DSP-128

Avia™ DSP Matrix Mixer

- *Engineered to deliver exceptional pro audio performance with faster, easier implementation*
- *Ready to go out of the box and extensively configurable*
- *Hybrid channel strip architecture*
- *Customizable input and output channel strips*
- *Eight internal aux buses with channel strips*
- *Full set of selectable channel strip objects*
- *Fully-adjustable matrix mixing*
- *Clean and intuitive software tool workspace*
- *Real-time configuration and adjustment without compiling*
- *Work offline or live via Ethernet or USB*
- *Native Crestron system integration for rapid programming*[1]
- *Seamless export of Smart Graphics™ touch screen UI files*
- *Seamless integration with Crestron RL® 2 and DM®*
- *Easy integration with third-party codecs and court recording systems*
- *SIP or POTS audio conferencing interface*[2]
- *High-performance acoustic echo cancellation (AEC)*[2]
- *Fully automatic microphone mixing*
- *Logic output for control of cameras and switchers*[1]
- *Built-in real-time spectrum analyzer and noise generator*
- *Highest-quality converters, preamps, and line amps*
- *Up to twelve mic/line inputs with 66 dB gain range*
- *Up to eight +24 dB balanced line outputs*
- *32x32 Dante™ audio network interface*[2]
- *8x8 USB Audio interface*[2]
- *Front panel meters on select models*
- *Up to 32 real-time VU meters on a touch screen*[1]
- *Enterprise-grade Ethernet networking*
- *Crestron Fusion® Cloud remote monitoring*[1]
- *Internal universal power supply*
- *Single-space 19" rack mountable*
- *Five models available*
- *Companion 8-channel power amplifiers available*

The Crestron® Avia™ DSP-128 family of digital signal processors leverages the highest quality components and the expertise of veteran audio industry engineers to deliver a revolutionary audio processing platform that's actually easy to integrate and configure. It has all the features and performance top sound system designers demand — complemented with an intuitive graphical workspace conceived to inspire exceptional results quickly.

Delivering quality sound in any meeting space, performance venue, courtroom, sports facility, or house of worship demands high-performance, professional audio signal processing. A good digital signal processor (DSP) must deliver sophisticated processing, mixing, and routing for all types of audio signals with a comprehensive set of controls and adjustments to manage the behavior and sound quality of each signal. Until now, DSP products have been either limited in their functionality and flexibility or too complicated and time consuming to implement economically. With Avia, Crestron has addressed all of those concerns.

Hybrid Channel Strip Architecture

Most common audio DSP products come in two basic forms — fixed layout or fully-configurable. One offers out-of-the-box readiness with limited flexibility while the other is best suited for use by advanced-level audio engineers who desire a fully custom signal path. Crestron marries the best of both by employing a flexible architecture using customizable channel strips and matrix mixing.

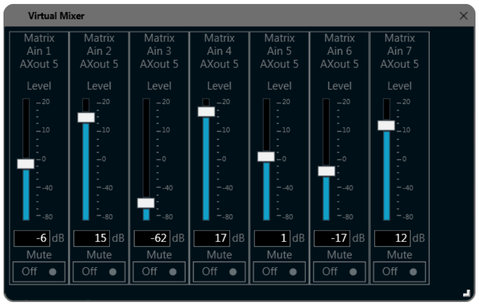
The channel strips in the DSP-128 are composed of a selection of essential signal processing objects that are ready to use and logically arranged for optimal functionality. Fully-adjustable matrix mixing enables complete customization of the signal flow, allowing inputs to be mixed in any combination and routed to any of the outputs. Eight internal auxiliary busses are included to allow any of the inputs to be mixed and processed as a subgroup. Each input, output, and aux bus includes its own independently configurable channel strip, and any two adjacent channels can be grouped together for stereo.

DSP-128 Tool Software

Rapid system configuration and adjustment is enabled using the DSP-128 Tool software. Its clean, modern user interface provides a workspace that's easy and intuitive to navigate. The DSP-128 Tool allows setup and operation to be performed live over a USB or LAN connection, or "virtual DSPs" can be configured offline and uploaded locally or remotely.

The DSP-128 Tool software workspace with several DSP objects open including a parametric equalizer with spectrum analyzer

The DSP-128 Tool accelerates system setup by allowing custom channel strips to be quickly configured and saved into libraries, and then simply dragged and dropped to any input, output, or aux bus. Settings within any channel strip object or matrix bus can be copied and pasted easily between channels. Pre-configured channel strips are even included to jump-start acoustic setup, with "speaker profile" settings optimized for Crestron Saros® speakers. Custom views and "virtual mixers" can be arranged and saved at any time to easily move between different setup scenarios while commissioning the system. Selected settings and objects can be saved and recalled as presets to address room combining, reconfigurable seating, zone paging, and acoustical variations in ballrooms, convention halls, stadiums, and other configurable spaces.



A "virtual mixer" in the DSP-128 Tool software

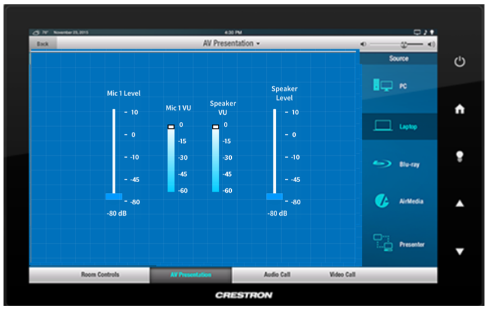
Built-in Real-Time Analyzer

There's no need to lug an expensive RTA or signal generator on site when setting up an Avia system. They're built in! Every EQ and Filter object includes a professional spectrum analyzer display that overlays the EQ/filter graph, providing precise real-time visualization to facilitate optimal adjustment of equalization and crossover settings. Your choice of test microphone(s) can be connected to any spare input(s) on the DSP-128 and routed to the analyzer via either of two reference buses. A pink noise, white noise, or sine wave signal can be generated and routed to any of the outputs via a dedicated matrix bus.

Crestron Control® Integration [1]

A complete audio DSP solution often includes integration with a control system to facilitate everyday operation and automation of the DSP and other equipment. A successful installation should be easy to operate using touch screens and keypads tailored to the customer's needs and preferences. But, the design and programming required to make that happen can be extensive and costly.

Avia offers native Crestron control system integration to substantially reduce the amount of programming required. It even eliminates much of the touch screen UI design work by allowing components within the DSP-128 Tool to be selected and exported as a Smart Graphics™ file containing ready-to-use SmartObject® touch screen controls and meters. In fact, along with a variety of buttons and sliders, Avia supports up to 32 real-time VU meters running simultaneously on a touch screen. With minimal programming, a system integrator can provide their customer with a touch screen control solution custom tailored to their needs, with anything from a few selectable presets and volume controls to a complete virtual mixing console.



A touch screen user interface featuring volume control sliders and real-time VU meters

Additional functions enabled through the control system include the ability to assign automixer logic signals to control a motorized camera or camera switcher, enabling the video image to follow the current active microphone automatically in a council chamber or courtroom. Additional integration with Crestron Fusion® Cloud allows DSP activity and status to be centrally controlled, monitored, logged, and reported as part of a complete managed enterprise. Whatever the customer requires, a complete integrated Crestron system makes it possible.

Dante™ Audio Networking

System expansion and integration with other audio devices is facilitated using Dante audio networking. Available on the DSP-1281 and DSP-1283 models, Dante networking provides an additional 32 mono inputs and 32 mono outputs with full matrix mixing. Multiple DSP-128 units can be linked via Dante to expand the number of inputs and outputs in the system. Dante also provides a multichannel digital audio bridge to a Crestron DigitalMedia™ system via Dante-enabled DM® switcher I/O blades. Additionally, Dante networking enables plug and play integration with other Dante-enabled mixers, switchers, sources, amplifiers, and computers. To simplify configuration of the Dante network, the Dante Controller application is built into the DSP-128 Tool.[2]

USB Audio Interface

The DSP-1282 and DSP-1283 models each include a USB Audio port, which can be connected to a computer, codec, or other USB Audio host. This USB 2.0 interface allows integration with a Crestron RL® 2 Group Collaboration System to enable the routing of stereo input and output signals between the DSP-128 and the Crestron RL 2 codec. It can also be used to provide up to eight independent output signals to feed a computer based court recording system.[2]

Acoustic Echo Cancellation

High-performance adaptive AEC on each analog input of the DSP-1282 and DSP-1283 models enables high-quality audio conferencing capability for systems with multiple table or ceiling microphones. Low-latency, full-bandwidth performance affords highly-effective echo cancellation with natural sound quality. AEC is employed independently per microphone, with a fast convergence rate of 100 dB per second and tail lengths up to 300 ms.

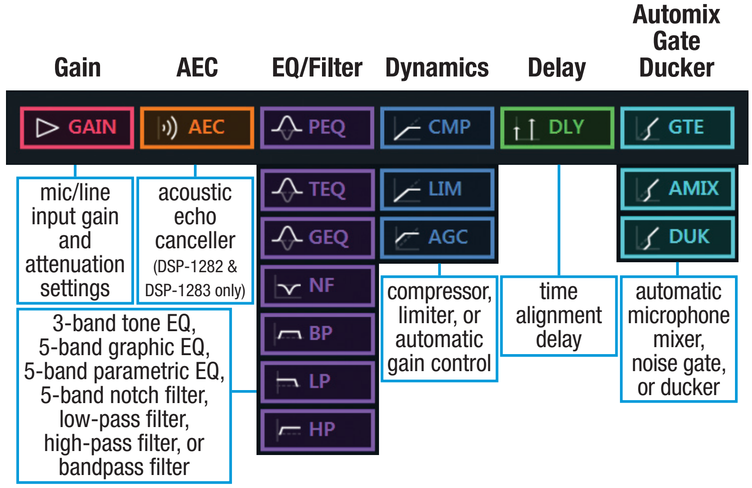
SIP & POTS Phone Interface

The DSP-1282 and DSP-1283 models each include a built-in audio conferencing interface and phone dialer to enable full-duplex voice conferencing capability over a SIP or POTS phone line. Crestron Rava® technology allows Avia to interface over Ethernet with a SIP-compatible phone system or other SIP compatible device. An RJ11 POTS interface is also built in to allow for direct connection to a conventional analog phone line or extension port. Using either SIP or POTS, participants can leverage a control system touch screen to place outgoing calls, receive incoming calls, and control the entire session.[1,2]

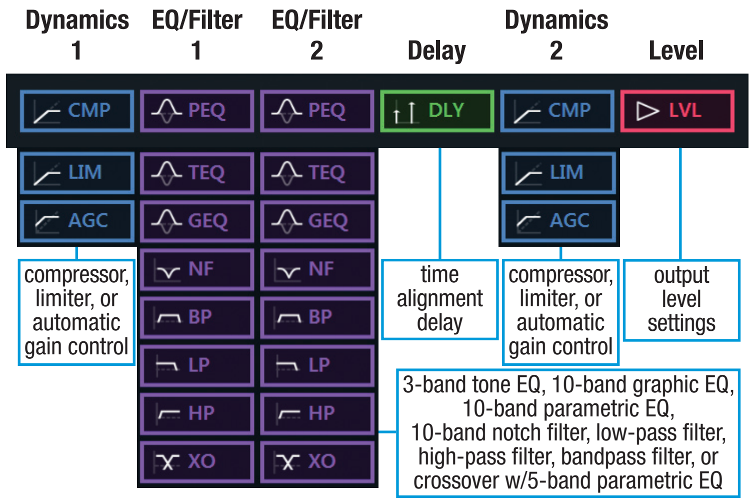
Channel Strip Objects

A complete arsenal of signal processing objects is included to customize each channel strip on every analog input, output, and aux bus:

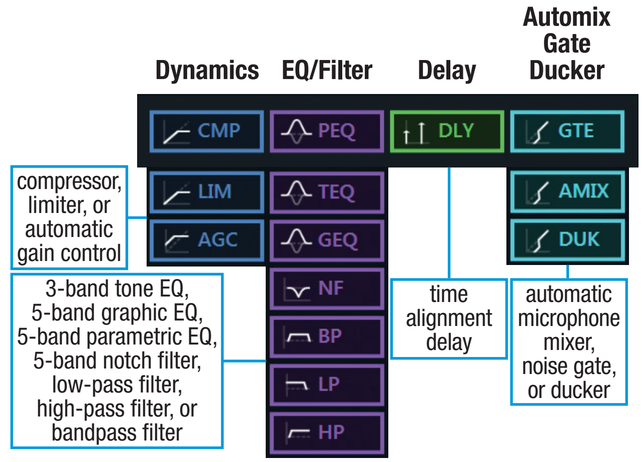
Analog Inputs



Analog Outputs



Aux Buses

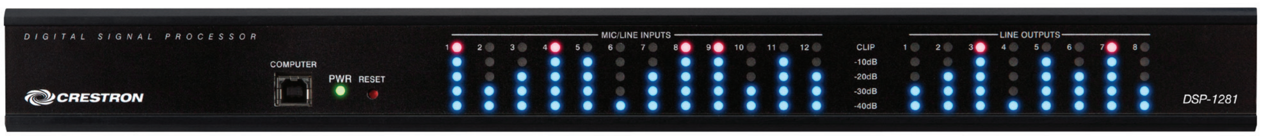


DSP-128 Avia™ DSP Matrix Mixer

The Avia DSP-128 family comprises five models designed to address a full range of applications:

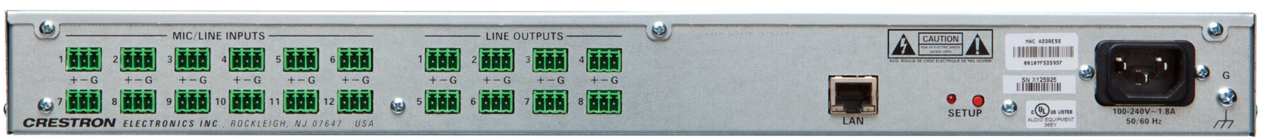
Model	Analog Mic/Line Inputs	+24 dB Analog Outputs	Internal Auxiliary Buses	Acoustic Echo Cancellation	SIP/POTS Phone Interface	8x8 USB Audio I/O	32x32 Dante™ I/O	Front Panel Meters
DSP-860	8	6	8					
DSP-1280	12	8	8					
DSP-1281	12	8	8				•	•
DSP-1282	12	8	8	•	•	•		
DSP-1283	12	8	8	•	•	•	•	•

DSP-1280 – Front View (Typical of DSP-860 & 1282)

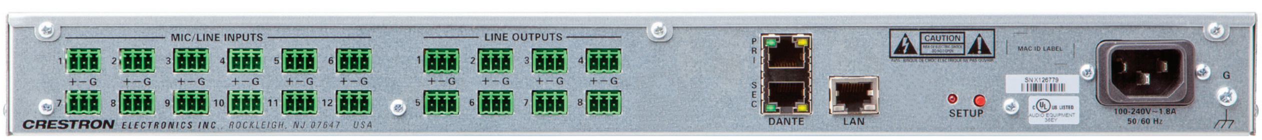


DSP-1281 – Front View (Typical of DSP-1283)

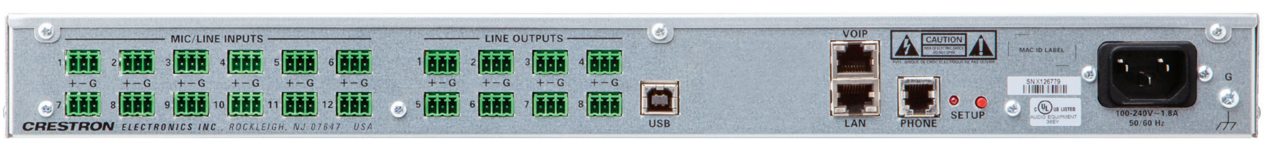
DSP-860 – Rear View



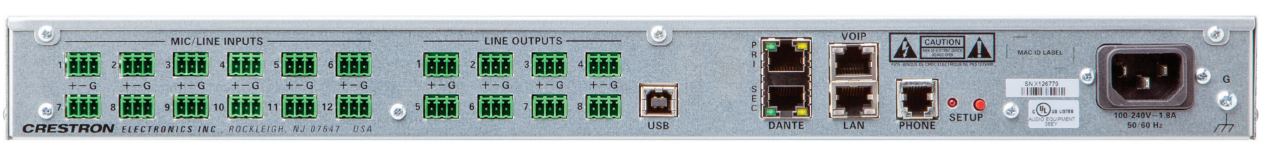
DSP-1280 – Rear View



DSP-1281 – Rear View



DSP-1282 – Rear View



DSP-1283 – Rear View

SPECIFICATIONS

Audio – General

Analog-To-Digital Conversion: 24-bit 48 kHz
Digital-To-Analog Conversion: 24-bit 48 kHz
Frequency Response: 20 Hz to 20 kHz ±0.5 dB
THD: 0.001%, 20 Hz to 20 kHz, 0 dB gain, +4 dBu input;
0.01%, 22 Hz to 22 kHz, 54 dB gain, -50 dBu input
EIN: -125 dBu, 22 Hz to 22 kHz, no weighting
Dynamic Range: 110 dB, 22 Hz to 22 kHz, 0 dB gain
Crosstalk: -85 dB, 1 kHz, +4 dBu input, channel to channel
-75 dB, 1 kHz, -50 dBu input, channel to channel
Latency: 3.0 ms (analog in to analog out)
Dante™ I/O (DSP-1281 & DSP-1283 only): 32 channels in, 32 channels out, at up to 24-bit 48 kHz
USB Audio I/O (DSP-1282 & DSP-1283 only): 2 or 8 channels in, 2 or 8 channels out, 16 or 24 bit, 48 kHz
AEC Performance (DSP-1282 & DSP-1283 only):
- **Bandwidth:** 20 Hz to 20 kHz
- **THD+N:** 0.001%, +4 dBu input
- **Convergence Rate:** 100 dB/s
- **Tail Length:** 300 ms

Audio – Matrix Mixer

Matrix Crosspoint Settings:
- **Gain:** -80 to +20 dB, Gain adjustment per crosspoint
- **Mute:** Mute enable/disable per crosspoint

Audio – Input Channels

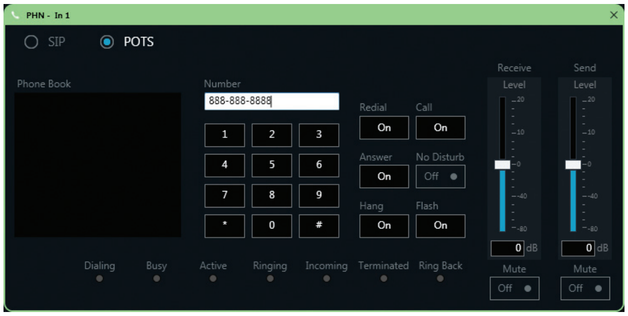
Analog Input Channels 1 – 8 (DSP-860) or 1 – 12 (DSP-128x):
- **Signal Types:** Mono analog mic or line level, adjacent channels groupable for stereo
- **Signal Flow:** Mic Preamp / Line Input (Gain) > ADC > VU Meter > Channel Strip > Channel Trim > Matrix Mixer
- **Channel Strip**:
  - **Object 1:** Gain
  - **Object 2:** Acoustic Echo Canceller (DSP-1282 & DSP-1283 only)
  - **Object 3:** EQ/Filter – Choice of 3-Band TEQ, 5-Band GEQ, 5-Band PEQ, 5-Band Notch Filter, LP Filter, HP Filter, or BP Filter
  - **Object 4:** Dynamics – Choice of Compressor, Limiter, or AGC
  - **Object 5:** Delay
  - **Object 6:** Automixer/Gate/Ducker – Choice of Automixer, Gate, or Ducker
- **Channel Trim Settings:**
  - **Mute:** Matrix input bus mute enable/disable per channel
  - **Gain:** -80 to +20 dB, matrix input bus gain adjustment per channel
  - **+48V:** Phantom power enable/disable per channel

- **VU Meter:** -80 to +20 dB, analog input level meter per channel (visible in System View only)

Auxiliary Input Channels 1 – 8 (Internal):
- **Signal Source:** Fed internally from each corresponding Auxiliary Output Channel, adjacent channels groupable for stereo
- **Signal Flow:** Aux Output > Channel Strip > Channel Trim > VU Meter > Matrix Mixer
- **Channel Strip**:
  - **Object 1:** Dynamics – Choice of Compressor, Limiter, or AGC
  - **Object 2:** EQ/Filter – Choice of 3-Band TEQ, 5-Band GEQ, 5-Band PEQ, 5-Band Notch Filter, LP Filter, HP Filter, or BP Filter
  - **Object 3:** Delay
  - **Object 4:** Automixer/Gate/Ducker – Choice of Automixer, Gate, or Ducker
- **Channel Trim Settings:**
  - **Mute:** Matrix input bus mute enable/disable per channel
  - **Gain:** -80 to +20 dB, matrix input bus gain adjustment per channel
- **VU Meter:** -80 to +20 dB, auxiliary input level meter per channel (visible in System View only)

Signal Generator (Internal):
- **Signal Flow:** Signal Generator > Matrix Mixer
- **Signal Generator Settings**:
  - **Mute:** On/Off, mute enable/disable
  - **Sine Freq:** 20 to 20000 Hz, sine wave frequency adjustment
  - **Source:** White noise, Pink noise, or Tone (sine wave); signal type selection
  - **Level:** -60 to 0 dB, level adjustment

Phone Input Channel (DSP-1282 & DSP-1283 only):
- **Signal Types:** SIP (Peer to Peer or SIP Server) or POTS
- **Signal Flow:** Codec > VU Meter > Channel Trim > Matrix Mixer
- **Features:** Dialer, Phone Book, Do Not Disturb, Caller ID



- **Send Level:** -80 to +20 dB, send level adjustment
- **Send Mute:** On/Off, send mute enable/disable
- **Receive Level:** -80 to +20 dB, receive level adjustment
- **Receive Mute:** On/Off, receive mute enable/disable
- **Channel Trim Settings:**
  - **Mute:** Matrix input bus mute enable/disable
  - **Gain:** -80 to +20 dB, matrix input bus gain adjustment
- **VU Meter:** -80 to +20 dB, phone input level meter (visible in System View only)

Dante Input Channels 1 – 32 (DSP-1281 & DSP-1283 only):
- **Signal Flow:** LAN > Channel Trim > VU Meter > Matrix Mixer
- **Channel Trim Settings:**
  - **Mute:** Matrix input bus mute enable/disable per channel
  - **Gain:** -80 to +20 dB, matrix input bus gain adjustment per channel
- **VU Meter:** -80 to +20 dB, Dante input level meter per channel (visible in System View only)

USB Input Channels 1 – 8 (DSP-1282 & DSP-1283 only):
- **Signal Flow:** USB > Channel Trim > VU Meter > Matrix Mixer
- **Channel Trim Settings:**
  - **Mute:** Matrix input bus mute enable/disable per channel
  - **Gain:** -80 to +20 dB, matrix input bus gain adjustment per channel
- **VU Meter:** -80 to +20 dB, USB input level meter per channel (visible in System View only)

Audio – Output Channels

Analog Output Channels 1 – 6 (DSP-860) or 1 – 8 (DSP-128x):
- **Signal Types:** Mono analog line level, adjacent channels groupable for stereo
- **Signal Flow:** Matrix Mixer > Channel Trim > Channel Strip > VU Meter > DAC > Output
- **Channel Trim Settings:**
  - **Mute:** Matrix output bus mute enable/disable per channel
  - **Gain:** -80 to +20 dB, matrix output bus gain adjustment per channel
- **VU Meter:** -80 to +20 dB, analog output level meter per channel (visible in System View only)
- **Channel Strip:**
  - **Object 1:** Dynamics – Choice of Compressor, Limiter, or AGC
  - **Object 2:** EQ/Filter – Choice of 3-Band TEQ, 10-Band GEQ, 10-Band PEQ, 10-Band Notch Filter, LP Filter, HP Filter, BP Filter, or Crossover w/5-Band PEQ
  - **Object 3:** EQ/Filter – Choice of 3-Band TEQ, 10-Band GEQ, 10-Band PEQ, 10-Band Notch Filter, LP Filter, HP Filter, BP Filter, or Crossover w/5-Band PEQ
  - **Object 4:** Delay
  - **Object 5:** Dynamics – Choice of Compressor, Limiter, or AGC
  - **Object 6:** Output Level

Reference Output Channels 1 – 2 (Internal):
- **Signal Flow:** Matrix Mixer > Reference Signal to Channel Strip Objects

Auxiliary Output Channels 1 – 8 (Internal):
- **Signal Flow:** Matrix Mixer > Channel Trim > VU Meter > Feed to Corresponding Auxiliary Input Channel
- **Channel Trim Settings:**
  - **Mute:** Matrix output bus mute enable/disable per channel
  - **Gain:** -80 to +20 dB, matrix output bus gain adjustment per channel
- **VU Meter:** -80 to +20 dB, auxiliary output level meter per channel (visible in System View only)

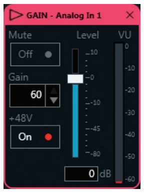
Phone Output Channel (DSP-1282 & DSP-1283 only):
- **Signal Flow:** Matrix Mixer > Channel Trim > VU Meter > Codec
- **Channel Trim Settings:**
  - **Mute:** Matrix output bus mute enable/disable
  - **Gain:** -80 to +20 dB, matrix output bus gain adjustment
- **VU Meter:** -80 to +20 dB, phone output level meter (visible in System View only)

Dante Output Channels 1 – 32 (DSP-1281 & DSP-1283 only):
- **Signal Flow:** Matrix Mixer > Channel Trim > VU Meter > LAN
- **Channel Trim Settings:**
  - **Mute:** Matrix output bus mute enable/disable per channel
  - **Gain:** -80 to +20 dB, matrix output bus gain adjustment per channel
- **VU Meter:** -80 to +20 dB, Dante output level meter per channel (visible in System View only)

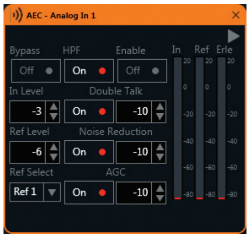
USB Output Channels 1 – 8 (DSP-1282 & DSP-1283 only):
- **Signal Flow:** Matrix Mixer > Channel Trim > VU Meter > USB
- **Channel Trim Settings:**
  - **Mute:** Matrix output bus mute enable/disable per channel
  - **Gain:** -80 to +20 dB, matrix output bus gain adjustment per channel
- **VU Meter:** -80 to +20 dB, USB output level meter per channel (visible in System View only)

Audio – Channel Strip Objects

Gain:
- **GAIN:** Mic Preamp / Line Input Gain Stage
  - **Settings:**
    - **Mute:** On/Off, mute enable/disable
    - **Gain:** -18 to +60 dB in 3 dB increments, analog gain adjustment
    - **+48V:** On/Off, phantom power enable/disable
    - **Level:** -80 to +10 dB, level adjustment
    - **VU:** -60 to 0 dB, level meter (post-Gain & ADC, pre-Level & Mute)



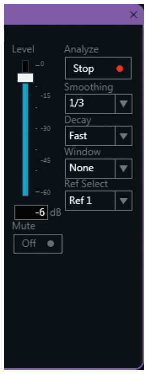
Acoustic Echo Canceller (DSP-1282 & DSP-1283 only):
- **AEC:** Acoustic Echo Canceller
  - **Global Settings:**
    - **Bypass:** On/Off, bypass all enable/disable
    - **Mode:** On/Off, AEC enable/disable
  - **AEC Settings:**
    - **Bypass:** On/Off, AEC bypass enable/disable
    - **HPF:** On/Off, AEC 80 Hz high-pass filter enable/disable
    - **Enable:** On/Off, AEC enable/disable
    - **In Level:** -60 to 0 dB, AEC input level adjustment
    - **Ref Level:** -60 to 0 dB, reference signal level adjustment
    - **Ref Select:** Ref 1 – 2, reference signal bus selection
    - **Double Talk:** On/Off, double-talk suppression enable/disable; -30 to 0 dB, double-talk suppression adjustment
    - **Noise Reduction:** On/Off, noise reduction enable/disable; -30 to 0 dB, noise reduction adjustment
    - **AGC:** On/Off, automatic gain control enable/disable; -30 to 0 dB, automatic gain control adjustment
    - **In:** -80 to +20 dB, AEC input level meter
    - **Ref:** -80 to +20 dB, reference signal level meter
    - **Erle:** -80 to +20 dB, echo return loss enhancement meter



EQ/Filter:

Spectrum Analyzer:

Analyzer Settings:



Level: -60 to 0 dB, signal generator level adjustment

Mute: On/Off, signal generator mute enable/disable

Analyze: Start/Stop, analyzer start/stop control

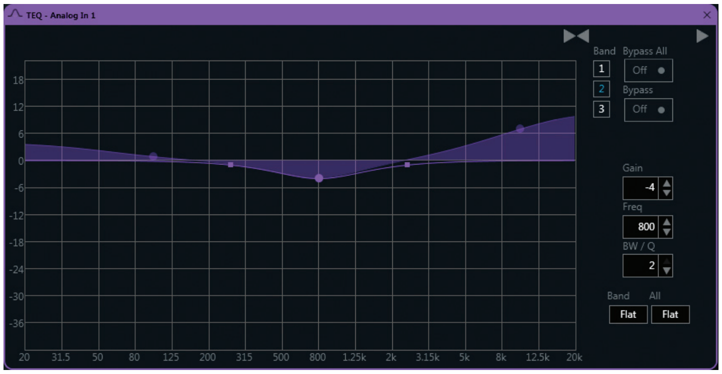
Smoothing: None, 1/24, 1/12, 1/6, 1/3, or 1 octave; display resolution selection

Decay: Fast, 1/2, 1, 2, 4, or 8 seconds; decay time selection

Window: None, Hamming, Hanning, or Blackman; window type selection

Ref Select: Ref 1 – 2, reference signal bus selection

TEQ: Tone Equalizer



3-Band TEQ Settings:

Band 1 – 3: Selects or indicates which band is being adjusted

Bypass All: On/Off, all EQ filters enable/disable

Bypass (per band): On/Off, individual EQ filter enable/disable

Gain (per band): -20 to +20 dB, EQ filter gain adjustment

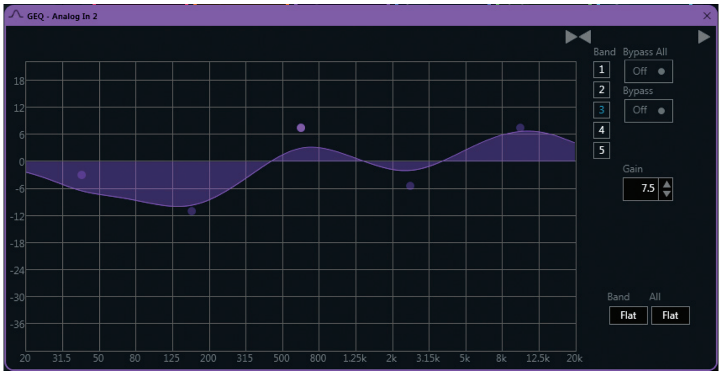
Freq (per band): 20 to 20000 Hz, EQ filter frequency adjustment

BW/Q (band 2 only): 0.1 to 2.0, EQ filter bandwidth adjustment

Band (per band): Flat, sets the individual EQ filter gain to 0 dB

All: Flat, sets all EQ filters to 0 dB

GEQ: Graphic Equalizer



5-Band GEQ Settings:

Band 1 – 5: Selects or indicates which band is being adjusted

Bypass All: On/Off, all EQ filters enable/disable

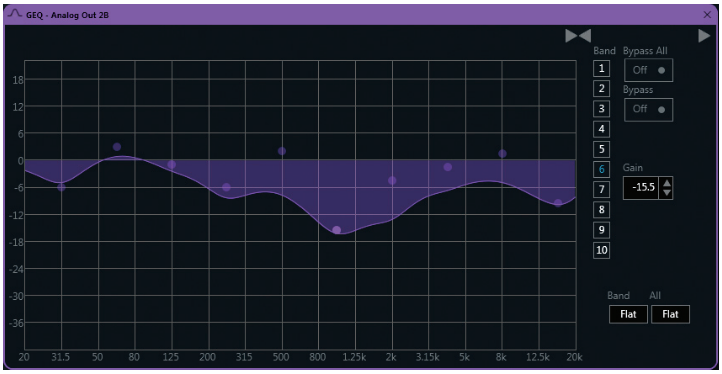
Bypass (per band): On/Off, individual EQ filter enable/disable

Center Frequencies: 40, 160, 630, 2.5k, 10k Hz (fixed)

Gain (per band): -20.0 to +20.0 dB in 0.5 dB increments, EQ filter gain adjustment

Band (per band): Flat, sets the individual EQ filter gain to 0 dB

All: Flat, sets all EQ filters to 0 dB



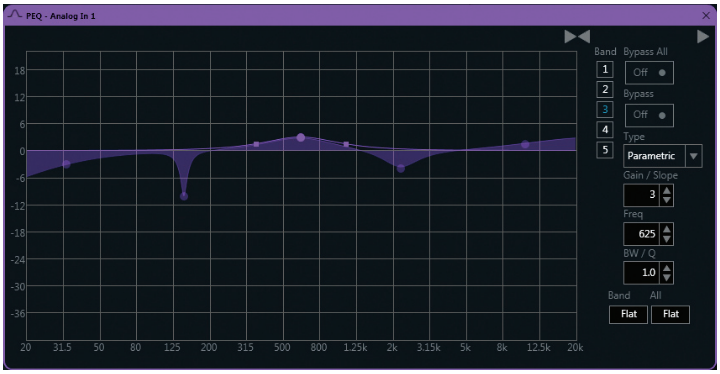
10-Band GEQ Settings:

Band 1 – 10: Selects or indicates which band is being adjusted

Center Frequencies: 31.5, 63, 125, 250, 500, 1k, 2k, 4k, 8k, 16k Hz (fixed)

(All other parameters same as 5-Band GEQ)

PEQ: Parametric Equalizer



5-Band PEQ Settings:

Band 1 – 5: Selects or indicates which band is being adjusted

Bypass All: On/Off, all EQ filters enable/disable

Bypass (per band): On/Off, individual EQ filter enable/disable

Type (per band): Parametric, Notch, Low shelf, High shelf, Low cut, or High cut; EQ filter type selection

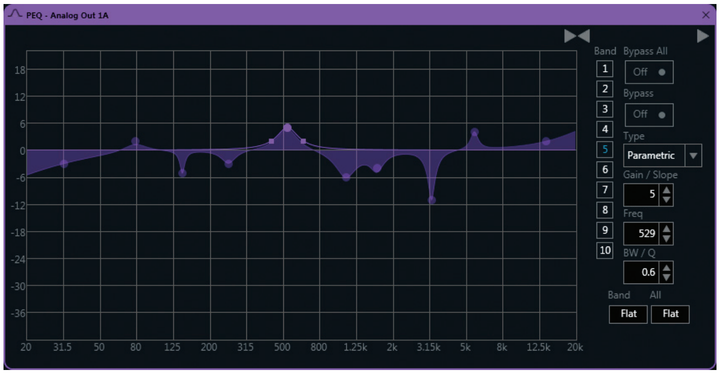
Gain/Slope (per band): -40 to +20 dB, parametric filter gain adjustment; or -40 to 0 dB, notch filter gain adjustment; or –20 to + 20 dB, shelf filter gain adjustment; or 6 or 12 dB, cut filter slope selection

Freq (per band): 20 to 20000 Hz, EQ filter frequency adjustment

BW/Q (per band): 0.1 to 2.0 octave, parametric or cut filter bandwidth adjustment; or 0.05 to 1.00 octave, notch filter bandwidth adjustment

Band (per band): Flat, sets the individual EQ filter gain to 0 dB (or 6 dB slope for cut filters)

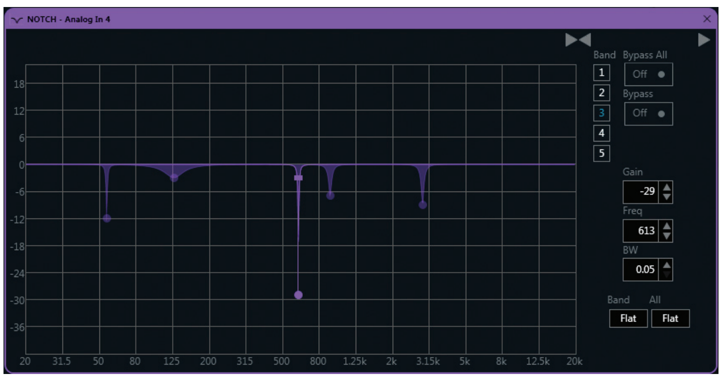
All: Flat, sets all EQ filters to 0 dB (or 6 dB slope for cut filters)



10-Band PEQ Settings:

Band 1 – 10: Selects or indicates which band is being adjusted (All other parameters same as 5-Band PEQ)

NOTCH: Notch Filter



5-Band Notch Filter Settings:

Band 1 – 5: Selects or indicates which band is being adjusted

Bypass All: On/Off, all notch filters enable/disable

Bypass (per band): On/Off, individual notch filter enable/disable

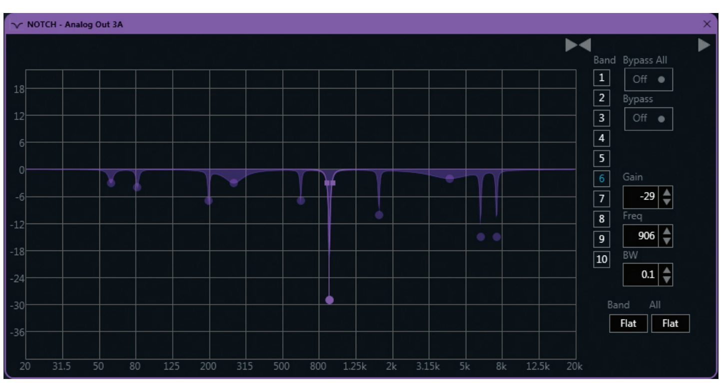
Gain (per band): -40 to 0 dB, notch filter gain adjustment

Freq (per band): 20 to 20000 Hz, notch filter frequency adjustment

BW (per band): 0.05 to 1.00 octave, notch filter bandwidth adjustment

Band (per band): Flat, sets the individual notch filter gain to 0 dB

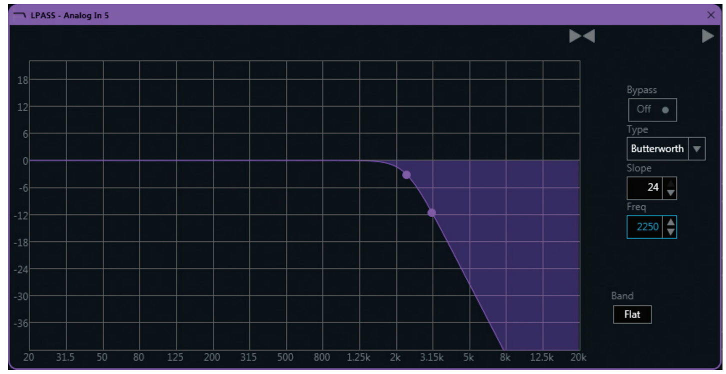
All: Flat, sets all notch filters to 0 dB



10-Band Notch Filter Settings:

Band 1 – 10: Selects or indicates which band is being adjusted (All other parameters same as 5-Band Notch Filter)

LPASS: Low-Pass Filter



LP Filter Settings:

Bypass: On/Off, LP filter enable/disable

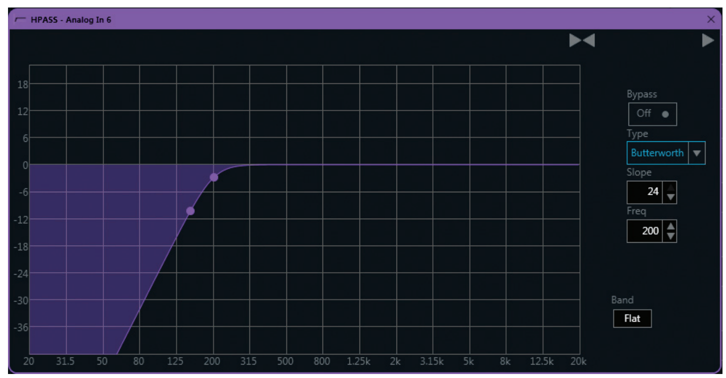
Type: Butterworth, Chebyshev, Bessel, or Link-Riley; LP filter type selection

Slope: 6, 12, 18, or 24 dB, LP filter slope selection

Freq: 20 to 20000 Hz, LP filter frequency adjustment

Band: Flat, sets the LP filter frequency to 20 kHz

HPASS: High-Pass Filter



HP Filter Settings:

Bypass: On/Off, HP filter enable/disable

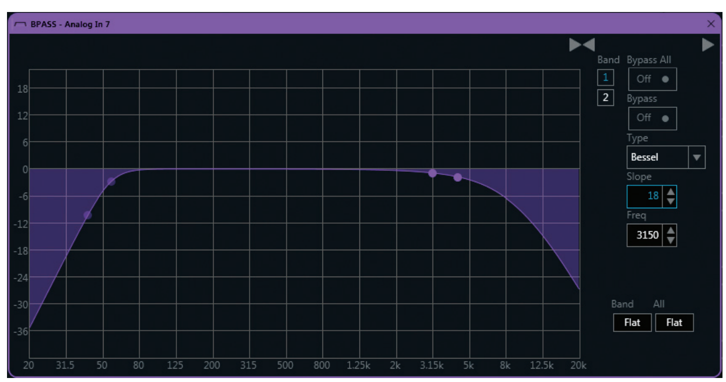
Type: Butterworth, Chebyshev, Bessel, or Link-Riley; HP filter type selection

Slope: 6, 12, 18, or 24 dB, HP filter slope selection

Freq: 20 to 20000 Hz, HP filter frequency adjustment

Band: Flat, sets the HP filter frequency to 20 Hz

BPASS: BandPass Filter



BP Filter Settings:

Band 1 – 2: Selects or indicates which band is being adjusted

Bypass All: On/Off, all filters enable/disable

Bypass (per band): On/Off, individual LP or HP filter enable/disable

Type (per band): Butterworth, Chebyshev, Bessel, or Link-Riley; LP or HP filter type selection

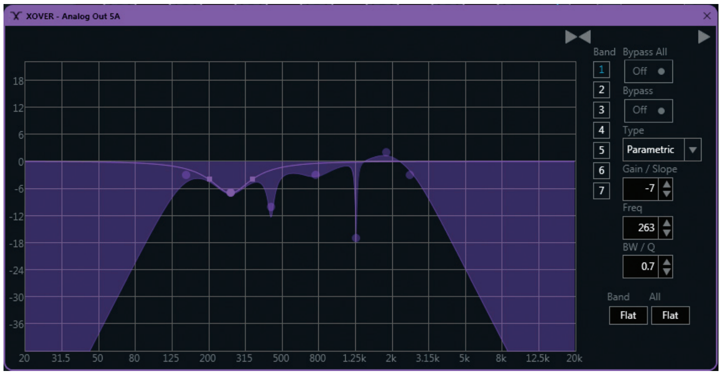
Slope (per band): 6, 12, 18, or 24 dB, LP or HP filter slope selection

Freq (per band): 20 to 20000 Hz, LP or HP filter frequency adjustment

Band (per band): Flat, sets the LP filter frequency to 20 kHz or the HP filter frequency to 20 Hz

All: Flat, sets the LP filter frequency to 20 kHz and the HP filter frequency to 20 Hz

XOVER: Low, Mid, or High Frequency Crossover w/Parametric Equalizer



Crossover & 5-Band PEQ Settings:

Band 1 – 7: Selects or indicates which band is being adjusted

Bypass All: On/Off, all filters enable/disable

Bypass (per band): On/Off, individual EQ, LP, or HP filter enable/disable

Type (per band 1 – 5): Parametric, Notch, Low shelf, High shelf, Low cut, or High cut; EQ filter type selection

Type (per band 6 – 7): Butterworth, Chebyshev, Bessel, or Link-Riley; LP or HP filter type selection

Gain/Slope (per band 1 – 5): -40 to +20 dB, parametric filter gain adjustment; or -40 to 0 dB, notch filter gain adjustment; or –20 to + 20 dB, shelf filter gain adjustment; or 6 or 12 dB, cut filter slope selection

Gain/Slope (per band 6 – 7): 6, 12, 18, or 24 dB, LP or HP filter slope selection

Freq (per band): 20 to 20000 Hz, EQ, LP, or HP filter frequency adjustment

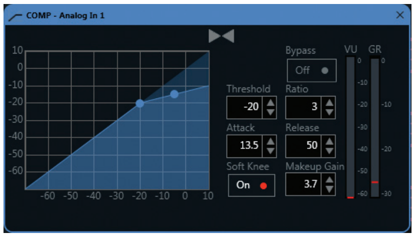
BW/Q (per band 1 – 5): 0.1 to 2.0 octave, parametric or cut filter bandwidth adjustment; or 0.05 to 1.00 octave, notch filter bandwidth adjustment

Band (per band 1 – 5): Flat, sets the individual EQ filter gain to 0 dB (or 6 dB slope for cut filters)

All (per band 1 – 5): Flat, sets all EQ filters to 0 dB (or 6 dB slope for cut filters)

Dynamics:

COMP: Compressor



Compressor Settings:

Bypass: On/Off, compressor enable/disable

Threshold: -60 to 0 dB, compression threshold adjustment

Ratio: 1.0:1 to 50.0:1, compression ratio adjustment

Attack: 0.1 to 500.0 ms, attack time adjustment

Release: 10 to 2000 ms, release time adjustment

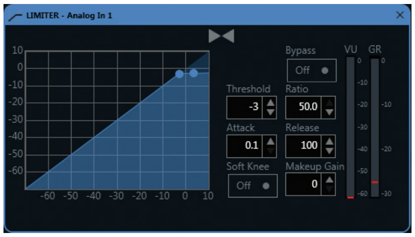
Soft Knee: On/Off, soft knee curve enable/disable

Makeup Gain: 0.0 to 20.0 dB, makeup gain adjustment

VU: -60 to 0 dB, post-compression level meter

GR: -30 to 0 dB, gain reduction level meter

LIMITER: Limiter



Limiter Settings:

Bypass: On/Off, limiter enable/disable

Threshold: -60 to 0 dB, limiting threshold adjustment

Ratio: 1.0:1 to 50.0:1, limiting ratio adjustment

Attack: 0.1 to 500.0 ms, attack time adjustment

Release: 10 to 2000 ms, release time adjustment

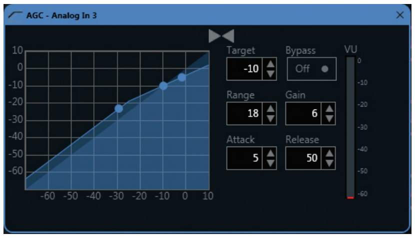
Soft Knee: On/Off, soft knee curve enable/disable

Makeup Gain: 0.0 to 20.0 dB, makeup gain adjustment

VU: -60 to 0 dB, post-limiting level meter

GR: -30 to 0 dB, gain reduction level meter

AGC: Automatic Gain Control



AGC Settings:

Bypass: On/Off, AGC enable/disable

Target: -60 to 0 dB, target level adjustment

Range: 0 to 40 dB, dynamic range adjustment

Gain: 0.0 to 20.0 dB, post-AGC gain adjustment

Attack: 0.1 to 50.0 ms, attack time adjustment

Release: 10 to 500 ms, release time adjustment

VU: -60 to 0 dB, post-AGC level meter

Delay:

DELAY: Time Alignment Delay

Delay Settings:

Bypass: On/Off, delay enable/disable

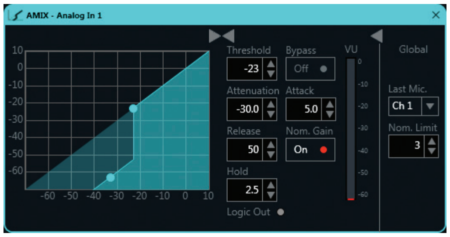
Delay: 0 to 2000 ms, delay time adjustment

DSP-128 Avia™ DSP Matrix Mixer

Automixer/Gate/Ducker:

AMIX: Automixer (Automatic Microphone Mixer)

Automixer Settings:



Bypass: On/Off, automixer enable/disable

Threshold: -60 to 0 dB, gate threshold adjustment

Attenuation: -40.0 to 0.0 dB, gate attenuation adjustment

Attack: 0.1 to 50.0 ms, attack time adjustment

Release: 10 to 500 ms, release time adjustment

Hold: 0.1 to 20.0 s, hold time adjustment

Nom. Gain: On/Off, NOM attenuation enable/disable

Logic Out: Gate open (mic on) indicator

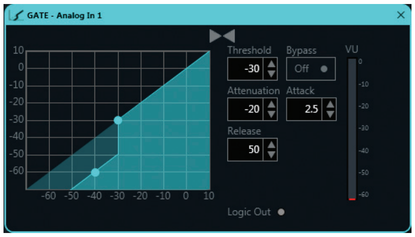
VU: -60 to 0 dB, post-gate level meter

Last Mic. (global): None, Ch 1 – 12, or Last; selects which analog input channel's gate remains open (mic on) when all input signals are below threshold

Nom. Limit (global): 1-10, maximum NOM (Number of Open Mics) selection

GATE: Noise Gate

Gate Settings:



Bypass: On/Off, gate enable/disable

Threshold: -60 to 0 dB, gate threshold adjustment

Attenuation: -40.0 to 0.0 dB, gate attenuation adjustment

Attack: 0.1 to 50.0 ms, attack time adjustment

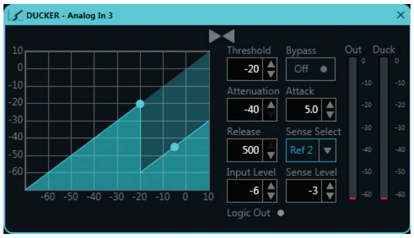
Release: 10 to 500 ms, release time adjustment

Logic Out: Gate open (channel on) indicator

VU: -60 to 0 dB, post-gate level meter

DUCKER: Ducker

Ducker Settings:



Bypass: On/Off, ducker enable/disable

Threshold: -60 to 0 dB, ducking threshold adjustment

Attenuation: -40.0 to 0.0 dB, ducking attenuation adjustment

Attack: 0.1 to 50.0 ms, attack time adjustment

Release: 10 to 500 ms, release time adjustment

Sense Select: Ref 1 – 2, reference signal bus selection

Input Level: -60 to 0 dB, input level adjustment

Sense Level: -60 to 0 dB, sense level adjustment

Logic Out: Ducking active indicator

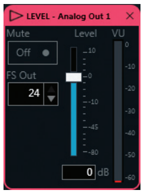
Out: -60 to 0 dB, post-ducking level meter

Duck: -60 to 0 dB, reference signal level meter

Output Level:

LEVEL: Output Level Control

Output Level Settings:



Mute: ON/OFF, output mute enable/disable

FS Out: 0, 6, 12, 18, or 24 dB, analog output gain selection

Level: -80 to +10 dB, output level adjustment

VU: -60 to 0 dB, output level meter (pre-DAC & "FS Out" Gain)

Communications

Ethernet: 10/100/1000 Mbps, auto-switching, auto-negotiating, auto-discovery, full/half duplex, TCP/IP, UDP/IP, CIP, DHCP, SSL, SSH, SFTP (SSH File Transfer Protocol)

Phone/VoIP (DSP-1282 & DSP-1283 only): POTS (RJ11 wired interface) supporting DTMF & Caller ID, SIP (via Ethernet) supporting SIP peer-to-peer or SIP server mode

Dante (DSP-1281 & DSP-1283 only): Dedicated 1000 Mbps primary and secondary Dante network ports

USB Audio Device (DSP-1282 & DSP-1283 only): USB 2.0 device port for a USB Audio device

USB Device: USB device port for computer console (setup)

Connectors

MIC/LINE INPUTS 1 – 8 (DSP-860) or 1 – 12 (DSP-128x):
(8 or 12) 3-pin 3.5 mm detachable terminal blocks;
Balanced microphone/line-level audio inputs;
Input Level: +24 dBu maximum;
Gain Range: 66 dB;
Input Impedance: 10k Ohms balanced;
Phantom Power: +48 Volts DC, 12 mA, software enabled/disabled per channel

LINE OUTPUTS 1 – 6 (DSP-860) or 1 – 8 (DSP-128x):
(6 or 8) 3-pin 3.5 mm detachable terminal blocks;
Balanced/unbalanced line-level audio outputs;
Output Level: +24 dBu maximum;
Output Impedance: 150 Ohms balanced

USB (DSP-1282 & DSP-1283 only): (1) USB Type B female;
USB 2.0 device port for USB Audio

DANTE, PRI (DSP-1281 & DSP-1283 only): (1) 8-pin RJ45 female;
1000Base-T Primary Dante network port

DANTE, SEC (DSP-1281 & DSP-1283 only): (1) 8-pin RJ45 female;
1000Base-T Secondary Dante network port

VOIP (DSP-1282 & DSP-1283 only): (1) 8-pin RJ45 female;
10Base-T/100Base-TX/1000Base-T Ethernet SIP VoIP network port

LAN: (1) 8-pin RJ45 female;
10Base-T/100Base-TX/1000Base-T Ethernet LAN port

PHONE (DSP-1282 & DSP-1283 only): (1) RJ11 female;
POTS analog telephone port

100-240V~1.8A 50/60 Hz: (1) IEC 60320 C14 main power inlet; Mates with removable power cord, included

G: (1) 6-32 screw, chassis ground lug

COMPUTER (front): (USB Type B female; USB computer console port (for setup only)

Controls & Indicators

PWR: (1) Bi-color green/amber LED, indicates operating power supplied from AC line power, turns amber while booting and green when operating, alternates colors if no network connection
RESET: (1) Recessed pushbutton, restores last saved settings
MIC/LINE INPUTS 1 – 12 (DSP-1281 & DSP-1283 only): (12) 5-segment LED bar graph audio level meters for each corresponding input; each contains (4) blue LEDs for -40, -30, -20, and -10 dBFS, and (1) red LED for CLIP (-2 dBFS)
LINE OUTPUTS 1 – 8 (DSP-1281 & DSP-1283 only): (8) 5-segment LED bar graph audio level meters for each corresponding output; each contains (4) blue LEDs for -40, -30, -20, and -10 dBFS, and (1) red LED for CLIP (-2 dBFS)
DANTE, PRI (rear, DSP-1281 & DSP-1283 only): (2) Bi-color green/amber LEDs, indicate primary Dante network activity and link status
DANTE, SEC (rear, DSP-1281 & DSP-1283 only): (2) Bi-color green/amber indicate secondary Dante network activity and link status
VOIP (rear, DSP-1282 & DSP-1283 only): (2) Bi-color green/amber LEDs, indicate SIP VoIP network activity and link status
LAN (rear): (2) Bi-color green/amber LEDs, indicate Ethernet activity and link status
SETUP (rear): (1) Red LED and (1) pushbutton for Ethernet setup

Power Requirements

Main Power: 1.8 Amps @ 100-240 Volts AC, 50/60 Hz
Power Consumption: 30 Watts typical

Environmental

Temperature: 41° to 104° F (5° to 40° C)
Humidity: 10% to 90% RH (non-condensing)
Heat Dissipation: 102 BTU/hr
Ambient Noise Level: 30 dBA

Enclosure

Chassis: Metal, fan-cooled, vented sides
Front Panel: Metal, black finish with polycarbonate label overlay
Mounting: Freestanding or 1 RU 19-inch rack-mountable (adhesive feet and rack ears included)

Dimensions

Height: 1.72 in (44 cm) without feet
Width: 17.28 in (439 mm);
19.00 in (483 mm) with rack ears
Depth: 14.35 in (365 mm)

Weight

DSP-860: 8.4 lb (3.9 kg)
DSP-1280: 8.8 lb (4.0 kg)
DSP-1281: 9.2 lb (4.1 kg)
DSP-1282: 9.2 lb (4.1 kg)
DSP-1283: 9.3 lb (4.2 kg)

MODELS & ACCESSORIES

Available Models

DSP-860: Avia™ 8x6 DSP Matrix Mixer
DSP-1280: Avia™ 12x8 DSP Matrix Mixer
DSP-1281: Avia™ 12x8 DSP Matrix Mixer w/Dante™
DSP-1282: Avia™ 12x8 DSP Matrix Mixer w/USB Audio, AEC, & Audio Conferencing Interface
DSP-1283: Avia™ 12x8 DSP Matrix Mixer w/Dante™, USB Audio, AEC, & Audio Conferencing Interface

Available Accessories

AMP-8000 Series: Networked 8-Channel Power Amplifiers

Notes:

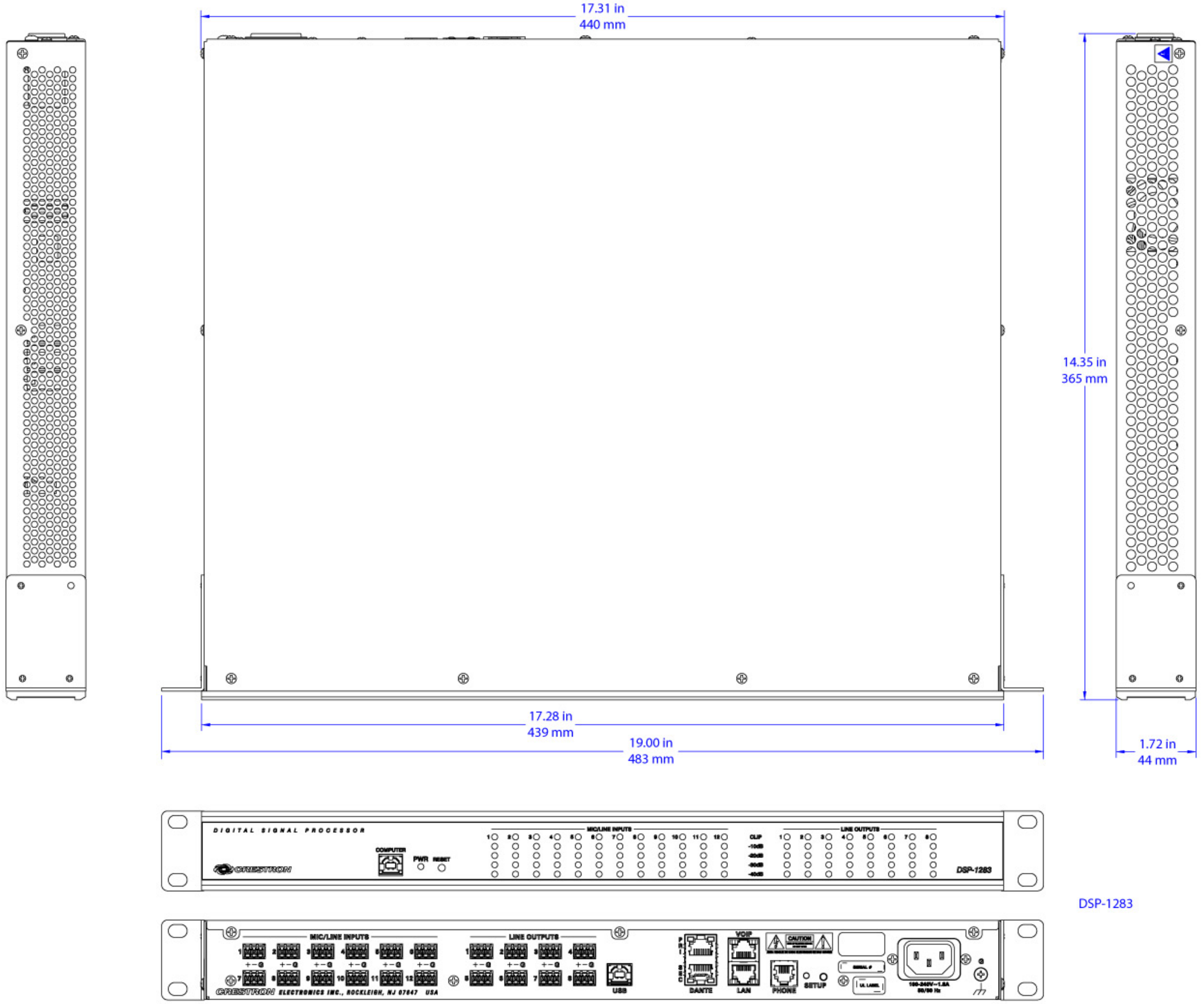
1. Crestron control system, touch screens, keypads, Crestron Fusion, and custom programming are sold separately. A control system is typically required for any installation that includes end-user adjustment or selection of presets, automated interaction with other equipment, and/or monitoring via Crestron Fusion.
2. Dante, USB Audio, SIP, and POTS inputs and outputs are available on select models only. They are fully routable via the matrix mixer but do not include channels strips. A channel strip may be added to any of these inputs or outputs by routing the signal through any of the eight auxiliary buses via the matrix mixer.

This product may be purchased from an authorized Crestron dealer. To find a dealer, please contact the Crestron sales representative for your area. A list of sales representatives is available online at www.crestron.com/salesreps or by calling 800-237-2041.

The specific patents that cover Crestron products are listed online at: patents.crestron.com.

Certain Crestron products contain open source software. For specific information, please visit www.crestron.com/opensource.

DIMENSIONAL DIAGRAMS



Model DSP-1283 shown. Dimensions typical of all models.

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