

Avantis

64 X 42 MIXER

The Avantis digital mixer provides 64 mono input channels, 12 stereo FX returns and 42 flexible mix buses.

The third mixer based on our 96kHz XCVI FPGA engine, Avantis puts all our next-generation technology in a standalone 64 channel / 42 configurable bus console, with dual Full HD touchscreens, a super-flexible workflow with Continuity UI, extensive I/O options, add-on processing from our flagship dLive mixing system, and a rugged full metal chassis.

Fully compatible with all the dLive I/O cards, the full 'Everything I/O' range and IP controllers.



Key Features

- 64 x 42 XCVI core
- Rugged all metal chassis
- Fully assignable layout
- 144 fader strips
- Full 'Everything I/O' range compatible

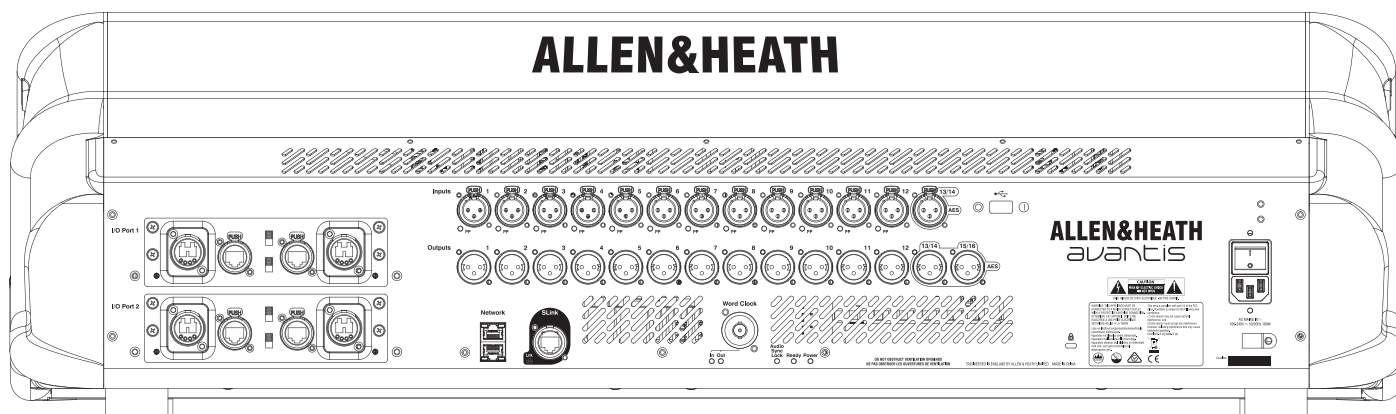
Benefits

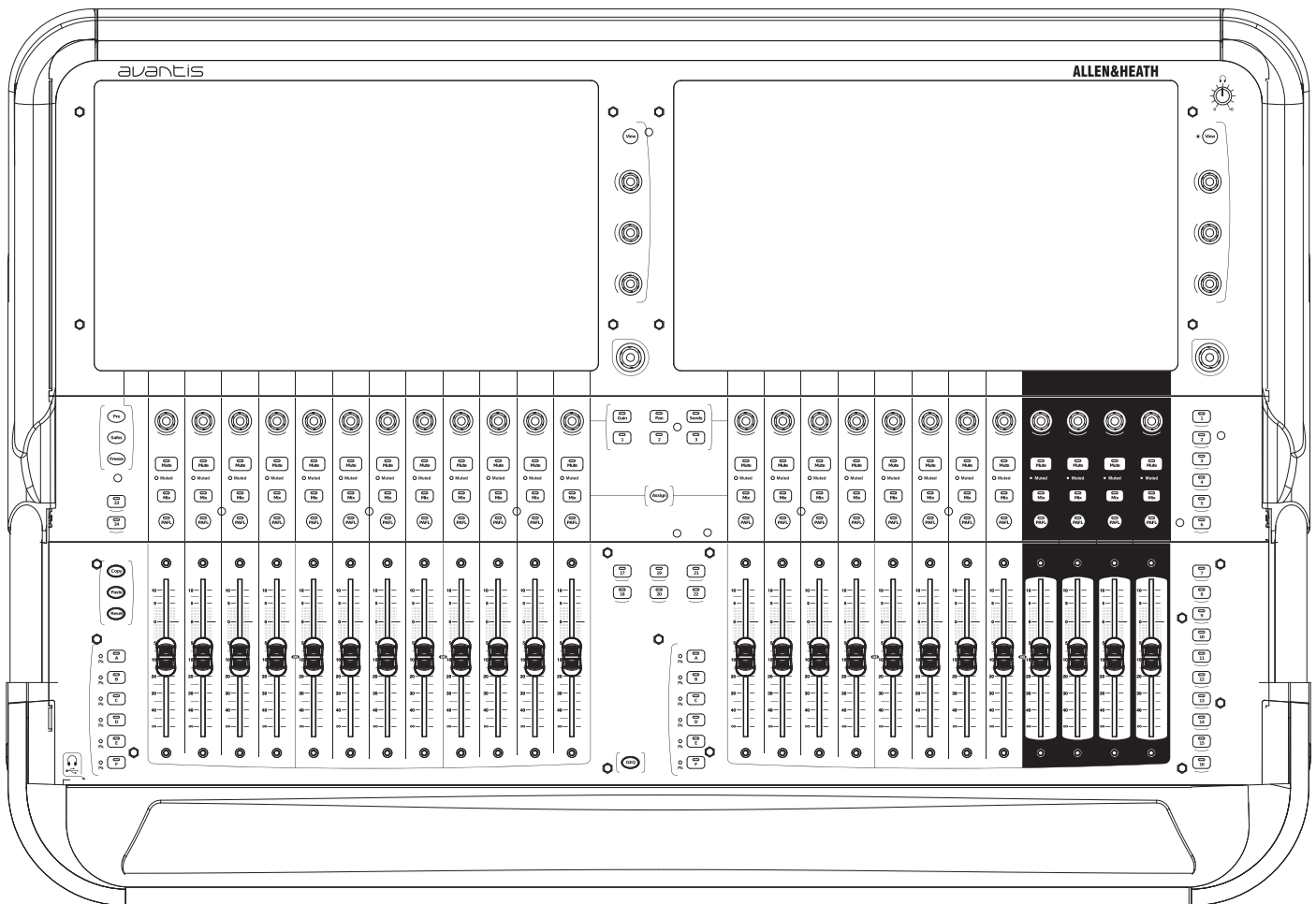
- 2 128x128 option slots
- Compatible with IP controllers
- dPack available
- 64 channels of AMM

Technical Datasheet

Overview

- **XCVI** FPGA core
 - 96kHz sample rate
 - Variable bit depth
 - 96bit accumulator
 - Class leading, ultra-low latency (0.7ms)
- Twin 15.6" Full HD capacitive touchscreens with gesture control
- Innovative compact design with super-tough, lightweight full-metal chassis
- 64 Input Channels with full processing (LPF/HPF, Gate, PEQ, Comp, Delay)
- Configurable 42 bus architecture with full processing (PEQ, GEQ, Comp, Delay) on Mix Outputs.
- **DEEP** Processing Ready
- 144 fader strips (24 faders, 6 layers) with optional dedicated master section
- **SLink** EtherCON connection for remote audio using gigaACE, GX, DX, ME and dSnake protocols.
- 12 Local Mic/Line Inputs (XLR)
- 12 Local Line Outputs (XLR)
- 1 Stereo AES Input (XLR)
- 2 Stereo AES Outputs (XLR)
- 2 I/O Ports, both capable of 128x128 operation at 96kHz – for optional cards inc. Dante, MADI and gigaACE
- 8 Mute Groups
- 16 DCA Groups with DCA Spill
- 12 Stereo FX with dedicated FX Returns
- 24 Assignable SoftKeys
- 6 Assignable Custom Rotaries
- Comprehensive metering with multipoint peak detection
- Integrated Fader Illumination
- Built-in signal generator, RTA and spectrogram
- AMM (Automatic Mic Mixer) up to 64 channels across 1,2 or 4 zones
- Quick copy/paste/reset for parameters
- User Permissions to restrict operator access
- 500 Scene memories per show
- Channel Safes, Global and per Scene Recall Filters
- USB stereo recording and playback
- USB transfer of Scenes, Libraries, Shows
- MIDI Control Driver for MIDI control via TCP/IP
- Wireless remote mixing apps for iPad
- BNC Word Clock connector (input or output)
- 1/4" TRS and 3.5mm Headphone outs with dedicated analogue volume control





A&E Specifications

The mixer shall be a compact digital mixer capable of standalone operation. It shall be capable of 64 input channels, 42 mix busses and 400 soft patchable system inputs and outputs, all independently assigned. There shall be 12 stereo rack FX engines with dedicated stereo returns and 16 DCAs. The mixer shall contain the mix engine. The mix engine shall run at 96kHz sampling rate and employ FPGA technology for digital signal processing. The system latency from analogue input to output shall not exceed 0.7ms.

The surface shall include 24 motorised touch-sensitive faders in two banks, each with 6 layers accessed by dedicated keys. Fader illumination shall be integrated into the armrest of the mixer and shall be adjustable via the software user interface. All fader strips shall be configurable as input channels, mixes, FX sends, FX returns, DCA masters or MIDI strips. Each fader strip shall have dedicated PAFL, Mix and Mute buttons with indicators and a rotary controller. 6 buttons with indicators shall provide global mode for the rotary controls being Gain, Pan, Sends and up to 3 custom functions. Send levels to mixes shall also be adjustable via the faders.

Dedicated keys shall be provided for quick Copy/Paste/Reset of mixes and processing parameters, pre/post and mix assignments, scene safes, freeze-in-layers and GEQ fader flip. Further physical controls shall be user assignable, including two sets of 3 rotary encoders and 24 user-defined keys.

Two 15.6" Full-HD (1080p) capacitive touchscreens shall be provided for access to the channel bank view, channel name

blocks, channel processing, meters, FX, I/O, scene management, setup, utilities and diagnostics. The channel name blocks shall display each visible channel's name, colour, type, number, status, pan, gain, trim, metering, processor gain reduction and other parameters. Touching a channel name block shall select a channel, with the capability to select one channel per touchscreen.

22-segment meters in the channel name block shall indicate channel and mix levels. An additional onscreen 22-segment meter shall be provided for monitoring the PAFL signal. A default Mains to PAFL sub-mix shall be provided.

Comprehensive and customisable input, output, and FX channel and RTA metering shall be provided on-screen in a dedicated meters interface.

Local analogue inputs shall be provided on 12 balanced XLR sockets and connect to fully recallable digitally controlled preamplifiers of high quality. These shall be able to provide up to +60dB of gain, industry standard 48V phantom power, and include a switchable -20dB Pad to allow a maximum input level of +30dBu. Local analogue outputs shall be provided on 12 balanced XLR sockets. These will have a nominal line output of +4dBu and a maximum output of +22dBu.

One stereo AES3 input with SRC (32-192kHz) shall be provided on one XLR socket. Two stereo AES3 outputs with SRC (96kHz, 48kHz, 44.1kHz) shall be provided on two XLR sockets. There shall be an option to bypass the Sample Rate Conversion on the digital inputs and outputs via the software user interface.

There shall be a local "SLink" Ethernet audio expansion port with locking EtherCON connector, supporting multiple protocols (gigaACE, DX, GX, dSnake, ME) and providing access to a maximum 128x128 digital channels, connected over a single cable 'digital snake' and allowing remote preamp control of Allen & Heath Remote Audio Units, as well as connection to Allen & Heath ME Personal Mixing Systems.

Two 128-channel I/O ports for optional digital interface modules shall also be provided. Optional interface modules shall include; MADI, Dante, Waves SoundGrid, gigaACE and AES/EBU.

A BNC connector shall be provided on the unit for Wordclock input or output, switchable in software.

All input channels shall contain the following processing: Trim, Polarity, High Pass Filter, Low Pass Filter, two Insert points, Gate, Parametric EQ, Compressor and Delay.

All output mix channels shall contain the following processing: External input, Polarity, Trim, Insert, Parametric EQ, and Graphic EQ with RTA and fader-flip mode, Compressor and Delay.

The channel processing shall also include high quality emulations of classic preamps, compressors and GEQs. These emulations shall be insertable on the fly on any channel without using the internal FX and without the need of any external plug-ins. The use of any emulation shall not result in added latency or affect processing abilities elsewhere within the system.

All signal delays in the system shall be adjustable in milliseconds, metres, feet and samples.

12 user-assignable effect racks shall be provided with a library of factory preset FX emulations. The FX racks shall be individually configurable as send/return from a channel or FX/Mix or inserted into input or output channels. The mixing system shall provide the options of FX racks processed signals to be returned to other mixes via dedicated FX input channels, with PEQ, assigned as sources to input channels, daisy-chained to another FX rack by internal patching, or routed to an output socket or interface channel.

There shall be 16 DCA groups, with DCA Spill functionality, and 8 Mute groups.

The mixer shall provide Automatic Mic Mixing (AMM) of up to 64 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.

A global source option for the direct out of each input channel shall be provided in the routing screen. The tap-off point shall be adjusted to the following positions in the processing path: post Preamp, post LPF, post Gate, post Insert A return, post PEQ, post Compressor, post Insert B

return and post Delay. There shall be further global options to follow Fader, DCA and Mute. Direct outputs shall be assignable via the mixer soft patch bay.

A Talkback facility shall be provided with the ability to send to any output mix with on screen status indication. An option to enable talkback latching and HPF shall be provided.

A signal generator shall be provided with the ability to send a variable level signal to any input and output mix with visual assignment status on-screen. The following types of signals shall be available: Sine, White Noise, Pink Noise, and Band-Pass.

The mixer shall provide two gigabit Ethernet (1,000 Mbit/s) port for Cat5 cable connection to a computer for MIDI over TCP/IP control of mixer parameters via a wireless router (access point) for live mixing control, and the mixing system shall include application software for tablet devices connected via a wireless network router to the LAN port.

Input and output channel processing and parameters in the mixer shall be saved on demand as a user library item for recall in other channels. All library items shall be archived with the show-file. Library items shall be transferrable to USB drive as portable data to be used in other systems.

The mixer shall provide the facility to save 500 scenes. The scenes shall be name-able and a descriptive text entry per scene provided. A comprehensive table of Scene Safes shall be provided to prevent selected items from being changed from their state when the safe was enabled. A comprehensive scene filter shall be provided per scene to Allow / Block each parameter saved in a scene from being changed as that scene is recalled. A Cue List of scenes shall be provided to facilitate a running order of scenes from the master list. A crossfade of up to 20 seconds shall be adjustable per scene. Scene recall may be embedded into a scene with the option to delay the start of the embedded scene recall by up to 4 minutes.

An option shall be provided for password protection for log-in of several users with different levels of system access and permissions. A specific scene may be chosen to be recalled per change of user-login if desired.

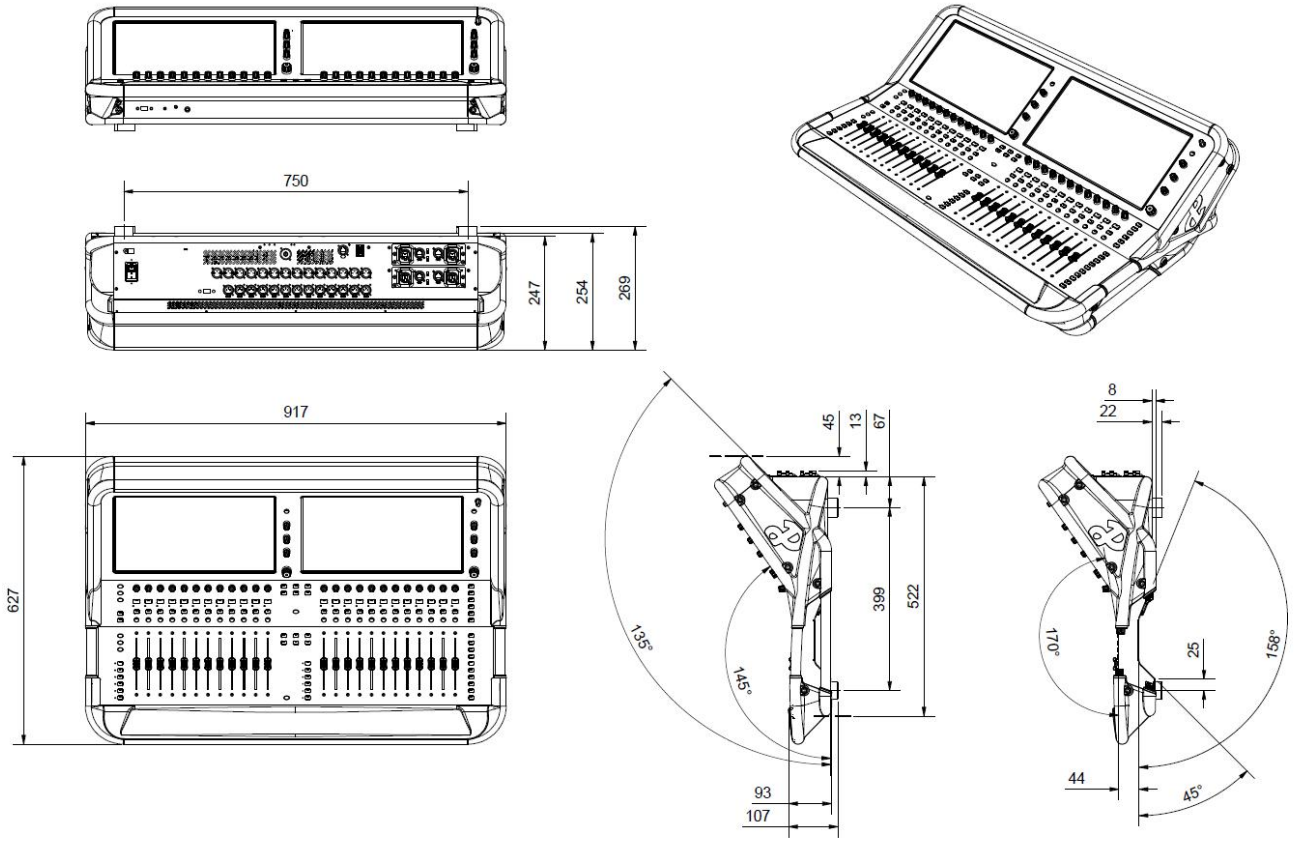
The mixing system shall periodically record all current settings and return the mixer to that state after reboot following a power-cycle.

The mixing control surface shall have a built-in power supply accepting AC mains voltages of 100~240V, 50/60 Hz, 150W max via an earthed 3-pin IEC male connector mounted on the rear chassis. A Two Pole Push-Button switch shall be provided near the mains input.

Recommended operating temperature for the mixer shall be 5 to 40 degrees Celsius.

The mixer shall be the Allen & Heath Avantis.

Dimensions



Mixer Specifications

Inputs

Mic/Line XLR Inputs	Balanced XLR, +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	>4k Ω (Pad out), >10k Ω (Pad in)
Mic EIN	-127dB with 150 Ω source
Phantom Power indication	Per socket, internal or external phantom power sensing, triggered at +24V
Digital Inputs	AES3 2 Ch XLR, 2.5Vpp balanced terminated 110 Ω SRC range 24 bit, 32k - 192kHz, with bypass option

Outputs

Analogue XLR Outputs	Balanced, Relay protected
Output Impedance	<75 Ω
Nominal Output	+4dBu = 0dB meter reading
Maximum Output Level	+22dBu
Residual Output Noise	-95dBu (muted, 22-22kHz) -90dBu (muted, 0-80kHz)
Digital Outputs	AES3 2 Ch XLR, 2.5Vpp balanced terminated 110 Ω 96kHz sampling rate, globally switchable to 48kHz, 44.1kHz

System

Measured balanced XLR In to XLR Out, 22-40kHz, minimum Gain, Pad out	
Dynamic Range	109dB
System Signal to Noise	-92dB
Frequency Response	20Hz - 30kHz (+0/-0.8dB)
THD+N (analogue in to out)	0.0015% @ +16dBu output, 1kHz 0dB gain
Headroom	+18dB
Internal operating Level	0dBu
dBFS Alignment	+18dBu = 0dBFS (+22dBu at XLR output)
Meter Calibration	0dB meter = -18dBFS (+4dBu at XLR out)
Meter Peak indication	-3dBFS (+19dBu at XLR out)
Sampling Rate	96kHz

Operating Temperature Range	0 deg C to 40 deg C 32 deg F to 104 deg F
Mains Power	100-240V AC, 50-60Hz, 150W max
USB Audio playback	Mono/stereo .WAV files, 16/24bit, 44.1/48/96kHz MP3 files FLAC files
USB Audio recording	Stereo .WAV files, 24bit 96kHz

Dimensions and Weights

	Unboxed		Boxed
Width x Depth x Height	917 x 627 x 269 mm 36.1" x 24.7" x 10.6"	Width x Depth x Height	1100 x 850 x 425 mm 43.3" x 33.5" x 16.7"
Weight	26kg 57.4lbs	Weight	34kg 75lbs