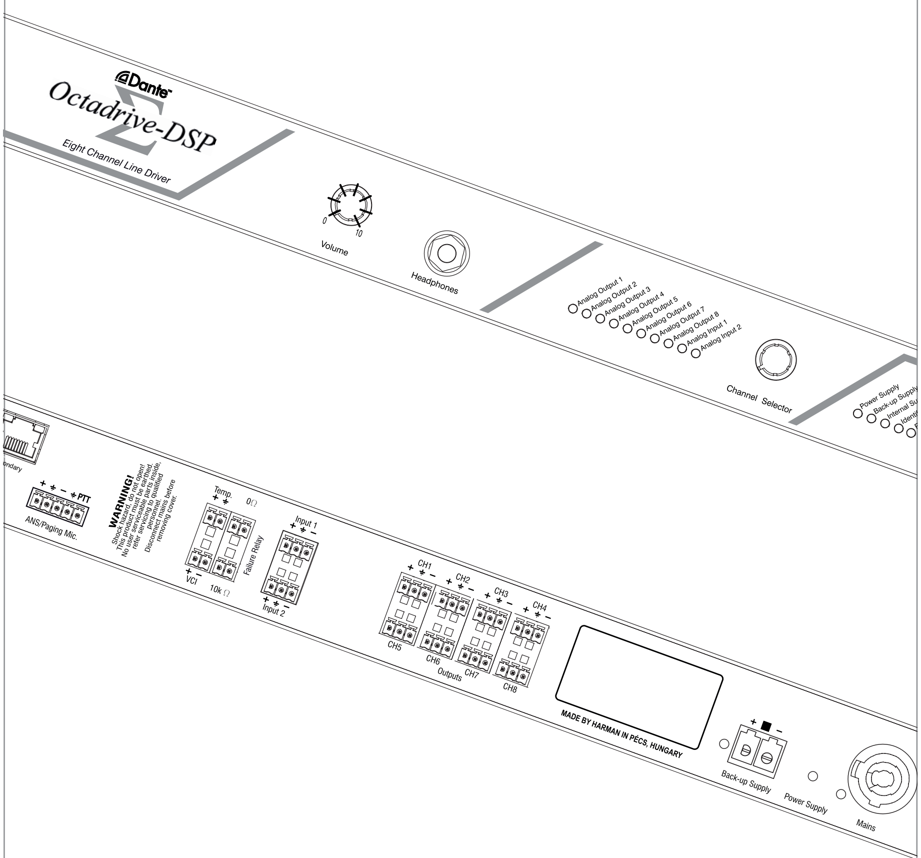
OCTADRIVE DSP-DN

Datasheet

Applies to Part Numbers:

TUN-391035

Octadrive DSP-DN (Dante™)



Delivering Clear and Intelligible Messages

AXYS® TUNNEL by HARMAN

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1. Architectural and Engineering Specifications

The unit shall be constructed in a 1U 19" rack enclosure. It shall be a multi channel line driver with on-board DSP and RISC processor that is intended for use in professional audio installations.

All signal processing functions, necessary to properly control and monitor each output channel shall be implemented on-board in order to reduce the overhead costs related to external processors. The electronics shall consist of a 2 channel analogue audio input module, 16 digital audio inputs and 18 digital audio outputs via Dante™, a Digital Signal Processor with 16 inputs and 8 transformer balanced analogue outputs.

The analogue audio line inputs shall be transformer balanced. All necessary signal processing shall be implemented in the digital domain by means of a 32 bit floating point DSP.

The DSP shall realize appropriate output channel gain, equalisation filters and delays. Besides the aforementioned, the DSP shall be able to realize EQ, volume and autogain, and compression as required. The DSP software and coefficients shall reside in non-volatile memory in order to facilitate adaptations and software updates. Audio AD and DA conversion shall be performed with high quality 24 bit converters.

The output section shall be equipped with eight independent low impedance drive channels. Each output shall be transformer balanced and be equipped with circuitry to compensate for the distortion created by the transformers. Hardware bypass functionality from analogue input 1 to all analogue outputs shall be provided. A headphone amplifier shall also be provided to allow local monitoring of all analogue inputs and outputs.

The device shall support a monitored microphone input for either Ambient Noise Sensing (ANS) or Paging applications. The monitoring scheme shall be capable to detect open circuit or short circuit failures for either electret or dynamic microphone types. For paging purposes, the microphone signal shall be made available to Dante™ and the microphone signal processing shall involve a low-cut filter, gain control and limiter. A monitored Push To Talk (PTT) switch input shall be provided by the device.

The device shall be controllable over Ethernet by using Layer 3 protocols. The DSP shall be controllable by means of the AxysBridge protocol. The Dante™ node shall be controllable via the Dante™ software tools. The interface shall be capable of redundant operation, both for the audio as well as for all control data.

The control unit shall serve four main functions:

- Remote monitoring of parameters like status of the DSP, monitoring and control of the analogue and digital input pilot tone detection, status of the optional microphone, ambient noise level, control for the input section etc.
- Remote control of DSP parameters: volume, pre-delay, EQ, output sections, autogain configuration and surveillance related parameters.
- Updating DSP software and factory unit programming.
- Configuration of the Dante™ node.

The device shall provide two voltage free failure relay contacts for monitoring purposes. The relay contacts shall NOT be individually switched, however one contact pair shall have internal resistors to allow for a direct connection to external impedance-sensing monitoring equipment.

The device shall be fitted with an opto-isolated control voltage input which shall trigger an "emergency preset".

The analogue audio signal line inputs and outputs shall be connected to Phoenix type MC 1,5/ 3-ST-3,81 connectors. The device shall be equipped with two RJ-45 sockets to enable either connection to primary and secondary 1 Gb/s Dante™ networks, or to provide daisy-chaining in non-redundant operation mode.

The device shall be equipped with an internal universal mains power supply. An external DC back-up supply input shall be provided as well. The device shall provide local as well as remote failure information regarding the mains-operated supply, back-up supply and internal supplies. The device shall be equipped with a Neutrik® PowerCon® mains inlet connector with on/off switch located on the rear panel. The back-up supply input shall be connected to a Phoenix type PC 4/ 2-ST-7,62 connector.

The enclosure shall be constructed of steel finished with a nickel plating. All connectors shall be grouped together on the rear of the chassis, with the exception of the headphone socket which shall be on the front of the unit. The front of the enclosure shall accommodate various status and failure LEDs, headphone volume control and a source selector control for the headphone output.

Dimensions are: 43.5 mm H x 482 mm W x 232 mm D. Weight 3.9 kg. The unit shall be the AXYS® model Octadrive DSP-DN.

2. Specifications

Electrical:

Analogue audio inputs[1]	- Number of inputs	: 2
	- Nominal level	: 0 dBV (RMS, line input)
	- Maximum level	: +21 dBV (peak, line input)
	- Type	: dual line input, transformer balanced
	- Impedance (balanced)	: 8k8 Ω (1k Hz)
	- Frequency range	: < 20 to > 22k Hz (-3 dB, analogue in to analogue out, 100k Ω load)
	- CMRR	: > 65 dB (1k Hz), > 90 dB (50 Hz)
Analogue audio outputs[2]	- Number of outputs	: 8
	- Nominal gain	: 2 dB (1k Hz, 'low' gain setting, from analogue input) 16 dB (1k Hz, 'high' gain setting, from analogue input)
	- Maximum level	: 5 dBV (RMS, 1k Hz, 'low' gain setting, default limiters) 19 dBV (RMS, 1k Hz, 'high' gain setting, default limiters)
	- Output impedance	: < 50 Ω (1k Hz)
	- Type	: transformer balanced, distortion compensated
ANS/paging mic input	- Type	: electronically balanced
	- Mic type	: electret or dynamic
	- Maximum level[3]	: -24 dBV (peak)
	- CMRR	: > 80 dB (1k Hz)
	- Dynamic range[4]	: > 80 dB
	- Monitoring[5]	: DC based open/short circuit detection
	- Low-cut filter[6]	: fixed HPF, F-3dB = 135 Hz
	- Gain[6]	: 30, 40, 50 or 60 dB (software configurable)
	- Limiter[6, 22]	: VCA operated, fixed parameters
	- Control[7]	: external Push To Talk (PTT) switch
General	- Dynamic range[8]	: > 90 dB
	- THD + N	: < 0.007 % (1k Hz, 3 dB below max. output level, 'low' gain setting, > 300 Ω load) < 0.03 % (50 to 10k Hz, 3 dB below max. output level, 'low' gain setting, > 300 Ω load)
Digital audio interface[9]	- Number of inputs	: 16
	- Number of outputs[10]	: 18
	- Format	: 48k Hz / 32, 24 or 16 bit PCM AES 67 supported
	- Type	: dual 1 Gb/s or 100 Mb/s Dante™ Ethernet connection
	- Implementation	: Brooklyn II Reference Design with internal Ethernet switch
	- Connection	: redundant or daisy-chain
	- Network latency	: 250 us, 500 us, 1 ms, 2 ms or 5 ms
DSP module	- Type	: floating point 32 bit
	- Memory	: 128 Mb SDRAM + 10 Mb non volatile
	- AD - DA conversion	: 24 bit sigma-delta 128 x oversampling
	- Auxilliary processor	: 200 nsec single cycle RISC
	- Sample rate	: 48k Hz (default)
	- Latency	: 3.43 ms (analogue in to analogue out) 3.68 ms (digital in to analogue line out)[11] 2.26 ms (analogue line in to digital out) 0.44 ms (microphone in to digital out)
	- Signal processing	: - input configuration (16 inputs to DSP) - individual input EQ, gain and polarity - volume - ambient noise level dependent gain adaptation ('fail-safe')

		- individual output EQ, delay, gain and polarity - output channel delay (43.6 seconds per output) - individual RMS and peak limiters on each output
	- Dante™ transmission routing	: - Ch 1 Break-out Analog In 1[12] - Ch 2 Break-out Analog In 2[12] - Ch 3 Break-out In 1 (processed) - Ch 4 Break-out In 2 (processed) - Ch 5 Break-out In 3 (processed) - Ch 6 Break-out In 4 (processed) - Ch 7 Break-out In 5 (processed) - Ch 8 Break-out In 6 (processed) - Ch 9 Analogue Output 1 - Ch 10 Analogue Output 2 - Ch 11 Analogue Output 3 - Ch 12 Analogue Output 4 - Ch 13 Analogue Output 5 - Ch 14 Analogue Output 6 - Ch 15 Analogue Output 7 - Ch 16 Analogue Output 8 - Ch 17 Microphone - Ch 18 Microphone (processed)
Control & monitoring		
	- Interface[13]	: AxysBridge protocol
	- Remote surveillance	: - general status (DSP running, signal monitoring etc.) - pilot tone detection on analogue inputs (20k - 28k Hz, level > -22 dBV) - pilot tone detection on Dante™ inputs (19k2 to 23k5 Hz, software configurable parameters) - monitoring of optional external microphone - ambient temperature monitoring and frost protection - control voltage input status - PTT switch monitoring - mains supply and back-up supply monitoring
	- Failure	: - internal hardware bypass circuit for analogue audio input 1 to all analogue outputs - failure relay contacts 1 (external connector, maskable conditions) SPST 100 mA / 24 V - failure relay contacts 2 with 10k / 20k Ω internal resistors (external connector, maskable conditions) SPST 100 mA / 24 V - failure status indicated at front by bi-colour LED
	- Analogue output pilot tone frequency[14]	: 18k to 30k Hz
	- Analogue output pilot tone level[15]	: -45 to -10 dBFs
	- Dante™ output pilot tone frequency[16]	: 19k5 to 23k25 Hz
	- Dante™ output pilot tone level[17]	: -60 to -20 dBFs
	- Control voltage input	: - optically isolated control input - state low for Vin <= 3.3 VDC - state high for Vin >= 3.4 VDC - max. input level 48 VDC
Headphone output	- Monitoring	: all analogue inputs and outputs
	- Channel select	: rotary control and LED indication
	- Control	: volume
	- Frequency range	: 20 to > 20k Hz (-3 dB)
	- Maximum output level	: 17.5 dBV (RMS, 1k Hz, 300 Ω load)

Connectors[18]	- Analogue audio inputs	: Phoenix type MC 1,5/ 3-ST-3,81 (2 x)[19] p1 = Line +, p2 = GND, p3 = Line -
	- Analogue audio outputs	: Phoenix type MC 1,5/ 3-ST-3,81 (8 x) p1 = Line +, p2 = GND, p3 = Line -
	- Dante™ interface	: RJ-45 (2 x)[9]
	- ANS/Paging mic	: Phoenix type MC 1,5/ 5-ST-3,81 Mic : p1 = In +, p2 = GND, p3 = In - PTT switch : p4 = GND, p5 = PTT
	- Ambient temperature sensor (NTC)	: Phoenix type MC 1,5/ 2-ST-3,81 p1 = In, p2 = GND
	- Control Voltage Input	: Phoenix type MC 1,5/ 2-ST-3,81 p1 = +, p2 = -
	- Failure relays[20]	: Phoenix type MC 1,5/ 2-ST-3,81 (2 x) contacts 1 : short / open circuit contacts 2 : 10k / 20k Ω
	- Headphone socket	: 1/4” (6.35 mm) stereo (TRS) jack tip = left channel, ring = right channel, sleeve = ground
	- Back-up supply	: Phoenix type PC 4/ 2-ST-7,62 [21] p1 = +, p2 = -
	- Mains	: Neutrik® PowerCon® NAC3FCA
Indicators	- Power supply LED	: green (OK) / off (failure), front and rear
	- Back-up supply LED	: green (OK) / off (failure), front and rear
	- Internal supplies LED	: green (OK) / off (failure)
	- Identify LED	: green, front and rear
	- Failure relay LED	: green (OK) / red (failure)
	- Serial bridge activity LED	: green (xmt or rcv) / off (no communication activity)
	- Microphone level LED[22]	: green (level OK) / red (limiter active)
	- Monitoring indication LEDs	: green (channel selected)
Mains supply	- Type	: Switched-mode
	- Mains voltage	: 100 V to 240 V, 50 or 60 Hz
	- Mains fuse(s)	: internal, not user serviceable
	- Power consumption	: 15 W typical, 28 VA
	- Protection	: - short circuit - overload - overvoltage
Back-up supply	- Type[23]	: Switched mode
	- Input voltage[24]	: 12 to 48 VDC
	- Fuse	: PPTC resettable fuse
	- Power consumption	: 13 W

General:

Temperature range (ambient)		: 0 to 40 °C (32 - 104 °F)
Dimensions (H x W x D)		: 43.5 x 482 x 232 mm (1U 19” rack enclosure)
Weight		: 3.9 kg (8.6 lbs)
Finish		: Nickel Plated
MTBF[25]		: 150000 hours
Standards	- EMC	: EN 55032:201/AC:2013 EN 55103-2:2009 EN 50130-4:2011
	- Safety	: IEC 60065: 2014 (Edition 8) and European Group Differences according to EN 60065:2002+A1:2006 +A11:2008+A2:2010+A12:2011
	- Mains harmonics	: EN 61000-3-2:2014
Certificates		: CE, CB, CSA/UL

Notes:

1. The device supports 2 analogue audio inputs and 16 digital Dante™ inputs. Either analogue input 1/2 or Dante™ input 1/2 can be routed to the DSP (software configurable).
2. The analogue output gain for all outputs is software configurable in 2 steps (with 14 dB gain difference). All measurements with 100k Ω load unless stated otherwise.
3. This input level results in a full-scale output on the 'unprocessed' microphone Dante™ transmit channel.
4. A-weighed, 10 to 22k Hz analyzer bandwidth, microphone input shorted, microphone gain setting is set to the lowest value (30 dB).
5. Software configurable open circuit and short circuit detection thresholds. The DC impedance of dynamic microphone types should be between 300 and 10k Ω to ensure proper capsule monitoring operation.
6. Only applied to the 'processed' microphone Dante™ transmit channel. Signal processing does not affect the local Autogain operation.
7. Connect PTT to GND to activate the 'processed' microphone Dante™ transmit channel. For monitored PTT operation, the DC impedance between PTT and GND should be 10k Ω (activated) respectively 20k Ω (deactivated). PTT control by network command is supported as well, this operates when PTT is unconnected.
8. Measured on analogue output with one analogue input active (open), flat filter settings and all gains 0 dB. A-weighed, 10 to 22k Hz analyzer bandwidth, 100k Ω load.
9. The device is equipped with 2 RJ-45 sockets for either redundant or daisy-chain connection to 1 Gb/s or 100 Mb/s Dante™ network(s). A 1 Gb/s connection is strongly recommended.
10. Various signals are available on the 18 Dante™ transmit channels, see section 'DSP block diagram' for details.
11. Including typical Dante™ latency setting of 1 ms. Available Dante™ latency settings are 250 us, 500 us, 1 ms, 2 ms and 5 ms (software configurable). Ensure that lower latency settings are supported by the network infrastructure.
12. Analogue input 1/2 are always available as Dante™ break-out, even when not routed to the DSP.
13. The DSP can be accessed over the Dante™ Ethernet interface(s) by using the UDP/IP based AxysBridge protocol. Service discovery (DNS-SD), redundant operation mode and redundant status monitoring are supported.
14. Software configurable in steps of 250 Hz.
15. Software configurable in steps of 1 dB. The actual output level is depending on the analogue output gain ('low' or 'high' setting). 0 dBFs refers to the maximum analogue output level. Pilot tone can be globally enabled/disabled for all local analogue outputs.
16. Software configurable in steps of 750 Hz.
17. Software configurable in steps of 0.1 dB. 0 dBFs refers to the maximum digital output level. Pilot tone can be individually enabled/disabled for each Dante™ transmit channel 1 to 16. Pilot tone summing is not supported for the microphone related Dante™ transmit channels.
18. All Phoenix type numbers refer to the required cable parts, a complete set of Phoenix connectors is supplied with the product.
19. For solid and stranded wires with conductor cross sections from 0.14 to 1.5 mm^2.
20. Contact 1 pins are shorted in case the device is powered and the status is OK (no masked failure). The impedance between contact 2 pins is 10k Ω in that case. In case of a masked failure, contact 1 pins are open circuit and the impedance between contact 2 pins is 20k Ω.
21. For solid and stranded wires with conductor cross sections from 0.2 to 4.0 mm^2.
22. Green LED is on for level >= -23 dBFs, red LED is on for level >= -11 dBFs. Level refers to the peak level before limiter when tested with 1 kHz sine. The limiter threshold is fixed at -13 dBFs.
23. Floating design, - input is not connected directly to chassis ground.
24. Absolute voltage on - as well as + input must be kept within 65 V from chassis ground voltage.
25. At ambient temperature of 20 °C.

3. Octadrive DSP-DN measurement plots

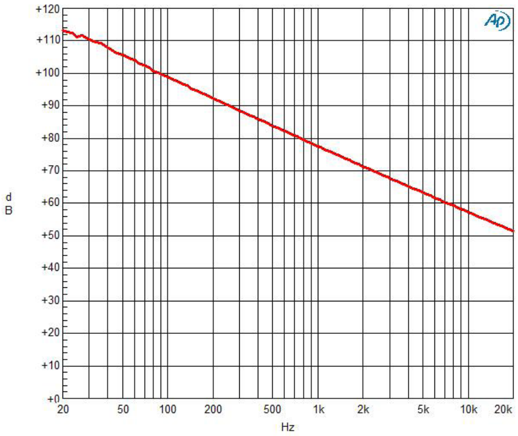


Fig 1 CMRR vs frequency, analogue input.

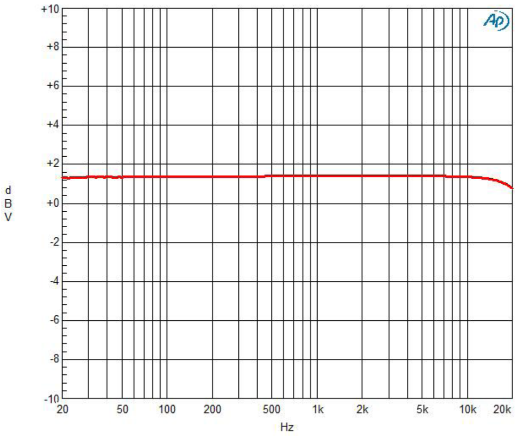


Fig 2 Magnitude vs frequency[1].

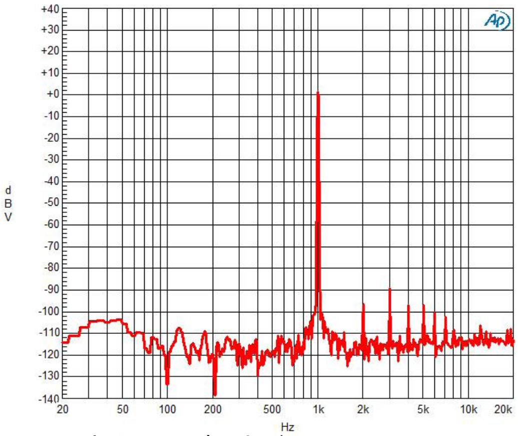


Fig 3 FFT, 16k points[1].

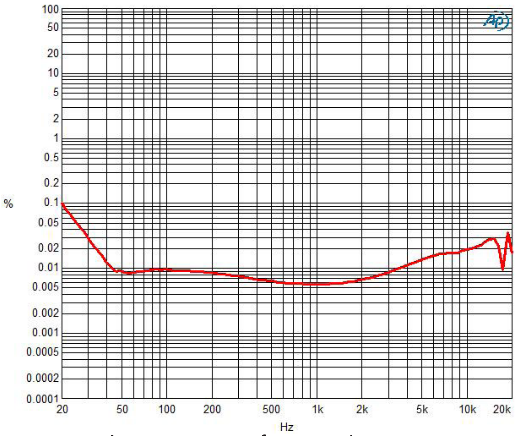


Fig 4 THD+N vs frequency[1].

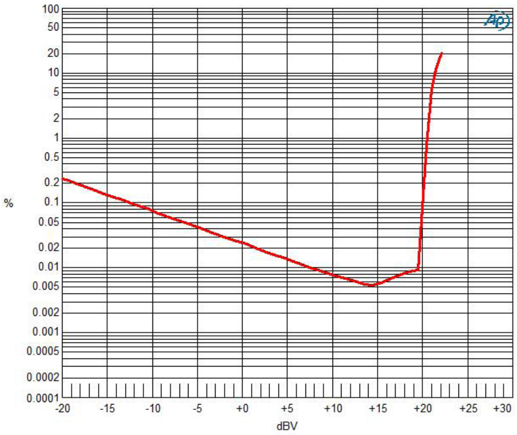


Fig 5 THD+N vs output level, 1k Hz, output limiters not active[2].

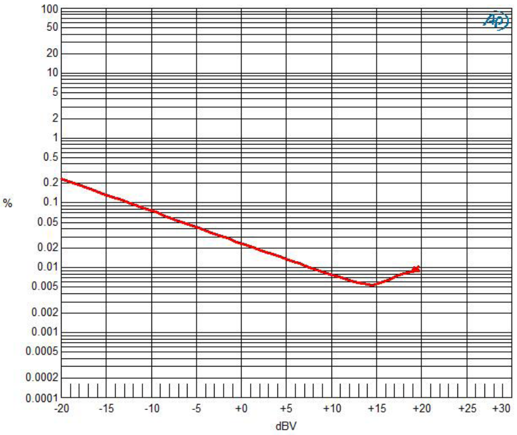


Fig 6 THD+N vs output level, 1k Hz, output limiters with default params[2].

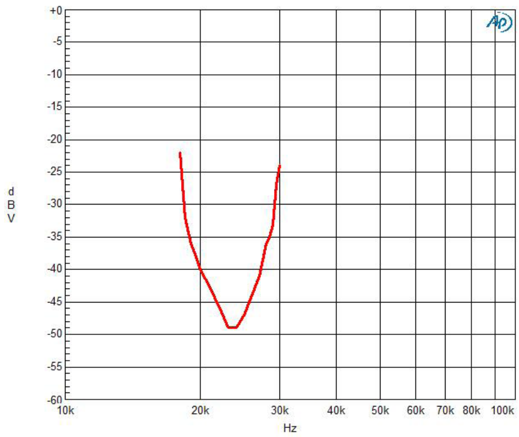


Fig 7 Analogue input pilot tone lock-in threshold vs frequency[3].

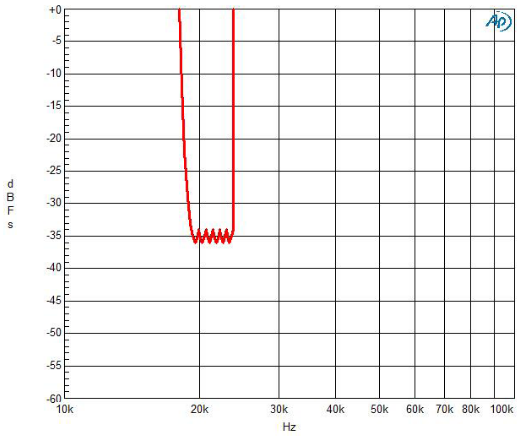
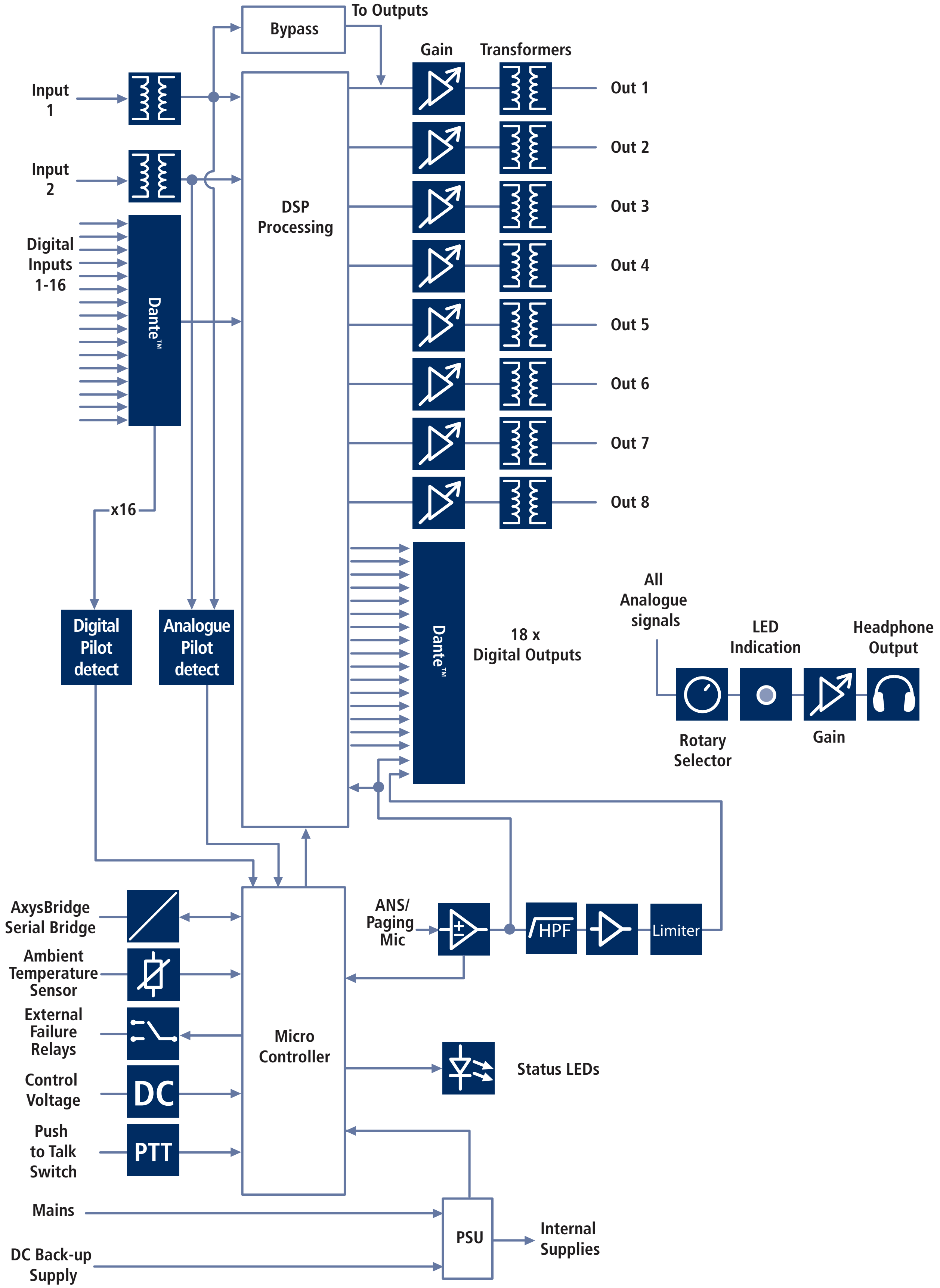


Fig 8 Dante™ input pilot tone lock-in threshold vs frequency[4].

Notes:

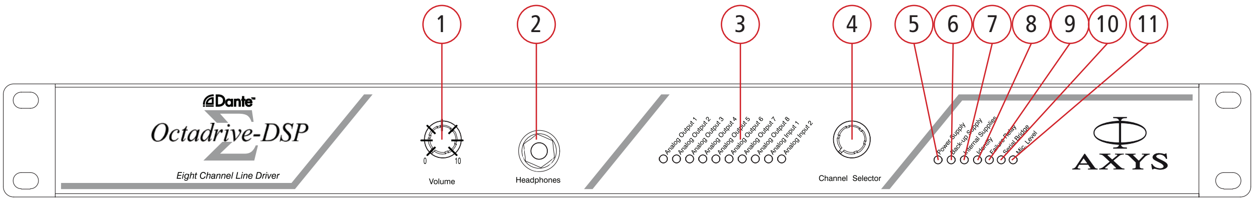
1. Analogue input to analogue output, 600 Ω load, 0 dBV in, 'low' analogue output gain setting.
2. Analogue input to analogue output, 600 Ω load, 'high' analogue output gain setting.
3. Lock-in occurs when RMS level of pilot tone >= threshold.
4. Detection threshold set to -36 dBFs, 'all frequencies', lock-in occurs when peak level of pilot tone >= threshold.

4. Functional Diagram

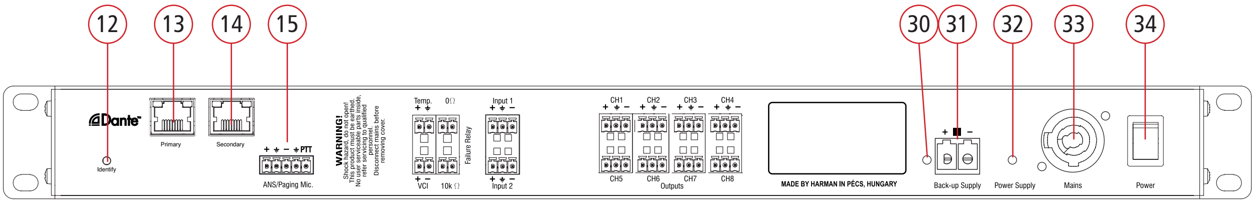


5. Mechanical Details

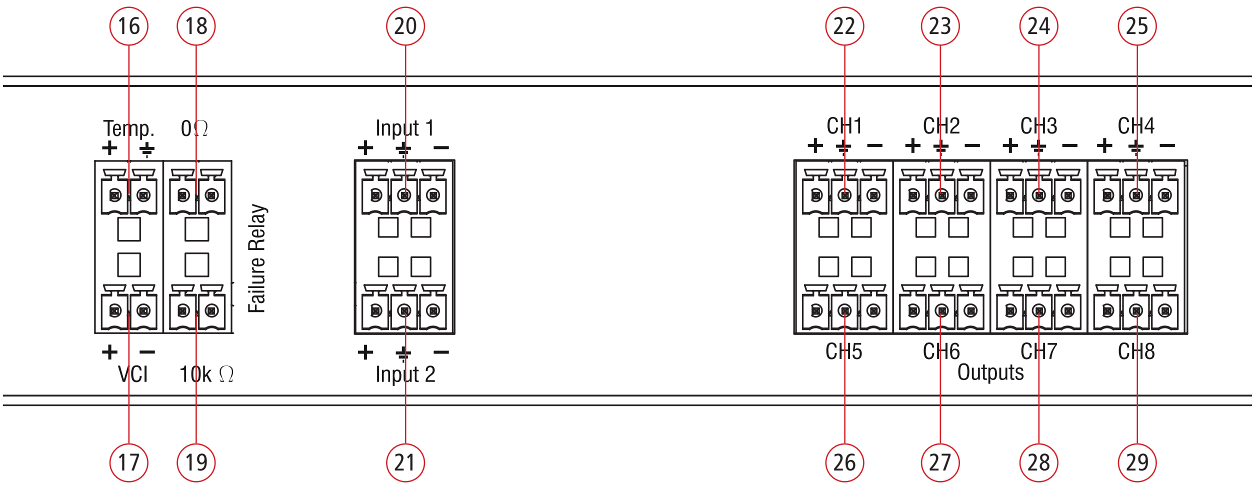
Front Panel



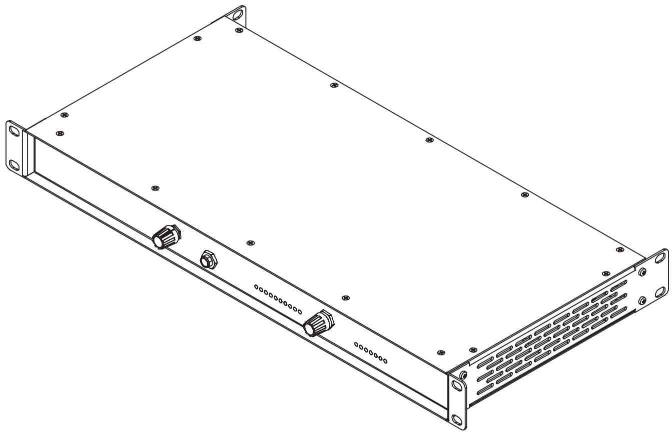
Rear Panel

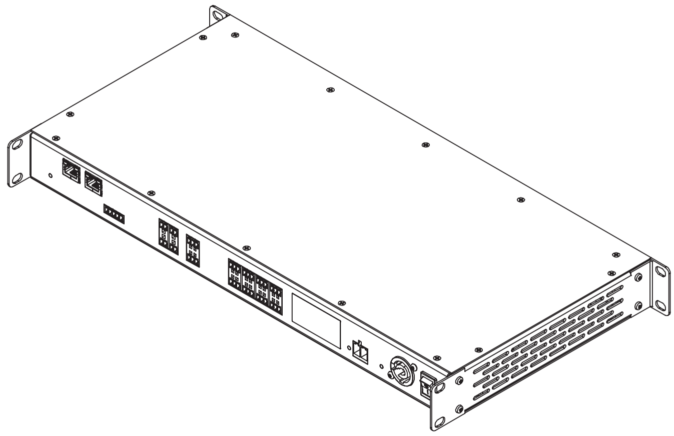


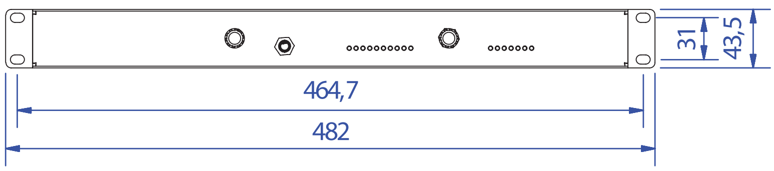
Detailed view of Input / Output panel

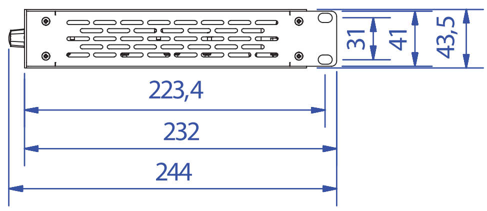


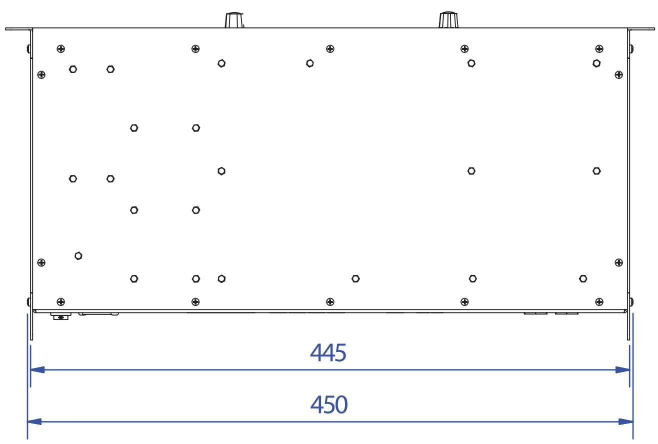
1. Headphone volume control
2. Headphone socket
3. Source selection LEDs
4. Headphone source selector encoder wheel
5. Power Supply LED
6. Back-up Supply LED
7. Internal Supplies LED
8. Identify LED
9. Failure relay LED
10. Serial Bridge activity LED
11. Mic Level LED
12. Identify LED (rear)
13. Dante™ Primary
14. Dante™ Secondary
15. ANS/Paging Mic
16. Ambient temperature sensor
17. Control voltage input
18. Failure relay contacts 1
19. Failure relay contacts 2
20. Input 1
21. Input 2
22. Output 1
23. Output 2
24. Output 3
25. Output 4
26. Output 5
27. Output 6
28. Output 7
29. Output 8
30. Back-up Supply LED (rear)
31. Back-up Supply connector
32. Power Supply LED (rear)
33. Mains power connector
34. Power switch





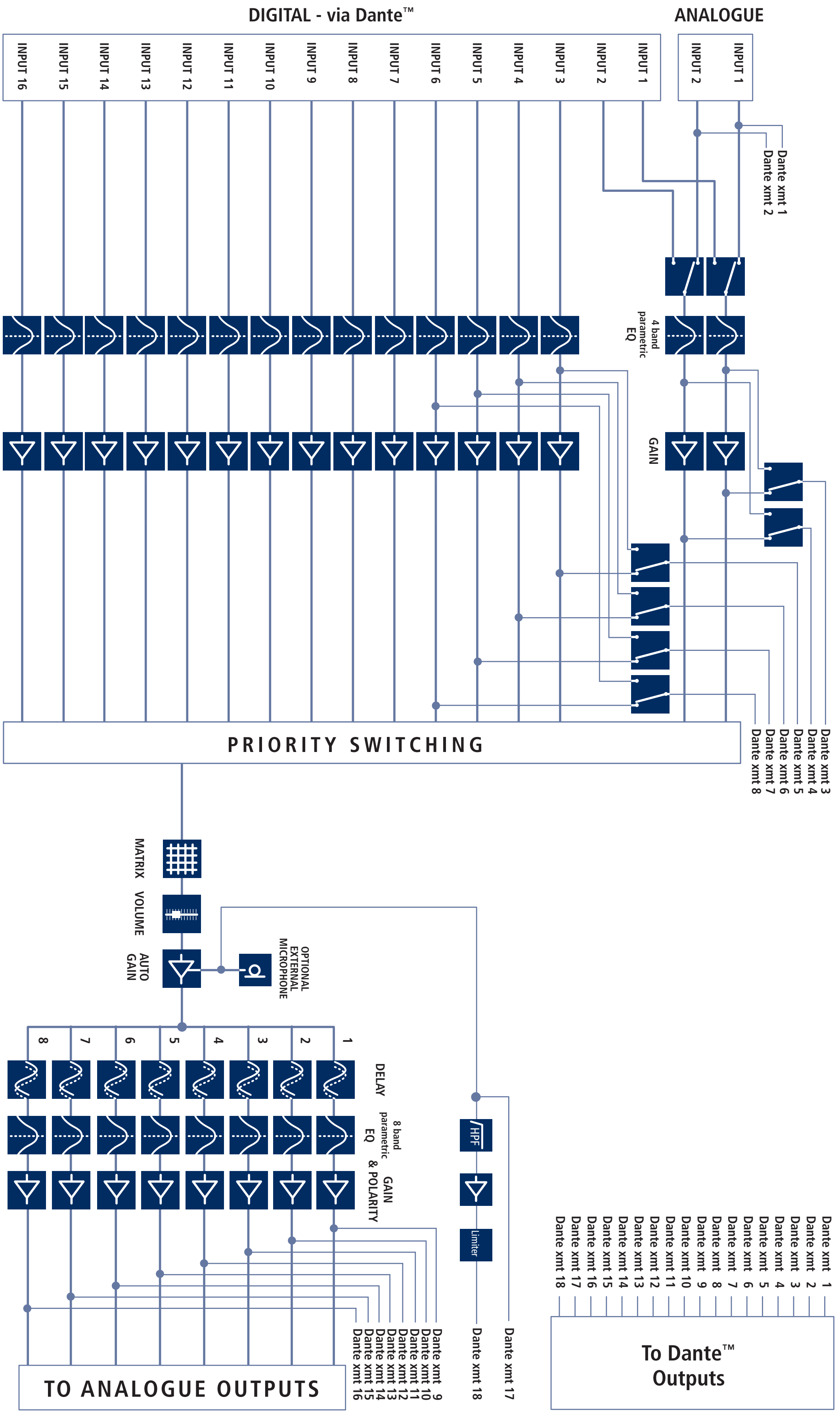






Note: All Dimensions in mm

6. DSP Block Diagram



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