AHM-32

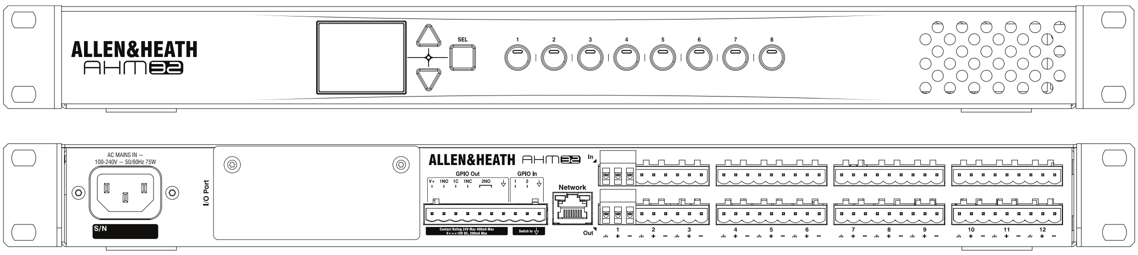
Technical Datasheet

AHM-32 is an audio matrix processor for sound management and installation. It is designed for audio distribution, paging, conferencing and speaker processing in a multitude of environments including corporate, hospitality, education, event and multi-purpose venues, retail, theatres, cruise ships and sports venues.

The AHM-32 processor is complemented by an extended ecosystem of remote audio expanders, remote controllers, interfaces, apps and software.

A family of portable, rack-mountable or wall-mount audio expanders is available with a choice of proprietary point-to-point Layer-2 or Dante transport protocols.

A range of IP remote controllers is available for volume control, music source selection, preset recall and more. AHM can also integrate with third party devices over GPIO, TCP/IP, or industry standard control systems. The Custom Control editor and app from Allen & Heath offer more control options and tailored user interfaces for multiple users and device types, with kiosk and BYOD capability.



Features

- 32x32 processing matrix
- 12x12 local analogue I/O
- I/O Port for expansion or audio networking, up to 128x128
- Dante 96kHz optional cards (AES67 and DDM ready)
- 32 configurable processing outputs – up to 32 mono / 16 stereo zones
- Sound management tools
    - 4x Automatic Mic Mixer
    - AEC (Acoustic Echo Cancellation)*
    - ANC (Ambient Noise Compensation)
    - Priority ducking
    - 8-band PEQ, dynamics and delay on every input and zone
    - Speaker processing with x-over filter, delay, limiter and PEQ

  *with optional module

- 96kHz FPGA core with ultra-low latency
- Compatible with Allen & Heath IP1, IP6, IP8 remote controllers
- 2x2 local GPIO plus networkable GPIO interface
- Front panel screen and 8x programmable SoftKeys
- 16 user profiles
- Event scheduler
- Internal stereo playback

The unit shall be a 1u rack-mountable digital matrix processor, capable of 32 input channels and 32 output channels, all independently assigned.

The unit shall operate at 96kHz sample rate and employ FPGA technology for digital signal processing. The system latency from analogue input to output shall not exceed 1ms.

All input channels shall be configurable mono/stereo and have access to any local or remote input. Output channels shall be configurable as mono/stereo zones or as speaker processing outputs with 2, 3 or 4-way Crossovers, allowing up to 32 mono zones / 16 stereo zones, or any combination of zones and speaker processing outputs not exceeding 32 total channels.

All input channels shall provide the following processing: Trim, Polarity, Gate, Insert point, 8-band Parametric EQ, Compressor, Delay and Automatic Mic Mixing (AMM).

All zones shall provide the following processing: Source Selector, Insert point, 8-band Parametric EQ, 28-band GEQ, Compressor, Delay, Ambient Noise Compensation (ANC) and Limiter.

All speaker processing outputs shall provide the following processing: Crossover filters with selectable filter type and slope, PEQ/GEQ, Delay and Limiter.

All output channels shall be routable to any local or remote output.

The 8-band Parametric EQ shall provide Bell, Constant Q, Shelving, LPF, HPF and Notch filter types selectable per band.

The unit shall have 12 balanced inputs on pluggable Phoenix terminal blocks. Each input shall have independent gain control with +60dB of gain, a -20dB active PAD and +48V phantom power.

The unit shall have 12 balanced outputs on pluggable Phoenix terminal blocks with a nominal level of +4dBu.

The routing matrix mixer shall be capable of mixing all inputs to all zones, as well as all zones to other zones.

The unit shall provide Automatic Mic Mixing (AMM) of up to 32 microphone sources into 1, 2 or 4 zones. The AMM shall be capable of running in classic gain sharing mode or optionally as a NOM (Number of Open Microphones) algorithm.

The unit shall offer a slot for optional processing modules including Acoustic Echo Cancellation.

An RJ45 Control Network port shall be provided on the rear of the unit for connection to System Manager software, IP remote controllers, Custom Control app and TCP control.

One 128x128 I/O port for optional digital interface modules shall be provided. A Dante optional module shall provide a minimum of 32x32 I/O at 96kHz, and be compliant with AES67 and Dante Domain Manager. An SLink optional module shall be available for Ethernet audio expansion, supporting multiple Audio-over-Ethernet protocols and providing access to up to 128x128 I/O.

The unit shall provide the facility to save 500 presets. The presets shall be nameable and a descriptive text entry per preset provided. A crossfade of up to 20 seconds shall be available to apply to any combination of Inputs, Zones, Groups, Input/Zone Crosspoints and Zone/Zone Crosspoints.

The unit shall provide the facility to save 50 events. The events shall be nameable and should allow for the scheduled recall of presets at a specified time on specific days, or every day, with the option for the event to be triggered repeatedly or just once.

The unit shall allow the creation and storage of up to 16 user profiles, each with an editable name and password.

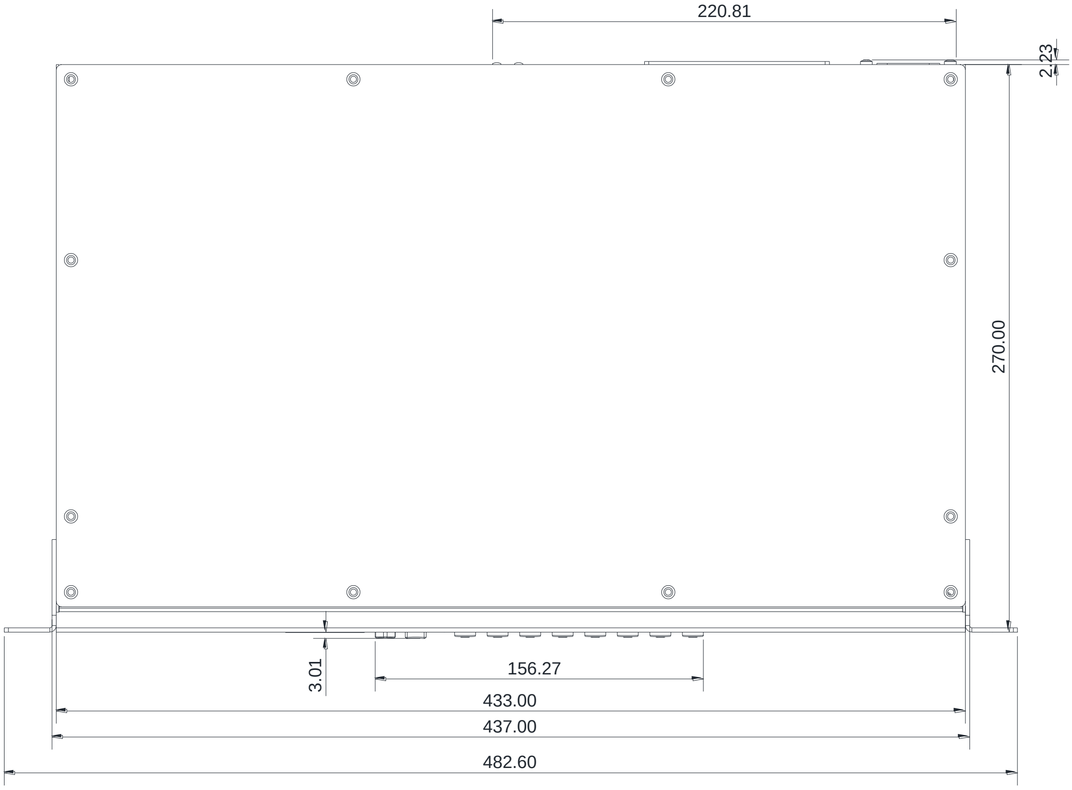
The unit shall allow the connection of two general purpose inputs, and two general purpose relay outputs, via pluggable Phoenix connectors on the rear of the chassis. Each input connector shall allow analogue control of Mutes, Levels, Preset Recall, Custom MIDI via a 0-10V control signal. Output 1 shall support normally closed and normally open operation, and output 2 shall support normally open operation. The outputs shall be configurable to respond to Mutes, Preset Recalls, and Level Sensing. An optional 8x8 networkable GPIO interface shall be available for expansion of the GPIO functionality.

Networkable, PoE-enabled remote controllers shall be available to complement the unit, including wallplate controllers in both US and EU formats, and desktop controllers with a minimum of 8 motorised faders and 8 LCD displays.

The unit shall have an integrated power supply accepting AC mains voltages of 100-240V, 50/60Hz, 70W max via an earthed 3-pin IEC male connector mounted on the rear chassis.

The unit shall be the Allen & Heath AHM-32.

Dimensions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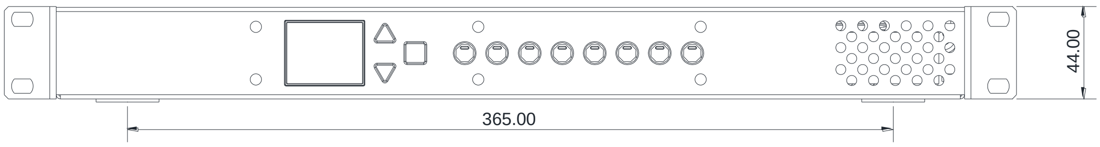
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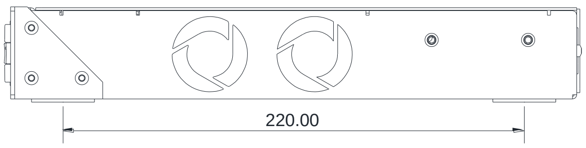
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Technical specs

Inputs

Mic/Line Inputs	Balanced, +48V phantom power
Mic/Line Preamp	Fully recallable
Input Sensitivity	-60 to +15dBu
Analogue Gain	+5 to +60dB, 1dB steps
Pad	-20dB Active PAD
Maximum Input Level	+30dBu (PAD in)
Input Impedance	>3kΩ (Pad out), >8kΩ (Pad in)
Mic EIN	-127dB with 150Ω source

Outputs

Analogue Outputs	Balanced, Relay protected
Output Impedance	<75Ω
Nominal Output	+4dBu = 0dB meter reading
Maximum Output Level	+21dBu
Residual Output Noise	-92dBu (muted, 22-22kHz)

Dimensions and Weights

Unboxed	Width x Depth x Height x Weight
AHM-32	482.6mm x 270mm x 44mm x 4kg (19” x 10.6” x 1.7” x 8.8lbs)
Boxed	
AHM-32	555 x 385 x 150 mm x 5.3kg (21.8” x 15.2” x 5.9” x 11.7lbs)

System

Measured balanced XLR in to XLR out, 20-20kHz, +5dB Gain, Pad out, signal @ 0dB (meter)

Dynamic Range	108dB
System Signal to Noise	-92dB
Frequency Response	20Hz - 20kHz +0/-0.5dB
THD+N (analogue in to out)	0.005% @ +16dBu output, 1kHz +5dB gain
Headroom	+18dB
Sampling Rate	96kHz +/- 20 PPM
Acoustic Noise	Typical loading, 23 deg C ambient
No I/O Port or AEC installed	<29dBA
With I/O Port and AEC installed	30dBA

Operating Temperature Range	0 deg C to 40 deg C (32 deg F to 104 deg F)
Mains Power	100-240V AC, 50-60Hz, 70W max

Processing specs

Input Processing

32 Input Channels	Configurable mono or stereo
Trim	+/-24dB digital trim
Polarity	Normal/Reverse
Stereo Width Control	L/R, R/L, L -Pol/R, R -Pol/L, Mono, L/L, R,R, M/S
Gate	
Sidechain	Self-key or source selectable, with 12dB/octave Lo-Pass and Hi-Pass
Threshold	-72dBu to +12dBu
Depth	0 to 60 dB
Attack	50us to 300ms
Hold	10ms to 5s
Release	10ms to 1s
Insert	In/Out, +4dBu/-10dBV level
PEQ	
Type	8-Band fully parametric, +/-15dB
Band 1 - 8	Selectable LF/HF Shelving, Bell (variable or constant Q), Hi-Pass / Lo-Pass
Bell Width	0.50 – 6.00 Q
Shelving Type	Classic Baxandall
Hi-Pass, Lo-Pass Filter	12dB/octave
Compressor	Peak or RMS sensing
Sidechain	Self-key or source selectable, with 12dB/octave Lo-Pass and Hi-Pass
Threshold	-46dBu to 18dBu
Compressor parameters	Threshold, Ratio, Attack, Release
Delay	Up to 683ms

Zone Processing

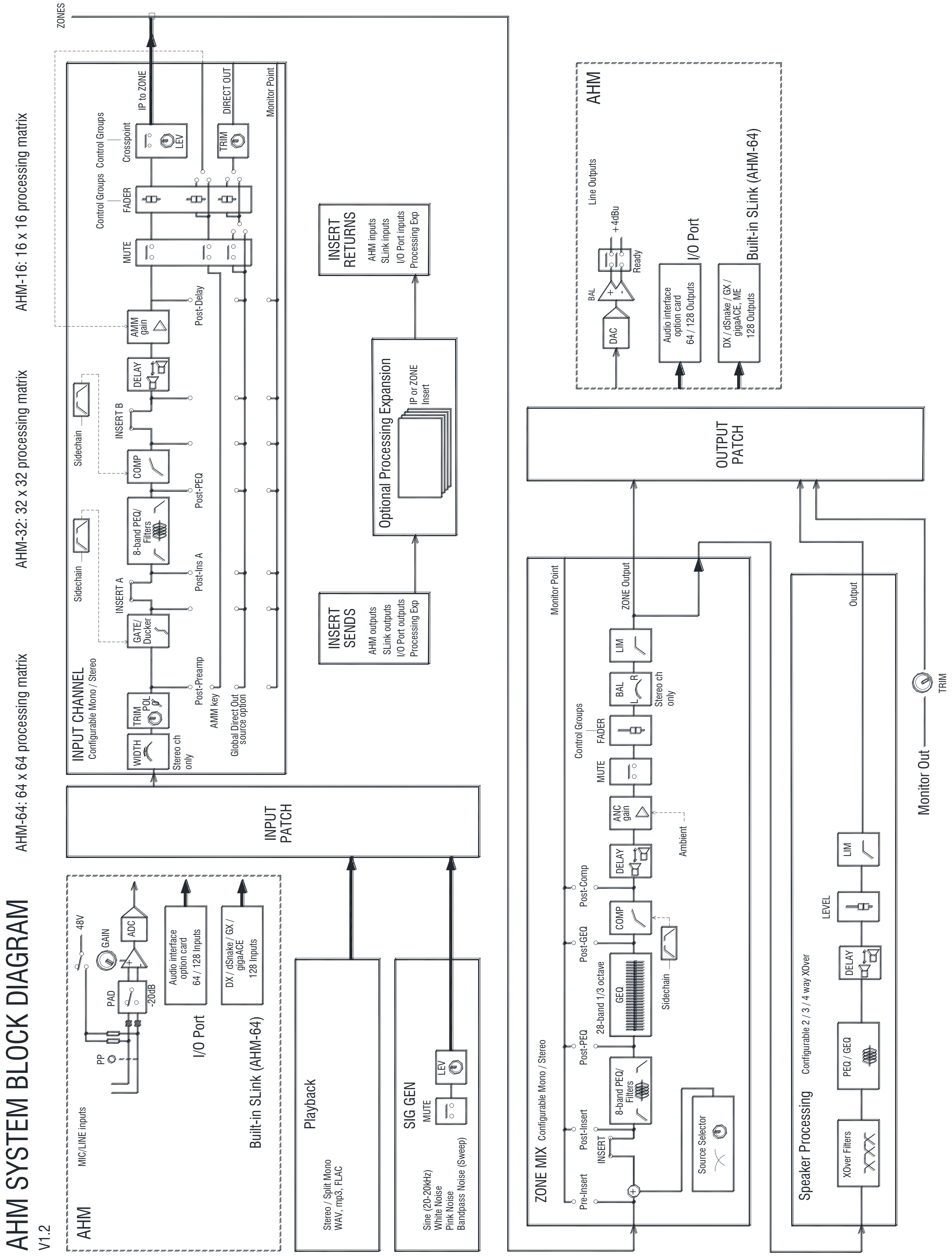
Up to 32 Zones	Configurable mono or stereo
Source Selector	Up to 20 sources, variable level, Fade In and Fade Out time <20s
Insert	In/Out, +4dBu/-10dBV level
GEQ	28 bands 31Hz -16kHz, +/-12dB, constant-Q
PEQ	See Input Processing
Compressor	See Input Processing
Delay	Up to 683ms
ANC	
Ambient Level	Selectable source and metering point, Gain Differential -18dB to 40dB
Gap	Selectable source and metering point, Threshold -62dB to -20dB, Time 0-5000ms
Gain Element	Min / Max Gain, Rate 0-30dB/s
Limiter	Variable Threshold, Attack and Release

Speaker Processing

Crossovers	Configurable 2, 3, 4 way
Filters	Asymmetrical, selectable 1st order, Butterworth 12/18/24 db/octave, LR 12/24 dB/octave
EQ	4-Band fully parametric, or 28 band GEQ
Delay	Up to 683ms
Limiter	See Zone Processing

AMM

Channels (AHM-16)	1x16
Channels (AHM-32)	1x32, 2x 16 or 4x 8
Modes	D-Classic gain sharing or NOM



AHM SYSTEM BLOCK DIAGRAM
V1.2
AHM-64: 64 x 64 processing matrix
AHM-32: 32 x 32 processing matrix
AHM-16: 16 x 16 processing matrix
ZONES
AHM
MIC/LINE inputs
PP
PAD
-20dB
GAIN
48V
ADC
I/O Port
Audio interface option card 64 / 128 Inputs
Built-in SLink (AHM-64)
DX / dSnake / GX / gigaACE 128 Inputs
Playback
Stereo / Split Mono WAV, mp3, FLAC
SIG GEN
MUTE
LEV
Sine (20-20kHz) White Noise Pink Noise Bandpass Noise (Sweep)
INPUT PATCH
INPUT CHANNEL
Configurable Mono / Stereo
WIDTH
Stereo ch only
TRIM
POL
Post-Preamp
AMM key
Global Direct Out source option
GATE/ Ducker
Sidechain
INSERT A
Post-Ins A
8-band PEQ/ Filters
Post-PEQ
COMP
Sidechain
INSERT B
DELAY
AMM gain
Post-Delay
MUTE
FADER
Control Groups
Control Groups
Crosspoint
LEV
IP to ZONE
TRIM
DIRECT OUT
Monitor Point
INSERT SENDS
AHM outputs SLink outputs I/O Port outputs Processing Exp
Optional Processing Expansion
IP or ZONE Insert
INSERT RETURNS
AHM inputs SLink inputs I/O Port inputs Processing Exp
ZONE MIX
Configurable Mono / Stereo
Pre-Insert
INSERT
Post-Insert
8-band PEQ/ Filters
Post-PEQ
28-band 1/3 octave GEQ
Post-GEQ
COMP
Sidechain
Post-Comp
DELAY
ANC gain
Ambient
MUTE
FADER
Control Groups
BAL
Stereo ch only
LIM
ZONE Output
Monitor Point
Source Selector
Speaker Processing
Configurable 2 / 3 / 4 way XOver
XOver Filters
PEQ / GEQ
DELAY
LEVEL
LIM
Output
Monitor Out
TRIM
OUTPUT PATCH
AHM
Line Outputs
DAC
BAL
+4dBu
Ready
I/O Port
Audio interface option card 64 / 128 Outputs
Built-in SLink (AHM-64)
DX / dSnake / GX / gigaACE, ME 128 Outputs