VIPEDIA-12-PRO

DSP AUDIO PROCESSOR

- IP NETWORKED DSP
- INTEGRATED DANTE NETWORKING
- HIGH QUALITY ANANLOGUE INPUTS/OUTPUTS
- 32-BIT AUDIO PROCESSING
- FLEXIBLE CONTROL OPTIONS

OVERVIEW

Simple

VIPEDIA-12-PRO is a high quality audio processor and routing matrix. Integrated Dante and Secure Loop audio support allow both centralised and IP distributed architectures. Powerful processing and a range of control options make VIPEDIA-12-PRO the ideal choice for demanding audio applications.

Perfectly suited to centralised architectures, single units support up to 12 analogue audio inputs, 32 Dante sources and 12 zones. A fixed DSP architecture and device-based graphical System Configuration Tool (SCT) make configuration intuitive and fast.

Scalable

For systems with more than 12 sources or zones, VIPEDIA-12-PRO includes integrated support for both Audinate's Dante™ and ASL's Secure Loop networking which enables up to 32 VIPEDIA-12-PRO to be networked as one group over Ethernet. A virtually unlimited number of groups can be linked to provide paging and music distribution using ASL's VIPA audio-over-IP software across larger systems.

Integrated Dante™ allows up to 32 sources to be sent over high quality uncompressed, digital media channels, with near-zero latency and exceptional synchronization across large decentralised systems.

Secure Loop offers a flexible alternative to Dante. Because broadcast streams are dynamically generated as required, there is no fixed limit on the number of inputs which can be used for network broadcasts. As many broadcasts can be made concurrently as the underlying network bandwidth can support.

Each audio source can be configured to use either Dante™ or ASL's PMC (Portable Media Carrier) networking. Both options support high quality, uncompressed audio and use UDP multicast for bandwidth efficiency in complex systems.

Flexible

ASL offer a range of compatible wall controllers and microphones. Volume adjustment and source selection can be made from ASL's WMC01 or by third party touchscreen devices such as Vity, Crestron and AMX - or any device capable of sending basic ASCII commands via UDP/IP messages, as required by ASL's Vipedia Control Protocol (VCP).

VIPEDIA-12-PRO works with both ASL and third party amplifiers over standard balanced analogue audio outputs. Sources can also be patched to Dante™ streams for simple integration with any compatible amplifiers, AV or audio processing systems.

Powerful

DSP features include parametric EQ, gate, limiter, compressor and digital audio delay.

Microphone and music inputs are connected to high quality balanced analogue audio inputs using line or microphone level audio. Each input also provides phantom power if required.

VIPEDIA-12-PRO also includes internal recorded message storage, ambient noise sensing, night-time volume control, fault reporting and integrated GPIO control inputs / outputs.

FEATURES

Network Audio
Analogue and Dante sources can be routed to local or remote analogue outputs over IP using Dante or Secure Loop protocols. For integration with external audio systems, analogue sources can also be converted to Dante channels.

Source Select & Volume Control
Surface mounted WMC01 wall controllers can be used to select audio sources and control broadcast volume. Each WMC01 also includes a 3.5mm socket to allow users to connect a music source directly into the unit. The WMC01 operates in either serial or IP mode.

Third Party Control
Vipedia Control Protocol (VCP) allows 3rd party systems such as touch screen panels and controllers to select sources and adjust levels by sending simple text based control messages over IP. Events such as *'play input 5 to outputs 1,3,7'* are defined in the Vipedia System Configuration which generates a simple ASCII control code. When sent by an external device this code triggers the associated action.

SIP Paging
In order to interfaces with existing SIP based telephony systems, VIPEDIA-12-PRO also operates as VoIP SIP telephone end point . Paging calls can be made to VIPEDIA-12-PRO from any standard telephone handset. Configuration of VoIP/SIP paging is simple and intuitive. Both live and store/forward modes are available. Store/forward mode automatically records an announcement and plays it back when the user replaces the telephone handset - particularly useful to prevent feedback.

SIP paging has been tested against common SIP servers including but not limited to; Cisco, Mitel, Avaya and Asterisk—please contact ASL to confirm interoperability with your system.

Live & Recorded Paging
For systems which require live broadcast capability, the MPS range of wall and desk mounted microphones enable zone-selectable paging. Microphones can also be configured to trigger pre-recorded messages and listen to broadcasts in any zone.

Network & Connectivity
VIPEDIA-12-PRO supports both copper and fibre Ethernet connectivity. In systems which do not include dedicated external Ethernet switches, four 100BASE-TX copper RJ45 ports greatly simplify interconnection of devices.

Two integrated 1GB SFP slots offer multi and single mode fibre connectivity, especially useful in larger systems where Cat5 restricts the distance between devices. Optional multi-mode (ASL p/n: VIPEDIA-NET-IF-MM) and single-mode (ASL p/n: VIPEDIA-NET-IF-SM) fibre modules should be ordered separately, according to system requirements.

GPIO
VIPEDIA-12 includes 12 on-board contact inputs and 12 outputs. Contacts are typically used to interface VIPEDIA-12 to fire alarm systems and external third party legacy systems. GPIO functions include audio routing, external system fault input , route busy indication and general fault indication.

On-board GPIO capability can be expanded using ASL's BMB01 if required. Each VIPEDIA-12 supports up to 9 BMB01 devices, each providing an additional 24 contact inputs and 12 contact outputs.

Ambient Noise Sensing
Ambient Noise Sensors (ANS) and Dynamic Ambient Noise Sensor (DANS) adjust output levels to ensure that if background noise is high, broadcasts remain audible and if it's low, broadcasts are made a comfortable level.

Standard ANS04-ES sensors fix the broadcast level for the duration of a broadcast. To adjust levels even during announcements, dynamic DANS01 noise sensors can be used. Up to 12 zones be controlled by ANS and up to 4 zones by DANS on each VIPEDIA-12-PRO. ANS interface using contact GPIO and DANS via a spare analogue audio input.

Monitoring
VIPEDIA-12-PRO provides full status monitoring and fault reporting including, amplifiers, microphones and associated peripherals.

ARCHITECTURES

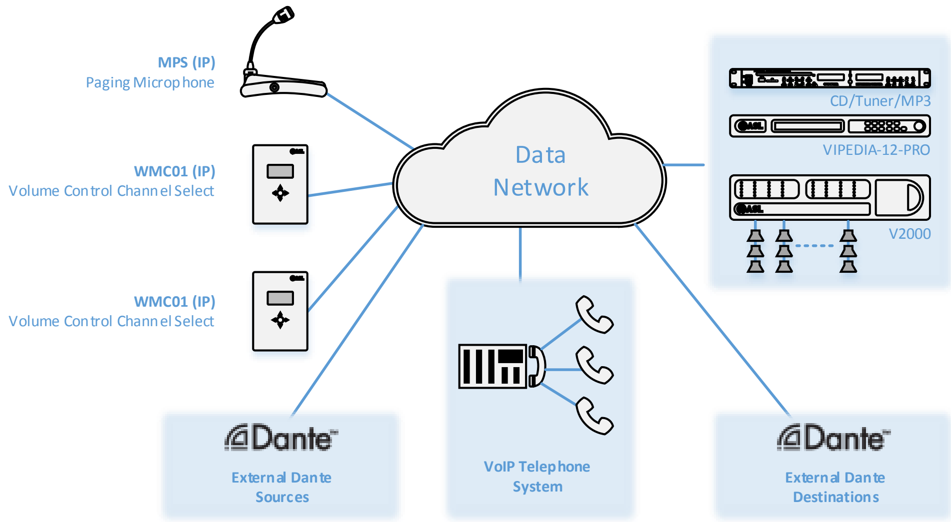
Standalone

In standalone mode, audio inputs can include paging microphones, SIP telephones, line/mic level audio signals or Dante channels from external systems such as AV or mixing devices.

Each VIPEDIA-12-PRO supports up to 12 analogue audio inputs, 32 Dante inputs and 12 output zones.

WMC01 wall controllers or the ASCII based Vipedia Control Protocol can be used to select audio sources or control broadcast levels.

Paging broadcasts can be made from any of ASL's microphone paging stations, such as the MPS01. Microphone buttons and the wall controller can also be configured to trigger pre-recorded messages.



Network System

In networked architectures, each VIPEDIA-12-PRO allows connection up to 12 local audio sources and supports 12 independent output zones.

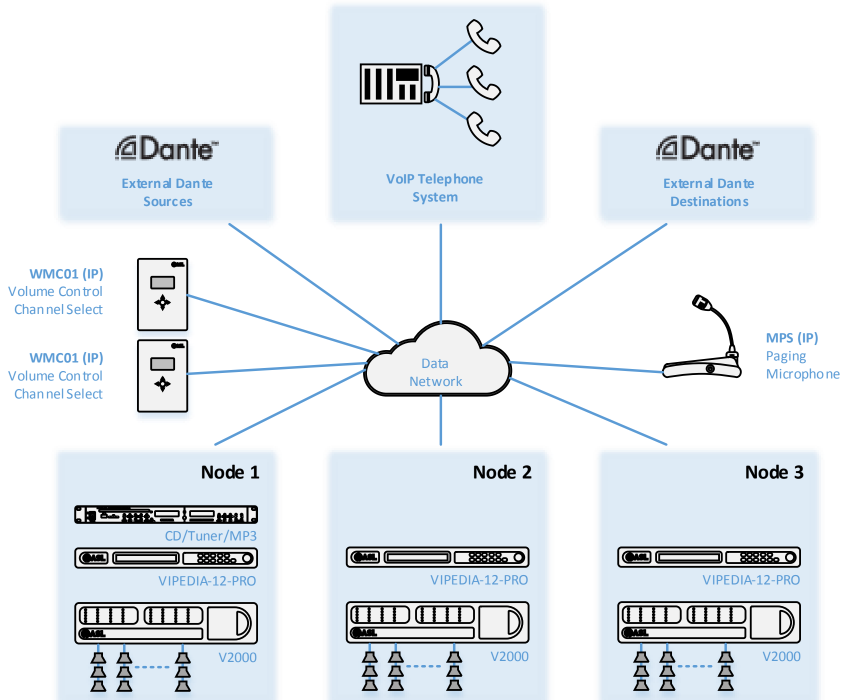
Local audio sources can include paging microphones or line/mic level audio signals. Broadcasts can also be made from external SIP/VoIP systems.

Up to 32 global sources can be configured to use Dante or Secure Loop networking making them available to all VIPEDIA-12-PRO devices within the system.

Existing external Dante sources can also be used - within the 32 global source limit.

Both local and global sources can be routed to any combination of outputs under the control wall controllers, microphones, touchscreen devices or the ASCII based Vipedia Control Protocol

To enable simple integration with external audio systems, analogue sources can also be permanently connected to Dante output channels.



VIPEDIA-12-PRO

Power Supply

Input Voltage	18—40 V DC
Current Consumption @24V (maximum)	695 mA
Current Consumption @24V (nominal)	650 mA
Current Consumption per SPF Interface	30mA

Ethernet Connectivity

100MB Copper	4 x 100BASE-T Ethernet
1GB Copper / Fibre	2 x SFP slot (modules optional)

Audio - General

Format	48kHz / 32-bit PCM
THD	<0.01% at 1 kHz
Crosstalk	>70 dB at 1 kHz
Residual Noise	<90 dBu (A)
Frequency Response	20 Hz to 20 kHz ±0.5 dB

Audio - Inputs

Analogue Input Channels	12
Input Sensitivity	-60 / - 40 / -20 / 0 dBu
Max Input Level	+20 dBu
Input Trim	–90 dB to +10 dB (1 dB steps)
Switchable HPF	20 to 500 Hz / Slope: 12 dB/oct
EQ	4 Band Parametric
Dynamics	Gate/Compressor/Limiter
Chime	Off / 1 note / 2 note / 3 note / Custom
Chime Level	-60 dB to +10 dB (1 dB steps)

Audio - Outputs

Analogue Output Channels	12
Nominal Output Level	0 dBu
Maximum Output Level	20 dBu
Output Impedance	660 Ω
Master Level	+10 to - 90 dB (1 dB steps)
Delay (per output)	1 ms to 5000 ms (1 ms steps)
EQ	10 Band Parametric
Dynamics	Limiter / Hard Clipper
Hardware Bypass Gain	-31.5 dB to 0 dB (1 dB steps)

I/O Interfaces

Inputs	12 x combined digital and analogue contacts
Outputs	12 x open-drain contacts
General Fault Relay	1
Voice Alarm Indicator Relay	1
IO Expansion Interface	1

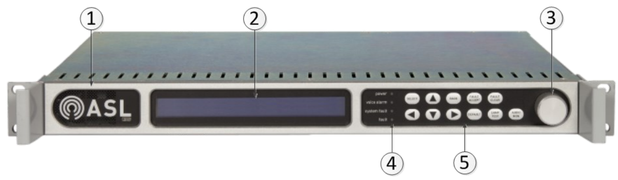
Mechanical

Dimensions	41.8 mm x 436 mm x 260 mm
Mounting	19-inch rack mounting (1U)
Weight	3.75 kg

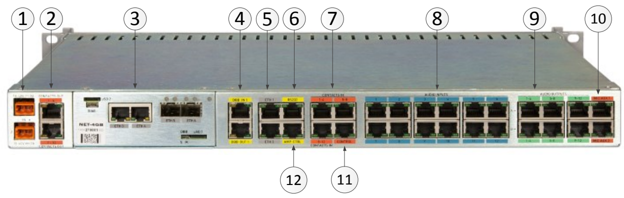
Environmental

Temperature (storage)	-20°C to +55°C
Temperature (operating)	-10°C to +55°C
Humidity	0% to 95% non-condensing
IP Rating	IP20

Front and Rear Panels



1. Fault Sounder and Audio Monitoring Loudspeaker
2. 2 x 40 backlit LCD Alphanumeric Display
3. Rotary Control for increment and decrement of menu items & volume control of monitor audio
4. Mandatory EN54 Indications
5. Menu Control Buttons



1. 18V -40V Dual DC Power Supply
2. Contact Outputs 1 to 12
3. Optional Fibre SPF Modules (with Dante Support)
4. DBB Expansion Ports
5. Dual Ethernet Ports
6. RS232 Port
7. Contact Inputs 1 to 8
8. Microphone / Audio Inputs 1 to 12
9. Audio Outputs 1 to 12 (A&B)
10. Hardware Bypass Emergency Microphone and Listen-in Interfaces
11. Fault & Voice Alarm Relays and ASL BMB01 Serial Interface
12. Amplifier Control and Monitor Interface (Audio-CAN)

This equipment is designed and manufactured to conform to the following EC standards:
EMC: EN55103-1/E1, EN55103-2/E5, EN50121-4, ENV50204
Safety: EN60065

Designed and manufactured by

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