LUNA-U
Unified audio matrix processor

Highlights:

- A scalable mixture of analogue and network channels
- 12 Mic/line inputs and 8 balanced outputs
- 12 GPIO inputs and 12 outputs
- Expandable up to 32 x 32 Dante™/AES67 channels (with optional extension card)
- Compatible with all Dante™/AES67 paging/in & output devices (Optional)
- 32 mono or 16 stereo flexible zones
- Works out of the box thanks to easy scalable architecture
- Unmatched flexibility for input, output & zone mapping
- Single design and control platform - AUDAC Touch™
- AudioBridge for flexible audio transfer for all inputs & outputs
- Auto-update (OTA) feature

Product information:

The LUNA-U is the next-generation audio matrix processor, offering highly flexible and scalable system solutions for audio distribution from medium-sized to the largest enterprise solutions. The internal structure provides unseen flexibility and a vast number of possibilities for an audio distribution system. The powerful DSPs (dual SHARC) combined with 32-bit ADC and DAC converters allow the most extensive signal processing with superb audio quality.
It includes 12 Mic/Line Inputs and 8 Line level outputs. Additionally, up to 32 Dante™/AES67 I/O Channels are available with a single ethernet port, allowing the integration of the system with any Dante™/AES67 compatible audio over IP I/O unit.
The flexible architecture allows fully flexible mapping of the signal path to any of the 32 mono or 16 stereo zones, while the powerful DSP resources provide unique processing features on all input channels, output channels, and zones. Dante-enabled paging microphones and smart wall panels make the complete system solution even more flexible and unique.
The internal Linux core brings the entire control of this system and family members together, while also accommodating messaging, event scheduling, and implementation possibilities for further Linux-based functions. Besides the full network control, other options like RS-232 and RS-485 are also implemented for third-party control. For your system solution, you can easily incorporate the control of the entire system and compatible devices into your user interface by using our simplified design and system control platform, AUDAC Touch™.

Certification:

Dante™
Option available

Additional Inputs:

USB TCP-IP

Applications:

- Government facilities
- Education
- Hotels
- Corporate
- Sporting facilities

System specifications:

Inputs	Mic / Line inputs	12
	GPIO Inputs	12 (in)
	USB Inputs (Type C)	Front (file transfer)
		Mass storage for messages
		Rear (file transfer)
Outputs	Line outputs	8
	GPIO Outputs	12 (out)
	Fault	NO & NC relay contact
Network audio I/O		Optional Dante™/AES67 (Up to 32x32 Channels with SL-NAC8)
Zones		32 Mono
		16 Stereo
Configurable settings	ALC (Automatic Gain control)	Yes
	AEC (Acoustic Echo cancellation)	No
	Integrated event scheduler	Yes
	Phantom power on inputs	+48 V DC
	WaveTune (input & zone)	7-band EQ
	WavePreset (output)	12xBiquad +LP+HP
	Delay (output)	2000 ms (target)
	Mono / stereo zones (configurable)	Yes
	Mixing	Yes
	Talkover	Yes
	Paging	Yes
	Priorities	4 each zone
	Integrated generator	Sine or Pnoise or Wnoise
	Output Volume offset	Yes
	Others	Antiphase, bass & treble, input volume, ...
DSP Processor		Dual core SHARC
		ARM Linux core
Configuration		Audac Touch™
Controls & indicators	Front panel	2.8" LCD Display with rotary encoder
	Interface ports	RS-232
		RS-485
		Gigabit Ethernet (RJ45 primary)

Power	Supply	100 ~ 240 V AC / 50 ~ 60 Hz
		24 V DC

Product Features:

Dimensions	482 x 44 x 335 mm (W x H x D)
Mounting	19″
Unit height	1 HE
Construction	Steel
Colours	Black (RAL9005)

Architects' and Engineers' Specifications:

The network audio processor shall be a flexible and scalable mixture of analog and network channels. It shall have dual SHARC DSPs with ARM core based on Linux combined with 32-bit ADC and DAC converters, allowing the most extensive signal processing with superb audio quality. It shall include 12 input channels and 8 output channels. Phantom power shall be available on all inputs. All analog inputs shall be able to carry line- and microphone-level signals. It shall have 8 x 8 Dante™/AES67 network audio I/O channels with an extension networked audio module and shall have the capability of increasing up to 32 x 32 Dante™/AES67 network audio I/O channels with additional licenses. It shall have a single Ethernet port, allowing the integration of the system with any Dante™/AES67 compatible audio over IP I/O units when an extension networked audio module is installed. It shall have USB ports on the front and rear panels, and they shall be Type C. These inputs shall be used for file transfer and mass storage for voice files being played on announcements. Pre-gain shall be available for microphone/line level adjustment on physical inputs. The available DSP processing functionality on the inputs shall include Automatic Gain Control (AGC), high pass/low pass filtering, 7-band parametric equalizing, anti-feedback, 100 milliseconds of delay possibility, and trigger threshold. It shall include 32 mono or 16 stereo zones. Up to 16 inputs or zones shall be mappable via an Audio Bridge which shall be mixable in each of the available zones using the zone mixer feature. Each zone shall include talk over, paging volume control, four levels of priority, 7-band parametric equalizing, and compressor/limiter functionality. The output channels shall include gain setting, high pass/low pass filtering, 12-band parametric equalizing, 2000 milliseconds of delay possibility, and antiphase functionality. The device shall have 12 GPIO inputs and 12 GPIO outputs. The GPIO ports shall have active low/high triggering, 0-33V configurable threshold level triggering, and edge triggering in the input configuration and open drain in the output configuration. The device shall be controllable intuitively through the 2.8" LCD display with a rotary encoder on the front panel or by using TCP/IP, RS-232, and RS-485 connection possibilities. A total system control platform shall be freely available and compatible with a wide variety of operating systems, including Android, iOS, Windows, and Mac. This platform shall allow simplified design and management of deployment with functional and informative widgets and dashboards in combination with third-party audio and video equipment control possibilities. It shall allow quick access to features such as I/O volume, mapping, mixing, source selection, and others. The device shall have an auto-update (OTA) feature. The power supply shall be a switching mode type operating on a 100~240 V AC/50~60 Hz mains network. Additionally, an emergency power inlet shall be provided to keep the system running on 24V emergency power when the main power is shut down. It shall be equipped with a removable power cord with a standard Schuko (CEE 7/7) AC plug. The connector on the chassis shall be a fused IEC C14 type. The network audio processor shall have a form factor of 19″ and 1 RU.

Technical drawings: