Plixus AE

Plixus Audio Engine

Description

Vital Functions in a Beautiful Form
The Plixus audio engine is a 19" rack mountable device that provides audio processing and signal handling required for the Plixus network. The Plixus AE follows the same design philosophy as other Plixus compliant devices: *a minimalistic user interface offers direct access to a number of vital functions*. This makes control of the system easy and intuitive for the most common functionality. A set of distinctive, yet unobtrusive handles gives the central equipment a sleek look. A touch of brushed aluminum combined with a captivating red glow underlines the unit’s exclusivity and makes it blend in with the Televic product family.

All in One Cat 5e Cable
The Plixus AE has four conference ports. A single cat 5e network cable interconnects the delegate units in daisy-chain or in closed loop for extra redundancy using Televic’s patented *Dual Branch technology*. This standard cable transports high-quality audio.

CoCon: Meeting Controls at Your Fingertips
The CoCon management software allows fine-grained control over every aspect of the meeting. The software connects the LAN connection to the central unit.

Dante™: Simple Third-party Interfacing
To interface with third-party equipment, the Plixus multimedia engine supports a Dante™ audio networking card (**71.98.2950**). As a result, the Plixus conference system can easily interconnect to Dante™-enabled devices such as DSPs, audio mixers or recording devices. Through the Dante controller software, the audio can then be routed between any Dante enabled device that is available on the network.

Smarter Grouping & Processing
In terms of audio signal processing, the central unit is capable to adjust the microphone sensitivity for each individual microphone via the webserver application. Several microphones can also be combined into one group and provided as an output towards the Dante interface. This opens up a host of features such as distributed echo cancellation, room equalization, or even recording different audio channels in court applications. An integrated dynamics processor with a programmable threshold, ratio, attack and release of noise gate, AGC, and limiter functions is also available to process the audio in function of the environment.

Analog Out & Power
The Plixus AE also provides analog interfacing to the outside world. It supplies 1 balanced and 2 unbalanced inputs and 1 balanced and 2 unbalanced outputs. Finally, the unit has an integrated 400 W power supply. A connector is provided at the back of the engine to power conference equipment that requires external power.

Benefits

Rock-solid Network Performance
Plixus is a *packet-based* network with a proprietary protocol developed by Televic, specifically for mission-critical conference applications. Through dynamic bandwidth attribution, it offers *guaranteed audio quality*.

HD Audio
The philosophy of Plixus is to maximize the use of available bandwidth so that there is no need to compromise on audio quality. 64 Channels of audio are passed uncompressed over the network at 48 kSps.

Closed Architecture, Open Interfacing
The Plixus conference network is closed and open at the same time. While for the benefit of security no third-party devices or connections are allowed on the mission critical part of the network, the Plixus Engine at the edge of the network has an open interface. In this way the best of both worlds are combined: *open yet secure interfacing*.

Self-healing Topology
The packet-based nature of Plixus allows the conference network to be aware of its topology at any moment. *During normal operation data will travel the shortest route from the Plixus Engine to the delegate unit and vice versa*. In case of a failure along that route (i.e. a unit failing or a cable breaking) Plixus will self-correct and calculate a new shortest route so that data packets still reach their intended destination.

Loop Cabling
For this self-healing mechanism to work, redundant paths must be provided through loop cabling. *You may also set up Plixus Network Extenders in a redundant configuration*.

Features

- Single conference network to transport audio,
- Cat5e cabling, maximum 100 m between two conference devices
- Single loop permits a maximum total cable length of 400 m
- Support for daisy chain and loop configuration
- Loop cabling redundancy capabilities
- Patent-pending HOT SWAP functionality. After replacement of a defective unit, the new unit will automatically be configured with the settings of the old one.
- Built-in power supply of 400 W
- External ‘Power out’ connector to drive equipment that needs separate power.
- Integrated dynamics processor with programmable threshold, ratio, attack and

release of noise gate, AGC and limiter functions.
- Dante™ multi-channel networked audio (maximum 64 channels) via separate add-on card (**71.98.2950** Dante Audio Networking Card)
- Individual microphone sensitivity and equalizer adjustments
- Patent-pending scalable software architecture The engine supports following microphone modes
    - Direct access
    - Request
    - FIFO
    - Vox
- Dynamic bandwidth assignment
- Advanced access control
- Rerouting of packets when needed
- Proprietary Protocol
- Optimized bandwidth handling for conference applications
- Extra level of security audio transfer
- Optimized for mission-critical conference audio
- No access for viruses
- No IP eavesdropping
- No interference from non-conference IP traffic
- No performance degradation through 3rd party devices
- No possibility for rogue devices to connect
- Only Televic devices on the network: no accountability issues
- Open edge
- Dante interface
- Gatekeeping by Plixus Engine

Buttons & Modes

- A Jog Wheel on the front of the engine gives the user direct access to following settings: Mode switching between:
    - System volume
    - Microphone mode
    - Max number open microphones (1-8)
    - Headphone volume
- To prevent accidental changes to the settings a long press on Jog Wheel will lock/unlock the controls of the Plixus AE
- Reset button
    - Short press: restart engine
    - Long press: set default IP address
- Headphone output

Connectivity

- 4 conference network ports
- 1 LAN configuration port
- 2 redundant Dante ports
- 2 USB 2.0 ports (Future use)
- 2 unpowered conference network ports (Future use)
- 1 Balanced XLR audio input
- 1 Balanced XLR audio output
- 2 Unbalanced Cinch audio inputs
- 2 Unbalanced Cinch audio outputs
- 1 HD-SDI video input (*Not functional*)
- 1 HD-SDI video output (*Not functional*)
- 1 HDMI output (*Not functional*)
- Mains power connection with ON/OFF button 110 - 230VAC 50-60 Hz
- 48V output Phoenix connector
    - Output to power Network Extender Units

Certifications

Region	Certification
Europe	CE

Specifications

Mechanical	
Material	Steel
Color	Black, RAL9011
Size (mm) Size packed (mm)	485 (w) × 420 (h) × 90 (d) 610 (w) × 510 (h) × 195 (d)
Weight (g) Weight packed (g)	8200 9520
Electrical	
Supply Voltage	Internal, 90-264 VAC, 47-63 Hz
Consumption	Max 445 W (including external power)
Audio quality	24 bit, 48 kSps
Power Over Cable	
Voltage	48 VDC
Continuous output current	2 A
Auxiliary Power Output	
Voltage	48 VDC
Continuous output current	8.33 A
Current limit	13.65 A
Network	
Cable type	Cat 5e, shielded, FTP
Maximum length between units	80 m
Maximum total cable length within a loop	400 m
Connector	RJ45 standard (shielded)
IP Control Port	
IP control port link speed	100 Mbps
AUX IN XLR Balanced	
Nominal input level	+4 dBu
Maximum input level	+24 dBu
Input impedance	10 kΩ
Dynamic range	> 90 dB
Frequency response	20-20,000 Hz
AUX OUT Balanced	
Nominal output level	+4 dBu
Maximum output level	+24 dBu
Dynamic range	> 90 dB
Frequency response	20-20,000 Hz
THD @ nominal level	0.1%
Load impedance	> 600 Ω
AUX IN RCA Unbalanced	
Nominal input level	-10 dBV
Maximum input level	10 dBV
Input impedance	10 kΩ
Dynamic range	> 90 dB
Frequency response	20-20,000 Hz
AUX OUT RCA Unbalanced	
Nominal output level	-10 dBV
Maximum output	10 dBV
Dynamic range	> 90 dB
Frequency response	20-20,000 Hz
THD @ nominal level	0.1%
Load impedance	> 10 kΩ
Headphone	
Minimum output power	10 mW 32 Ω
Dynamic range	> 90 dB
Frequency response	20-20,000 Hz
THD @ nominal level	0.1%
Load impedance	16-32 Ω
Dante Interface	
Link Speed	1 Gbps
Sample Rate	48 kSps
Sample width	24 bit
Maximum number of input channels	64
Maximum number of output channels	64
Environment	
Operation temperature	5-50 °C

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