC, 8 mic/line level outputs
- Gigabit Ethernet port
- Up to 8 channels of configurable USB audio
- RS-232 serial port
- 4-pin GPIO
- 2-line OLED display with capacitive-touch navigation
- Rack mountable (1RU)
- System configuration and control via Ethernet
- Internal universal power supply
- SIP VoIP interface via a RJ-45 connector
- Standard FXO telephone interface via RJ-11 connector
- Fully compatible with Tesira AVB servers, endpoints, expanders, and controllers
- Signal processing via intuitive software allows configuration and control for signal routing, mixing, equalization, filtering, delay and much more
- CE marked, UL listed, and RoHS compliant
- Covered by Biamp Systems' 5-year warranty

ARCHITECTS & ENGINEERS SPECIFICATION

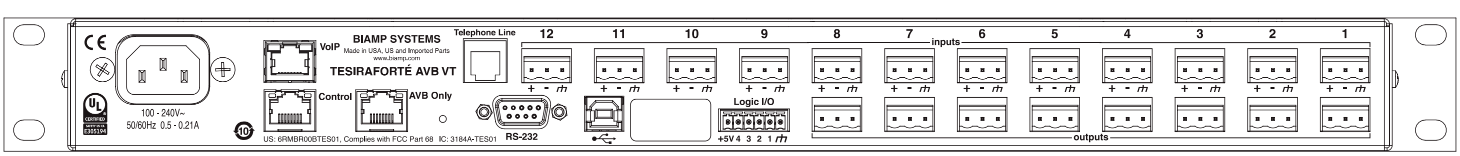
The fixed audio DSP shall be designed exclusively for use with Tesira® systems. The audio DSP shall support Audio Video Bridging (AVB) digital audio networking that shall allow up to 128 x 128 channels. The AVB networking connection shall be implemented on a RJ-45 connector. The audio DSP shall support Ethernet connection for programming and control on a RJ-45 connector. The audio DSP shall have internal DSP processing. The audio 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 audio DSP shall include a RS-232 connection for control data transmission into or out of the audio DSP and such operation shall be software programmable. The audio DSP shall include a Universal Serial Bus (USB) connection on a standard USB-B type connector. The audio DSP shall be software configurable to stream up to 8 channels of digital USB Class 1 Audio transmission either into or out of the audio DSP or simultaneous input and output. The audio DSP shall provide 12 balanced input connections for receiving of microphone or line level analog audio signals on screw-down, removable connectors. The input connections shall include Acoustic Echo Cancellation (AEC) hardware and firmware, the parameters, routing and operation of which shall be software programmable. The audio DSP shall provide 8 balanced output channels for the transmission of microphone or line level analog audio signals on screw-down, removable connectors. Each individual channel shall have its own dedicated connection. The audio DSP shall integrate to Voice Over Internet Protocol (VoIP) systems on a RJ-45 connector for two lines of VoIP communication and shall support Session Initiation Protocol (SIP) v2.0 or later. The audio DSP shall integrate to standard telephony communications on a RJ-11 connector for a single line of telephone communication. The audio DSP shall provide front panel OLED identification of device power, status, alarm, and activity as well as system-wide alarm. The audio DSP shall be rack mountable (1RU) and 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 audio DSP shall control and proxy all Tesira expander-class devices and Tesira control devices. The audio DSP shall be CE marked, UL listed, and shall be compliant with the RoHS directive. Warranty shall be five years. The fixed audio DSP shall be TesiraFORTÉ AVB VT.

TESIRAFORTÉ AVB VT SPECIFICATIONS

Frequency Response:	
20Hz to 20kHz, +4dBu output:	+0.25 dB/-0.5 dB
THD+N (22Hz to 22kHz):	
0dB gain, +4dBu input:	< 0.006%
54dB gain, -50dBu input:	< 0.040%
EIN (no weighting, 22Hz to 22kHz):	< -125dBu
Dynamic Range (in presence of signal)	
22Hz to 22kHz, 0dB gain:	> 108dB
Input Impedance (balanced):	8kΩ
Output Impedance (balanced):	207Ω
Maximum Input:	+24dBu
Maximum Output (selectable):	+24dBu, +18dBu, +12dBu, +6dBu, 0dBu, -31dBu
Input Gain Range (6dB steps):	0-66dB
Overall Dimensions:	
Height:	1.75 inches (44 mm)
Width:	19.0 inches (483 mm)
Depth:	10.5 inches (267 mm)
Weight:	8 lbs (3.63 kg)
Environment:	
Ambient Operating Temperature Range:	32-104° F (0–40° C)
Humidity:	0–98%, non-condensing
Altitude:	0-6,600 feet (0–2000 Meters) MSL

Phantom Power:	+48VDC (7mA/input)
Crosstalk, channel to channel, 1 kHz:	
0dB gain, +4dBu input:	< -85dB
54dB gain, -50dBu input:	< -75dB
Sampling Rate:	48kHz
A/D - D/A Converters:	24-bit
Power Consumption:	
100-240VAC 50/60Hz:	< 35W
USB:	
Bit Depth:	16- or 24-bit
Number of Channels:	up to 8
Sample Rate:	48kHz
Compliance:	FCC Part 15B (USA) FCC Part 68 (USA) Industry Canada CS-03 (Canada) CE marked (Europe) UL und C-UL listed (USA and Canada) RCM (Australia) RoHS Directive (Europe)

TESIRAFORTÉ AVB VT BACK PANEL



DATA SHEET
TESIRAFORTÉ® DAN VT FIXED AUDIO DSP

The TesiraFORTÉ® DAN VT is a fixed audio DSP with 32 bi-directional channels of Dante™ digital audio, 12 analog inputs with Acoustic Echo Cancellation (AEC) technology, and 8 analog outputs. It also includes up to 8 channels of configurable USB audio, a 2-channel VoIP interface, and a standard FXO telephone interface. USB audio allows TesiraFORTÉ to interface directly with USB audio hosts, as well as to take full advantage of today's most sophisticated conferencing solutions. TesiraFORTÉ DAN VT also 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; all configured through the Tesira configuration software. TesiraFORTÉ DAN VT is best-suited for rooms that require high-quality audio solutions using VoIP, voice lift, mix-minus, and AEC, such as conference rooms or distance learning environments.

BENEFITS

- Integrates VoIP, POTS, and USB audio in one product allowing integrators to choose the type of audio conferencing that works best for their installation
- Includes default configuration file allowing for plug-and-play usage
- Highly scalable and cost-effective solution that can grow over time with the needs of the customer
- SpeechSense™ technology to enhance speech processing
- Integrates directly with soft codecs and other USB audio hosts

FEATURES

- 32 x 32 channels of digital audio networking via the Dante protocol
- 12 mic/line level inputs with AEC, 8 mic/line level outputs
- Gigabit Ethernet port
- Up to 8 channels of configurable USB audio
- RS-232 serial port
- 4-pin GPIO
- 2-line OLED display with capacitive-touch navigation
- Rack mountable (1RU)
- System configuration and control via Ethernet
- Internal universal power supply
- SIP VoIP interface via a RJ-45 connector
- Standard FXO telephone interface via RJ-11 connector
- Signal processing via intuitive software allows configuration and control for signal routing, mixing, equalization, filtering, delay and much more
- CE marked, UL listed, and RoHS compliant
- Covered by Biamp Systems' 5-year warranty

ARCHITECTS & ENGINEERS SPECIFICATION

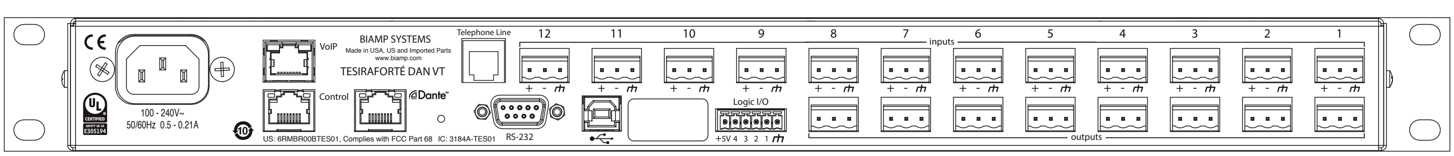
The fixed audio DSP shall be designed exclusively for use with Tesira® systems. The audio DSP shall support Dante™ digital audio networking that shall allow up to 32 x 32 channels. The Dante networking connection shall be implemented on a RJ-45 connector. The audio DSP shall support Ethernet connection for programming and control on a RJ-45 connector. The audio DSP shall have internal DSP processing. The audio 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 audio DSP shall include a RS-232 connection for control data transmission into or out of the audio DSP and such operation shall be software programmable. The audio DSP shall include a Universal Serial Bus (USB) connection on a standard USB-B type connector. The audio DSP shall be software configurable to stream up to 8 channels of digital USB Class 1 Audio transmission either into or out of the audio DSP or simultaneous input and output. The audio DSP shall provide 12 balanced input connections for receiving of microphone or line level analog audio signals on screw-down, removable connectors. The input connections shall include Acoustic Echo Cancellation (AEC) hardware and firmware, the parameters, routing and operation of which shall be software programmable. The audio DSP shall provide 8 balanced output channels for the transmission of microphone or line level analog audio signals on screw-down, removable connectors. Each individual channel shall have its own dedicated connection. The audio DSP shall integrate to Voice Over Internet Protocol (VoIP) systems on a RJ-45 connector for two lines of VoIP communication and shall support Session Initiation Protocol (SIP) v2.0 or later. The audio DSP shall integrate to standard telephony communications on a RJ-11 connector for a single line of telephone communication. The audio DSP shall provide front panel OLED identification of device power, status, alarm, and activity as well as system-wide alarm. The audio DSP shall be rack mountable (1RU) and 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 audio DSP shall be CE marked, UL listed, and shall be compliant with the RoHS directive. Warranty shall be five years. The fixed audio DSP shall be TesiraFORTÉ DAN VT.

TESIRAFORTÉ DAN VT SPECIFICATIONS

Frequency Response:	
20Hz to 20kHz, +4dBu output:	+0.25 dB/-0.5 dB
THD+N (22Hz to 22kHz):	
0dB gain, +4dBu input:	< 0.006%
54dB gain, -50dBu input:	< 0.040%
EIN (no weighting, 22Hz to 22kHz):	< -125dBu
Dynamic Range (in presence of signal)	
22Hz to 22kHz, 0dB gain:	> 108dB
Input Impedance (balanced):	8kΩ
Output Impedance (balanced):	207Ω
Maximum Input:	+24dBu
Maximum Output (selectable):	+24dBu, +18dBu, +12dBu, +6dBu, 0dBu, -31dBu
Input Gain Range (6dB steps):	0-66dB
Overall Dimensions:	
Height:	1.75 inches (44 mm)
Width:	19.0 inches (483 mm)
Depth:	10.5 inches (267 mm)
Weight:	8 lbs (3.63 kg)
Environment:	
Ambient Operating Temperature Range:	32-104° F (0–40° C)
Humidity:	0–98%, non-condensing
Altitude:	0-6,600 feet (0–2000 Meters) MSL

Phantom Power:	+48VDC (7mA/input)
Crosstalk, channel to channel, 1 kHz:	
0dB gain, +4dBu input:	< -85dB
54dB gain, -50dBu input:	< -75dB
Sampling Rate:	48kHz
A/D - D/A Converters:	24-bit
Power Consumption:	
100-240VAC 50/60Hz:	< 35W
USB:	
Bit Depth:	16- or 24-bit
Number of Channels:	up to 8
Sample Rate:	48kHz
Compliance:	FCC Part 15B (USA) FCC Part 68 (USA) Industry Canada CS-03 (Canada) CE marked (Europe) UL und C-UL listed (USA and Canada) RCM (Australia) RoHS Directive (Europe)

TESIRAFORTÉ DAN VT BACK PANEL



DATA SHEET
TESIRAFORTÉ® AVB VT4 FIXED AUDIO DSP

The TesiraFORTÉ® AVB VT4 is a fixed audio DSP with 4 analog inputs, 4 channels of Acoustic Echo Cancellation (AEC) technology, and 4 analog outputs. It also includes up to 8 channels of configurable USB audio, a 2-channel VoIP interface and a standard FXO telephone interface. USB audio allows TesiraFORTÉ to interface directly with USB audio hosts, as well as to take full advantage of today's most sophisticated conferencing solutions. TesiraFORTÉ AVB VT4 utilizes Audio Video Bridging (AVB) for digital audio networking, and can be used as a standalone device or combined with other TesiraFORTÉ AVB devices and Tesira servers, expanders, and controllers. TesiraFORTÉ AVB VT4 also 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; all configured through the Tesira configuration software. TesiraFORTÉ AVB VT4 is best-suited for smaller rooms that require high-quality audio solutions using VoIP, voice lift, mix-minus, and AEC, such as conference rooms or distance learning environments.

BENEFITS

- Integrates VoIP, POTS, and USB audio in one product allowing integrators to choose the type of audio conferencing that works best for their installation
- AVB allows audio networking via IEEE open standards protocol
- Includes default configuration file allowing for plug-and-play usage
- Highly scalable and cost-effective solution that can grow over time with the needs of the customer
- SpeechSense™ technology to enhance speech processing
- Integrates directly with soft codecs and other USB audio hosts

FEATURES

- 128 x 128 channels of AVB
- 4 mic/line level inputs with AEC, 4 mic/line level outputs
- Gigabit Ethernet port
- Up to 8 channels of configurable USB audio
- RS-232 serial port
- 4-pin GPIO
- 2-line OLED display with capacitive-touch navigation
- Rack mountable (1RU)
- System configuration and control via Ethernet
- Internal universal power supply
- SIP VoIP interface via a RJ-45 connector
- Standard FXO telephone interface via RJ-11 connector
- Fully compatible with Tesira AVB servers, endpoints, expanders, and controllers
- Signal processing via intuitive software allows configuration and control for signal routing, mixing, equalization, filtering, delay and much more
- CE marked, UL listed, and RoHS compliant
- Covered by Biamp Systems' 5-year warranty

ARCHITECTS & ENGINEERS SPECIFICATION

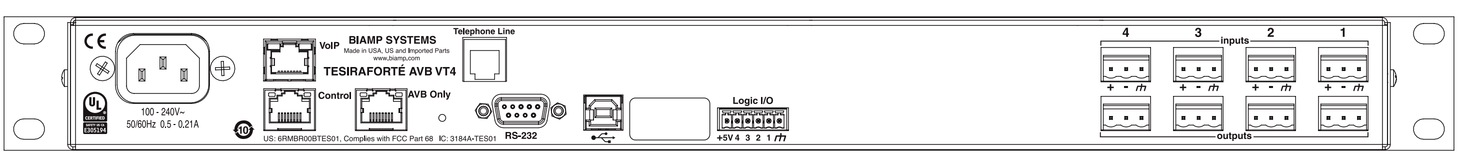
The fixed audio DSP shall be designed exclusively for use with Tesira® systems. The audio DSP shall support Audio Video Bridging (AVB) digital audio networking that shall allow up to 128 x 128 channels. The AVB networking connection shall be implemented on a RJ-45 connector. The audio DSP shall support Ethernet connection for programming and control on a RJ-45 connector. The audio DSP shall have internal DSP processing. The audio 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 audio DSP shall include a RS-232 connection for control data transmission into or out of the audio DSP and such operation shall be software programmable. The audio DSP shall include a Universal Serial Bus (USB) connection on a standard USB-B type connector. The audio DSP shall be software configurable to stream up to 8 channels of digital USB Class 1 Audio transmission either into or out of the audio DSP or simultaneous input and output. The audio DSP shall provide 4 balanced input connections for receiving of microphone or line level analog audio signals on screw-down, removable connectors. The input connections shall include Acoustic Echo Cancellation (AEC) hardware and firmware, the parameters, routing and operation of which shall be software programmable. The audio DSP shall provide 4 balanced output channels for the transmission of microphone or line level analog audio signals on screw-down, removable connectors. Each individual channel shall have its own dedicated connection. The audio DSP shall integrate to Voice Over Internet Protocol (VoIP) systems on a RJ-45 connector for two lines of VoIP communication and shall support Session Initiation Protocol (SIP) v2.0 or later. The audio DSP shall integrate to standard telephony communications on a RJ-11 connector for a single line of telephone communication. The audio DSP shall provide front panel OLED identification of device power, status, alarm, and activity as well as system-wide alarm. The audio DSP shall be rack mountable (1RU) and 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 audio DSP shall control and proxy all Tesira expander-class devices and Tesira control devices. The audio DSP shall be CE marked, UL listed, and shall be compliant with the RoHS directive. Warranty shall be five years. The fixed audio DSP shall be TesiraFORTÉ AVB VT4.

TESIRAFORTÉ AVB VT4 SPECIFICATIONS

Frequency Response:	
20Hz to 20kHz, +4dBu output:	+0.25 dB/-0.5 dB
THD+N (22Hz to 22kHz):	
0dB gain, +4dBu input:	< 0.006%
54dB gain, -50dBu input:	< 0.040%
EIN (no weighting, 22Hz to 22kHz):	< -125dBu
Dynamic Range (in presence of signal)	
22Hz to 22kHz, 0dB gain:	> 108dB
Input Impedance (balanced):	8kΩ
Output Impedance (balanced):	207Ω
Maximum Input:	+24dBu
Maximum Output (selectable):	+24dBu, +18dBu, +12dBu, +6dBu, 0dBu, -31dBu
Input Gain Range (6dB steps):	0-66dB
Overall Dimensions:	
Height:	1.75 inches (44 mm)
Width:	19.0 inches (483 mm)
Depth:	10.5 inches (267 mm)
Weight:	8 lbs (3.63 kg)
Environment:	
Ambient Operating Temperature Range:	32-104° F (0–40° C)
Humidity:	0–98%, non-condensing
Altitude:	0-6,600 feet (0–2000 Meters) MSL

Phantom Power:	+48VDC (7mA/input)
Crosstalk, channel to channel, 1 kHz:	
0dB gain, +4dBu input:	< -85dB
54dB gain, -50dBu input:	< -75dB
Sampling Rate:	48kHz
A/D - D/A Converters:	24-bit
Power Consumption:	
100-240VAC 50/60Hz:	< 35W
USB:	
Bit Depth:	16- or 24-bit
Number of Channels:	up to 8
Sample Rate:	48kHz
Compliance:	FCC Part 15B (USA) FCC Part 68 (USA) Industry Canada CS-03 (Canada) CE marked (Europe) UL und C-UL listed (USA and Canada) RCM (Australia) RoHS Directive (Europe)

TESIRAFORTÉ AVB VT4 BACK PANEL



DATA SHEET
TESIRAFORTÉ® DAN VT4 FIXED AUDIO DSP

The TesiraFORTÉ® DAN VT4 is a fixed audio DSP with 32 bi-directional channels of Dante™ digital audio, 4 analog inputs, 4 channels of Acoustic Echo Cancellation (AEC) technology, and 4 analog outputs. It also includes up to 8 channels of configurable USB audio, a 2-channel VoIP interface and a standard FXO telephone interface. USB audio allows TesiraFORTÉ to interface directly with USB audio hosts, as well as to take full advantage of today's most sophisticated conferencing solutions. TesiraFORTÉ DAN VT4 also 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; all configured through the Tesira configuration software. TesiraFORTÉ DAN VT4 is best-suited for smaller rooms that require high-quality audio solutions using VoIP, voice lift, mix-minus, and AEC, such as conference rooms or distance learning environments.

BENEFITS

- Integrates VoIP, POTS, and USB audio in one product allowing integrators to choose the type of audio conferencing that works best for their installation
- Includes default configuration file allowing for plug-and-play usage
- Highly scalable and cost-effective solution that can grow over time with the needs of the customer
- SpeechSense™ technology to enhance speech processing
- Integrates directly with soft codecs and other USB audio hosts

FEATURES

- 32 x 32 channels of digital audio networking via the Dante protocol
- 4 mic/line level inputs with AEC, 4 mic/line level outputs
- Gigabit Ethernet port
- Up to 8 channels of configurable USB audio
- RS-232 serial port
- 4-pin GPIO
- 2-line OLED display with capacitive-touch navigation
- Rack mountable (1RU)
- System configuration and control via Ethernet
- Internal universal power supply
- SIP VoIP interface via a RJ-45 connector
- Standard FXO telephone interface via RJ-11 connector
- Signal processing via intuitive software allows configuration and control for signal routing, mixing, equalization, filtering, delay and much more
- CE marked, UL listed, and RoHS compliant
- Covered by Biamp Systems' 5-year warranty

biamp.

A: 9300 S.W. Gemini Drive Beaverton, OR 97008 USA T: +1 503.641.7287 W: www.biamp.com

ARCHITECTS & ENGINEERS SPECIFICATION

The fixed audio DSP shall be designed exclusively for use with Tesira® systems. The audio DSP shall support Dante™ digital audio networking that shall allow up to 32 x 32 channels. The Dante networking connection shall be implemented on a RJ-45 connector. The audio DSP shall support Ethernet connection for programming and control on a RJ-45 connector. The audio DSP shall have internal DSP processing. The audio 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 audio DSP shall include a RS-232 connection for control data transmission into or out of the audio DSP and such operation shall be software programmable. The audio DSP shall include a Universal Serial Bus (USB) connection on a standard USB-B type connector. The audio DSP shall be software configurable to stream up to 8 channels of digital USB Class 1 Audio transmission either into or out of the audio DSP or simultaneous input and output. The audio DSP shall provide 4 balanced input connections for receiving of microphone or line level analog audio signals on screw-down, removable connectors. The input connections shall include Acoustic Echo Cancellation (AEC) hardware and firmware, the parameters, routing and operation of which shall be software programmable. The audio DSP shall provide 4 balanced output channels for the transmission of microphone or line level analog audio signals on screw-down, removable connectors. Each individual channel shall have its own dedicated connection. The audio DSP shall integrate to Voice Over Internet Protocol (VoIP) systems on a RJ-45 connector for two lines of VoIP communication and shall support Session Initiation Protocol (SIP) v2.0 or later. The audio DSP shall integrate to standard telephony communications on a RJ-11 connector for a single line of telephone communication. The audio DSP shall provide front panel OLED identification of device power, status, alarm, and activity as well as system-wide alarm. The audio DSP shall be rack mountable (1RU) and 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 audio DSP shall be CE marked, UL listed, and shall be compliant with the RoHS directive. Warranty shall be five years. The fixed audio DSP shall be TesiraFORTÉ DAN VT4.

TESIRAFORTÉ DAN VT4 SPECIFICATIONS

Frequency Response:	
20Hz to 20kHz, +4dBu output:	+0.25 dB/-0.5 dB
THD+N (22Hz to 22kHz):	
0dB gain, +4dBu input:	< 0.006%
54dB gain, -50dBu input:	< 0.040%
EIN (no weighting, 22Hz to 22kHz):	< -125dBu
Dynamic Range (in presence of signal)	
22Hz to 22kHz, 0dB gain:	> 108dB
Input Impedance (balanced):	8kΩ
Output Impedance (balanced):	207Ω
Maximum Input:	+24dBu
Maximum Output (selectable):	+24dBu, +18dBu, +12dBu, +6dBu, 0dBu, -31dBu
Input Gain Range (6dB steps):	0-66dB
Overall Dimensions:	
Height:	1.75 inches (44 mm)
Width:	19.0 inches (483 mm)
Depth:	10.5 inches (267 mm)
Weight:	8 lbs (3.63 kg)
Environment:	
Ambient Operating Temperature Range:	32-104° F (0–40° C)
Humidity:	0–98%, non-condensing
Altitude:	0-6,600 feet (0–2000 Meters) MSL

Phantom Power:	+48VDC (7mA/input)
Crosstalk, channel to channel, 1 kHz:	
0dB gain, +4dBu input:	< -85dB
54dB gain, -50dBu input:	< -75dB
Sampling Rate:	48kHz
A/D - D/A Converters:	24-bit
Power Consumption:	
100-240VAC 50/60Hz:	< 35W
USB:	
Bit Depth:	16- or 24-bit
Number of Channels:	up to 8
Sample Rate:	48kHz
Compliance:	
	FCC Part 15B (USA)
	FCC Part 68 (USA)
	Industry Canada CS-03 (Canada)
	CE marked (Europe)
	UL und C-UL listed (USA and Canada)
	RCM (Australia)
	RoHS Directive (Europe)

TESIRAFORTÉ DAN VT4 BACK PANEL