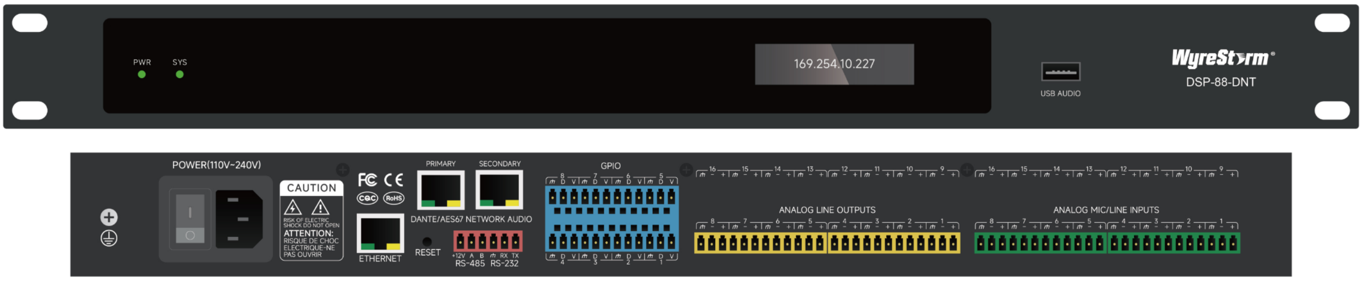


DSP-88-DNT

Digital Signal Processor

16 input 16 output Matrix DSP with Dante

SHARC, 40-bit floating point processor

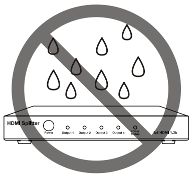
24bit/48KHz sampling rate

In-built USB sound card

User Manual

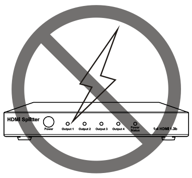
Version: V1.0.3

Important Safety Instructions

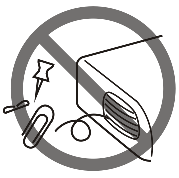


1. Do not expose this apparatus to rain, moisture, dripping or splashing and that no objects filled with liquids, such as vases, shall be placed on the apparatus.

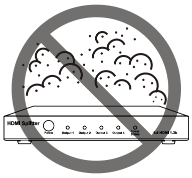
2. Do not install or place this unit in a bookcase, built-in cabinet or in another confined space. Ensure the unit is well ventilated.



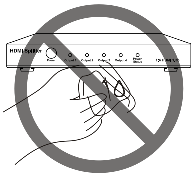
3. To prevent risk of electric shock or fire hazard due to overheating, do not obstruct the unit's ventilation openings with newspapers, tablecloths, curtains, and similar items.



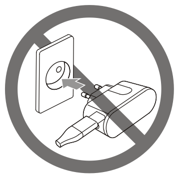
4. Do not install near any heat sources such as radiators, heat registers, stoves, or other apparatus (including amplifiers) that produce heat.



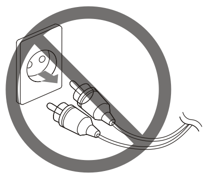
5. Do not place sources of naked flames, such as lighted candles, on the unit.



6. Clean this apparatus only with dry cloth.

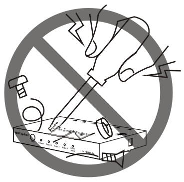


7. Unplug this apparatus during lightning storms or when unused for long periods of time.



8. Protect the power cord from being walked on or pinched particularly at plugs.

9. Only use attachments / accessories specified by the manufacturer.



10. Refer all servicing to qualified service personnel.

Table of Contents

1. Introduction

1.1 Overview

The WyreStorm DSP-88-DNT is a high-performance audio processing platform designed for professional AV integration, conferencing, and live sound applications. Built around an Analog Devices SHARC processor, the DSP Series delivers 32-bit and 40-bit floating-point processing — providing exceptional dynamic range, minimal distortion, and virtually no clipping artefacts across the entire signal chain.

The DSP is configured and controlled via DSP Controller, a Windows-based application that provides full access to all processing modules, parameter adjustment, GPIO mapping, panel configuration, and preset management. Up to four users can connect simultaneously, and configurations can be saved to any of the device's 16 onboard presets.

Typical deployment scenarios include boardrooms, lecture theatres, houses of worship, broadcast studios, and any environment requiring precise audio routing, mixing, and processing.

1.2 Features

- ADI SHARC DSP engine with 32-bit / 40-bit floating-point processing.
- Up to 16 analog inputs and 16 analog outputs via removable balanced Phoenix connectors.
- Per-channel +48 V phantom power for condenser microphones.
- Comprehensive DSP module library: Expander, Compressor/Limiter, Auto Gain Control, Parametric and Graphic Equalizers, Feedback Inhibition, Noise Gate, Ducker, AutoMixer, Echo Canceller, Noise Suppression, Matrix Router, High/Low Pass Filters, Delayer, and more.
- 16 user-definable presets with flexible startup mode selection.
- 8 GPIO ports configurable as digital inputs or outputs for preset recall, mute, gain control, and command sending.
- Optional OLED display panel and key panel for on-site control.
- USB soundcard (1-in / 1-out) for recording and PC-based conferencing.
- External control via UDP (default port 50000) and RS-232 / RS-485.
- Windows-based DSP Controller software with drag-and-drop module design, custom operator panels, and built-in level metering.

1.3 Package Contents

- 1x DSP-88-DNT
- 1x IEC Power Cable (region-specific)
- Phoenix connector blocks for Input, Output, RS-232/RS-485, and GPIO
- 1x CAT5 Ethernet cable
- 4x rack-mounting screws
- 1x Quick Start Guide

Note: *Inspect the package on arrival. If any items are missing or damaged, contact your supplier immediately.*

1.4 Specifications

Processor	ADI SHARC 21489
Sampling Rate / Bit Depth	48 kHz / 24-bit
Input Gain Range	0 / 3 / 6 / 9 / 12 / 15 / 18 / 21 / 24 / 27 / 30 / 33 / 36 / 39 / 42 / 45 / 48 dBu
Phantom Power	+48 V DC (per channel, switchable)
Frequency Response (20 Hz – 20 kHz)	±0.3 dB
Maximum Input / Output Level	+18 dBu
THD + Noise	0.001% @ 4 dBu (A-weighted)
Input Dynamic Range	113 dB
Output Dynamic Range	113 dB
Channel Isolation @ 1 kHz	108 dB
Input Impedance (Balanced)	5.4 kΩ
Output Impedance (Balanced)	600 Ω
System Delay	< 3 ms
A/D Sampling Rate	48 kHz
A/D Audio Format	24-bit MSB TDM
D/A Sampling Rate	48 kHz
D/A Audio Format	24-bit MSB TDM
Power Supply	AC 110 – 240 V, 50/60 Hz
Maximum Power Consumption	< 40 W
Dimensions (W × D × H)	482 × 260 × 45 mm (1U)
Shipping Weight	4 kg
Operating Temperature	0 to 45°C (32 to 113°F), 10% to 90% non-condensing

2. Technology Overview

2.1 A/D Conversion

The DSP Series supports up to 16 fixed analog audio inputs connected via removable balanced Phoenix connectors. Per-channel preamp gain and phantom power are controlled via DSP Controller. The analog-to-digital converter operates at 48 kHz with 24-bit resolution in MSB TDM format.

Sampling Rate	48 kHz
THD+N	≤ 0.001% @ 4 dBu (A-weighted)
Dynamic Range	110 dB
Audio Format	24-bit MSB TDM

2.2 D/A Conversion

The digital-to-analog converter uses a 24-bit 256× oversampling architecture. Unit gain (0 dB) is calibrated to +4 dBu with 20 dB of headroom, meaning 0 dBFS at the digital input corresponds to +24 dBu at the analog output. Output level may be adjusted globally via the output volume fader in DSP Controller.

Sampling Rate	48 kHz
THD+N	≤ 0.001% @ 4 dBu (A-weighted)
Dynamic Range (A-weighted)	112 dB
Audio Format	24-bit MSB TDM

2.3 Floating-Point Processing

The SHARC DSP engine provides 32-bit and 40-bit floating-point processing. Unlike fixed-point systems, floating-point arithmetic distributes available bits across all signal levels, eliminating the data loss that occurs in fixed-point systems when gain is attenuated or boosted significantly.

In fixed-point systems, attenuating a 24-bit signal by 42 dB leaves only 17 bits of valid data. Floating-point processing retains full precision regardless of gain stage changes. Additionally, clipping distortion introduced early in a fixed-point chain is permanent — even if the level is corrected downstream. Floating-point processing allows the output section to simply correct the level without any audible artefact.

Practical benefit: the gain structure between processing modules can be managed freely. If a module significantly attenuates a signal and a later module restores it, no information is lost. Designers need only ensure the final output level before the D/A converter is set correctly.

3. Panel Description

3.1 Front Panel

The front panel provides status indication, USB soundcard access, and the USB connection required for firmware upgrades.

#	Name	Description
1	POWER LED	On: Device is powered on. Off: Device is powered off.
2	STATUS LED	On: Device is operating normally. Blinking: Device is processing or initialising. Off: Device is in standby or fault state.
3	USB AUDIO	USB Type-A soundcard connector (1-in / 1-out). Connect to a PC via USB Type-A to Type-A cable to enable recording, playback, and PC-based teleconferencing via the built-in USB audio driver.

3.2 Rear Panel

All audio connections, control interfaces, and the power inlet are located on the rear panel.

#	Name	Description
1	POWER	IEC power inlet and rocker on/off switch. Connect to AC 110–240 V, 50/60 Hz mains supply.
2	ETHERNET	RJ-45 network control port. Connect to a PC or LAN to access the DSP Controller software. Default IP: 169.254.10.227 / Subnet mask: 255.255.0.0.
3	RS-232 / RS-485	Serial interface for connection to central control systems or third-party controllers. Default baud rate: 115200, 8N1.
4	GPIO 1–8	Eight user-configurable general-purpose I/O ports. Each port may be independently set as a digital input or output (logic levels: 0 V or 5 V). Supports preset recall, gain control, mute, routing, and command sending functions.
5	OUTPUT 1–16	Balanced analog audio outputs via Phoenix connectors. Connect to power amplifiers, active speakers, or other downstream devices.
6	INPUT 1–16	Balanced analog audio inputs via Phoenix connectors. Accepts microphone (with or without phantom power) and line-level sources including mixers, DVD players, and other balanced equipment.

4. Installation and Wiring

> **WARNING:** Disconnect all power before making or modifying any connections. Connect and disconnect cables gently to avoid damage to connectors and equipment.

4.1 Rack Mounting

1. Align the device with the rack rails. The DSP occupies 1U of rack space.
2. Insert the supplied rack screws through the front-panel ears into the rack rails.
3. Tighten the screws securely. Do not overtighten.
4. Ensure at least 1U (44.5 mm) of free space above and below the unit for adequate ventilation.

4.2 Network Connection

5. Connect an RJ-45 CAT5 (or better) cable from the ETHERNET port on the rear panel to a PC or network switch.

6. If connecting directly to a PC (not via a switch), ensure the PC's network adapter is configured in the same subnet as the device. The default device address is 169.254.10.227 / 255.255.0.0.
7. Launch DSP Controller. Click the search button to discover the device on the network.

4.3 Audio Connections

4.3.1 Balanced Connection (Recommended)

The device supports balanced XLR and 3-pin Phoenix connections. Balanced connections provide superior noise rejection and are strongly recommended for all signal runs longer than a few metres. For XLR: pin 1 = Ground, pin 2 = Hot (+), pin 3 = Cold (–). For Phoenix connectors, observe the silkscreen labelling on the rear panel.

4.3.2 Unbalanced Connection

Unbalanced signals (RCA or 1/4" TS) may be connected via an appropriate adapter. Use the shortest cable run practical and avoid routing unbalanced cables parallel to mains power wiring. Signal levels must be compatible with the device's +18 dBu maximum input level.

4.4 Phantom Power

Phantom power (+48 V DC) is available on each input channel independently and is enabled via DSP Controller. Enable phantom power only when connecting condenser microphones that require it. Do not enable phantom power when connecting dynamic microphones, ribbon microphones, or line-level sources, as doing so may damage the connected equipment.

4.5 GPIO Wiring

The 8 GPIO ports accept or output 0 V (logic low) or 5 V (logic high) signals. Refer to section 3.4.3 (GPIO Setting) for configuration. When using GPIO inputs, ensure the driving source can supply or sink adequate current at the correct voltage levels. Do not connect GPIO ports to mains voltage or signals exceeding 5 V.

4.6 RS-232 / RS-485 Connection

Connect the RS-232/RS-485 port on the rear panel to the serial port of a central control processor or computer using an appropriate cable. Default serial parameters: 115200 baud, 8 data bits, no parity, 1 stop bit (8N1). These parameters may be changed in DSP Controller under Device Setting.

5. Software

5.1 Software Installation

DSP Controller is a Windows-based application that runs on any PC meeting the following minimum requirements:

Operating System	Windows 7 or later
Processor	1 GHz or higher
Memory	2 GB RAM or more
Storage	1 GB of free disk space
Display	1024 × 768, 24-bit colour or higher
Network	Ethernet (RJ-45) interface
Cable	CAT5 Ethernet cable

The DSP Controller software is embedded in the device and can be downloaded directly from it:

8. Connect a PC to the ETHERNET port on the rear panel.
9. Add a static IP address in the 169.254.x.x range to your PC's network adapter (e.g., 169.254.10.1 / 255.255.0.0).
10. Open a web browser (Chrome or Internet Explorer recommended) and navigate to http://169.254.10.227/
11. Click the download link on the page to download the DSP Controller installer.
12. Before installing, ensure Microsoft .NET Framework (latest version) is installed on the PC.
13. Run the installer and follow the on-screen prompts.

Note: *If the download does not start, disable any browser download pop-up blockers, or try an alternative browser.*

5.2 Connecting to a Device

14. Launch DSP Controller.

15. Click the Search button (top-right of the main window). The software will broadcast a discovery request and list all DSP devices found on the network.
16. Click the desired device in the list to connect. The indicator on the device tile will flash to confirm the connection.

Note: A single DSP device supports simultaneous connection from up to 4 users. Changes made by any user are reflected in real time on all other connected sessions.

Two additional controls are available on the main menu toolbar:

- Restore Default Scene — reloads the last saved preset configuration from the device.
- Factory Reset — restores the device to its factory defaults. All presets and configuration are cleared.

5.3 Module Editing

All DSP processing is configured through a drag-and-drop canvas in DSP Controller. Each channel strip displays its assigned processing modules in signal-flow order. There are two ways to access module parameter controls:

- Left-click a module to open its parameter interface directly.
- Right-click a module to open a context menu with options including parameter access, group setting, and gain limits.

Parameters take effect immediately when changed while connected online. Changes are stored to the current preset only when explicitly saved.

6. Processing Modules

6.1 Input Source

Each input channel provides the following configuration options:

Parameter	Description
Sensitivity (Preamp Gain)	Sets the microphone preamp gain in steps: 0 / 3 / 6 / 9 / 12 / 15 / 18 / 21 / 24 / 27 / 30 / 33 / 36 / 39 / 42 / 45 / 48 dBu. Select the lowest gain that provides an adequate signal level to maximise headroom and minimise noise.
Phantom Power	Enables +48 V DC phantom power on the selected input. Enable only for condenser microphones that require it. Do not enable for dynamic microphones, ribbon microphones, or line-level sources.
Sine Wave Generator	Generates a sine wave test tone at the selected frequency (20 Hz – 20 kHz). Level is adjustable in dBFS. Useful for system calibration and level alignment.
White Noise Generator	Generates broadband white noise with a flat frequency spectrum across the audible range. Level is adjustable. Frequency parameter has no effect in this mode.
Pink Noise Generator	Generates pink noise with energy distributed primarily in the mid and low frequency bands, decreasing at 3 dB/octave. Level is adjustable. Frequency parameter has no effect in this mode.

Note: *Right-click any channel fader to access Group Setting, and to set Minimum and Maximum Gain limits for that channel. Maximum gain limits prevent accidental system instability during operation.*

6.2 Expander

The expander extends the dynamic range of a signal by attenuating signals that fall below a set threshold. In contrast to a compressor (which attenuates signals above the threshold), the expander makes quiet signals quieter. At high expansion ratios, the expander behaves similarly to a noise gate.

Parameter	Description
Threshold	The level at which expansion begins. Signals below this level are attenuated. Typically set just above the ambient noise floor of the input source.
Ratio	The expansion ratio below the threshold. At 1:2, a signal 20 dB below the threshold is attenuated by a further 20 dB at the output (total 40 dB below threshold). Higher ratios approach noise gate behaviour.
Attack Time	The time required for the expander to engage when the signal exceeds the threshold (gate opens). Shorter attack times allow the expander to open more quickly.
Release Time	The time required for the gain to recover to the attenuated level after the signal drops below the threshold (gate closes).

6.3 Compressor & Limiter

6.3.1 Compressor

The compressor reduces the dynamic range of signals that exceed a set threshold. It is used to control transient peaks, even out inconsistent microphone levels, and add perceived loudness.

Parameter	Description
Threshold	The level above which compression begins. Any signal exceeding this threshold is attenuated in proportion to the compression ratio.
Ratio	The compression ratio (e.g., 2:1 means a 2 dB increase above the threshold produces only a 1 dB increase at the output). Adjustable from 1:1 (no compression) to 20:1 (near-limiting).
Attack Time	Controls how quickly the compressor engages when the signal exceeds the threshold. Slower attack times allow initial transients to pass through unaffected.
Release Time	Controls how quickly the gain recovers after the signal falls below the threshold. Faster release can cause 'pumping' artefacts.
Output Gain	Makeup gain applied after compression to compensate for overall level reduction. Applied uniformly to all signal levels regardless of compression activity.
G.R. Meter	Displays the current gain reduction in dB (shown as an inverse meter). For example, -12 dB on the G.R. meter indicates 12 dB of active compression.

6.3.2 Limiter

The limiter is a specialised compressor configured to ensure the signal does not exceed a maximum level under any circumstances. It employs a two-stage process: the first stage applies gentle attenuation, and the second stage applies aggressive gain reduction if the threshold is exceeded. Occasional transient clipping is addressed by the limiter; repeated clipping indicates the overall signal level should be reduced upstream.

Parameter	Description
Threshold	The absolute ceiling level. The output signal will not exceed this level.
Release Time	Controls how quickly the gain recovers after a limiting event.

6.4 Auto Gain Control (AGC)

AGC normalises signals of varying or unpredictable levels to a consistent target output level. It operates as a compressor with a low threshold, moderate-to-slow attack, long release, and low ratio. Includes silence detection to prevent gain hunting during quiet periods. Typical uses include background music normalisation and paging microphone level consistency.

Parameter	Description
Threshold	The level below which the AGC operates at a 1:1 ratio. Set just above the ambient noise floor of the input signal.
Ratio	The compression ratio applied to signals above the threshold.
Target Threshold	The desired output level. The AGC will attempt to bring signals that exceed the threshold to this target level.
Attack Time	Response time for signals exceeding the threshold.
Release Time	Response time for signals falling below the threshold.

6.5 Equalizer

The parametric equalizer allows precise shaping of the frequency response. Each segment can be independently configured as a parametric EQ, high/low shelf, or high/low pass filter. Available configurations include 5, 8, and 12-band parametric equalizers.

Parameter	Description
Type	Selects the filter type: Parametric EQ (default), High Shelf, Low Shelf, High Pass, or Low Pass. Different types produce different frequency response curves.
Frequency (Hz)	Center frequency of the filter. For pass-type filters, this is the cut-off frequency.
Gain (dB)	The amount of boost or cut applied at the center frequency.
Q	Quality factor controlling the bandwidth of the filter. Range: 0.02–50. Higher Q values create narrower, more surgical cuts or boosts. For shelf and pass filters, Q values above 0.707 introduce a resonant peak; values below 0.707 produce a gentler roll-off.
Segment Switch	Enables or disables each individual filter segment. The master switch enables or disables the entire equalizer module.

6.6 Graphic Equalizer

The graphic equalizer uses constant-Q technology with push-pull faders for each frequency band. The bandwidth of each filter remains fixed regardless of the amount of cut or boost applied. Available configurations: 10-band, 15-band, and 31-band (spanning 20 Hz to 20 kHz).

6.7 Feedback Inhibition

The feedback inhibitor automatically detects and suppresses acoustic feedback in the system by placing narrow-band notch filters at the feedback frequency. It should be used to supplement — not replace — good system design practices such as minimising the number of open microphones, positioning microphones away from loudspeakers, and using proper room equalization.

Note: *The feedback inhibitor cannot overcome fundamental system design limitations or compensate for excessive system gain beyond the room's physical capability.*

Parameter	Description
Panic Threshold	A level above which any signal is considered feedback. When exceeded: the output gain is temporarily attenuated, the output level is clamped, and filter sensitivity is increased for rapid detection. Returns to normal once the level drops below the threshold. Set to 0 to disable.
Feedback Threshold	A minimum level below which signals are not considered feedback. Prevents the module from activating on low-level noise or quiet programme material.
Filter Depth	Maximum attenuation (dB) applied by a single notch filter. Shallower depths reduce collateral damage to programme audio but may require more filters to fully suppress feedback.
Bandwidth	1/10 or 1/5 octave. Narrower bandwidth (1/10 Oct) is recommended for musical content; wider bandwidth (1/5 Oct) is recommended for speech applications or persistent feedback.
Preset	Four built-in presets: Large Music Room, Small Music Room, Large Speech Room, Small Speech Room. Each preset optimises the module parameters for the selected environment.
Auto / Manual / Fixed modes	Auto: the notch filter is placed and adjusted automatically. Manual: the filter gain may be manually overridden. Fixed: the filter is permanently active and persists through device reboots. Useful for known persistent feedback frequencies.
Clear	Clears all currently active notch filters. Use this before re-commissioning the feedback module in a modified system.

6.8 Noise Gate

Parameter	Description
Threshold	Signals above the threshold pass; signals below are attenuated. The noise gate is typically used to suppress ambient noise on microphone inputs when no speech is present.
Depth	The amount of attenuation (dB) applied to signals below the threshold.
Attack Time	The speed at which the gate opens when the signal exceeds the threshold.
Release Time	The speed at which the gate closes when the signal falls below the threshold.
Hold Time	The minimum time the gate remains open after the signal drops below the threshold. Prevents the gate from closing on brief pauses within speech.

6.9 Ducker

The ducker reduces the level of one channel (the 'ducked' channel) whenever the level of a reference channel exceeds a defined threshold. Typical use cases include reducing background music when a microphone is active, or dimming an announcement channel when another channel is in use.

Parameter	Description
Threshold	The reference channel level above which ducking begins; below which the ducked channel recovers.
Depth	The amount of attenuation applied to the ducked channel when the reference is active.
Attack Time	The time taken to begin attenuating the ducked channel after the reference exceeds the threshold.
Recovery Time	The time taken to restore the ducked channel after the reference drops below the threshold.
Hold Time	The minimum time ducking is maintained after the reference drops below the threshold.

6.10 SPL Compensator

The SPL compensator automatically adjusts output volume based on ambient noise level detected by a reference microphone. As background noise increases, output gain is raised; as noise decreases, gain is reduced. This maintains consistent perceived loudness in variable acoustic environments.

Parameter	Description
Maximum Gain	The maximum amount (dB) by which the output gain can be increased.
Minimum Gain	The minimum amount (dB) by which the output gain can be decreased.
Gain Sensing Ratio	The ratio of gain adjustment to sensed noise change.
Speed	The rate at which gain adjustments are made.
Trim	A fixed gain offset applied to the output.
Noise Threshold	Ambient noise above this level triggers gain increase; below this level triggers gain reduction.
Distance	The distance between the reference microphone and the local sound source. Used to calibrate the sensing ratio.

6.11 AutoMixer

The AutoMixer manages gain distribution across multiple open microphone channels. In a conference environment, having multiple microphones open simultaneously increases ambient noise pick-up, room reverberation, and feedback risk. The AutoMixer addresses this by attenuating unused microphone channels while maintaining the active channel at the appropriate level.

The DSP Controller includes a built-in gain-share AutoMixer supporting up to 32 channels, plus a threshold-based AutoMixer variant. Each channel has a direct output that is controlled by mute only — not by auto gain — making it suitable for recording feeds.

6.11.1 Main Control Parameters

Parameter	Description
Master Switch	Enables or disables the AutoMixer globally.
Gain	Controls the master output volume of the AutoMixer.
Slope	Controls how aggressively lower-level channels are attenuated relative to the active channel. Recommended value: 2.0. At 1.0, the AutoMixer has no effect. At 3.0 or above, gain adjustments may sound unnatural.
Response Time	The speed at which the AutoMixer adjusts gain. Range: 100 ms to several seconds. Faster values prevent the beginning of speech from being cut off; slower values provide smoother, more natural transitions.

6.11.2 Per-Channel Parameters

Parameter	Description
Auto Mix Switch	Enables or excludes the channel from AutoMix gain calculations. Channels such as background music should have Auto Mix disabled.
Mute	Mutes the channel output. Note: a muted channel's level still influences AutoMix gain calculations for other channels. To fully exclude a channel from AutoMix, disable its Auto Mix switch rather than muting it.
Gain	Adjusts the channel's level contribution within the AutoMix calculation.
Priority	A value from 0 (lowest) to 10 (highest), default 5. When two channels have the same signal level, the higher-priority channel receives more auto gain. A one-unit priority difference results in approximately 2 dB of additional gain at Slope 2.0.

6.11.3 Threshold AutoMixer (Alternative Mode)

A threshold-based AutoMixer variant uses noise gates to determine whether each channel is open or closed. Parameters specific to this mode:

Parameter	Description
Hold Time	How long a channel stays open after speech pauses, to avoid unnatural chopping of speech.
Off Gain	The gain applied to channels that are below the threshold (closed state).
Sensitivity	The ease with which a channel opens. Higher sensitivity allows channels to open at lower signal levels relative to the adaptive noise floor.
NOM Attenuation	Attenuation applied as more microphones are opened simultaneously, based on the Number of Open Microphones (NOM). At 3 dB, opening two microphones attenuates the output by 3 dB; four microphones by 6 dB; eight by 9 dB, and so on.
NOM Limit	The maximum number of microphones permitted to be open simultaneously.
Noise Threshold	Signals above this level are considered speech (channel opens); signals below are attenuated (channel closes).

6.12 Echo Canceller

Acoustic Echo Cancellation (AEC) is used in conferencing scenarios to prevent the far-end caller from hearing their own voice returned as an echo. The AEC module simulates the echo path in the local room, estimates the echo component of each microphone signal, and subtracts it — removing the echo before the signal is transmitted to the far end.

Parameter	Description
NLP (Non-Linear Processing)	Determines the residual echo suppression level after linear cancellation. Options: Conservative, Moderate, or Aggressive. More aggressive settings provide greater echo suppression but may affect speech naturalness.

Note: *The Echo Canceller module must be used in conjunction with the Matrix module to correctly configure the local and remote signal routing paths.*

6.13 Noise Suppression

The noise suppression module distinguishes human speech from non-speech noise and attenuates the non-speech components. The result is a signal containing predominantly speech with residual noise attenuated.

Parameter	Description
Suppression Level	The amount of non-speech noise attenuation. Options: 6 dB, 10 dB, 15 dB, 18 dB. Higher suppression values provide greater noise reduction but may introduce audible artefacts on speech.

6.14 Matrix Router

The matrix provides full-crosspoint routing and mixing between all input and output channels. The horizontal axis represents input channels; the vertical axis represents output channels. By default, the matrix is configured one-to-one (input 1 to output 1, input 2 to output 2, etc.).

To mix multiple inputs to a single output: enable multiple checkboxes in the same output row. For example, to mix inputs 1 and 2 to output 1, enable both input 1 and input 2 checkboxes in the output 1 row. Channels participating in AutoMix, Echo Cancellation, or Noise Suppression must also be routed correctly in the matrix for those modules to function properly.

6.15 High & Low Pass Filters

Each output channel includes dedicated high-pass and low-pass filter modules. These are independent of the equalizer module.

Parameter	Description
Frequency	The cut-off frequency. For Bessel and Butterworth types, the cut-off is defined at –3 dB; for Linkwitz-Riley, at –6 dB.
Gain	Broadband gain adjustment applied to the entire channel (not frequency-selective).
Type	Filter topology: Bessel (gentle phase response), Butterworth (flattest passband), or Linkwitz-Riley (ideal for crossover alignment).
Slope	Attenuation rate in the transition band: 6, 12, 18, 24, 30, 36, 42, or 48 dB/octave. For example, 24 dB/Oct indicates 24 dB of attenuation per octave beyond the cut-off frequency.

6.16 Delayer

The delayer inserts a fixed delay into the audio signal path for time-alignment purposes.

Parameter	Description
Enable	Activates or bypasses the delay module.
Delay Time	Sets the delay in milliseconds. Range: 1 – 1200 ms. Alternate display units (metres or feet) are also available, based on the speed of sound, for distance-based alignment.

6.17 Output

Parameter	Description
Invert	Applies a 180° phase inversion to the output signal. Use to correct polarity mismatches in the signal chain.
Mute	Mutes or unmutes the output channel. Right-click the output channel strip for additional options including group assignment and step-size configuration.

6.18 USB Soundcard

The front-panel USB port provides a built-in USB audio interface (1-in / 1-out). Connect a PC to the device using a USB Type-A to Type-A cable. On first connection, Windows will automatically install the USB audio driver. The device will appear in the PC's audio device list as a USB soundcard.

DSP Controller's USB Soundcard interface provides playlist management, recording controls, and soundcard routing. Users can build and save playlists for background music playback, and record audio from the DSP directly to the PC.

Parameter	Description
Play / Pause / Stop	Playback controls for the active playlist.
Previous / Next Track	Navigate between tracks in the playlist.
Song Volume	Playback volume for the active track.
Start / Stop Recording	Begin or end audio recording to the PC.
Recording Volume	Input level for the recording path.

7. Software Settings

7.1 File Menu

In offline mode (no device connected), use the File menu to open an existing configuration file (*.uma extension) or right-click a *.uma file and open it directly with DSP Controller. Use Save As to export the current preset configuration to local storage for backup or deployment to other devices.

7.2 Device Setting

Parameter	Description
Device Name	Assign a descriptive name to the device (maximum 16 characters or 5 Chinese characters). The name is displayed in the device discovery list.
Network Address	Configure the device's IP address, subnet mask, and gateway.
Serial Baud Rate (SERBAUD)	Set the baud rate for RS-232/RS-485 communication. Default: 115200.
Default Startup Preset	Select whether the device boots with a fixed preset (any of the 16 presets) or restores the last preset active before the previous power cycle.

7.3 GPIO Setting

The device provides 8 GPIO ports, each independently configurable as an input or output. Open the GPIO setting interface via the Setting menu.

7.3.1 Input GPIO Functions

Parameter	Description
Preset	Recalls a specified preset when the selected trigger condition is met. Trigger types: high-level, low-level, rising-edge, falling-edge (and combinations with cancellation).
Routing	Opens or closes a specified matrix routing path (input-to-output mix) on trigger.
Gain	Increases or decreases the gain of a specified input or output channel by a defined step size (dB) on each trigger.
Mute / Unmute	Mutes or unmutes a specified input or output channel on trigger.
Command	Sends a user-defined RS-232 command string on trigger.
Analog-to-Digital Gain	Connects an external potentiometer to continuously adjust input or output channel gain. The potentiometer's voltage position is translated to a gain value.

7.3.2 Output GPIO Functions

Parameter	Description
Preset	Outputs a high or low logic level when the device changes to the specified preset.
Level	Outputs a high or low logic level when a specified channel's signal level crosses a threshold.
Mute	Outputs a high or low logic level that tracks the mute state of a specified channel.

7.4 Group Setting

The Group Setting interface allows input and output channels to be organised into groups. Channels within the same group have their volume and mute controls synchronised — adjusting one channel in the group adjusts all others by the same amount simultaneously.

Each channel may belong to only one group. Groups take priority over the LINK function: channels assigned to a group will not participate in LINK. The key difference between Group and LINK is that groups synchronise only gain and mute, whereas LINK synchronises all module parameters of the linked channels.

7.5 Panel Setting

Physical control panels (OLED display panel and key panel) connect to the DSP device via cable. The Panel Setting interface allows panel parameters to be configured offline and then downloaded to the panel hardware.

7.5.1 OLED Panel

The OLED panel combines a 1.3-inch OLED display with a rotary knob. The display organises controls into a menu hierarchy. Menus, buttons, and preset selections can be assigned to the OLED interface. Operating the panel:

17. The main interface displays the device name and IP address. Rotate the knob to scroll between menus.
18. Press the knob to enter edit mode (the selected item begins flashing).
19. Rotate the knob to change the value.
20. Press the knob again to confirm and return to menu mode.

7.5.2 Key Panel

The key panel provides 8 programmable keys and one rotary knob. Drag functions from the function library in DSP Controller to the desired key to program it. Available functions: Volume Adjustment, Mute, Preset Recall, and Command Send. The rotary knob controls the gain of the assigned channel.

Key indicator behaviour:

- Steady on: Key is configured for a Mute function.
- Slow flashing: Key is configured for a Gain function. The ring of 13 indicators around the knob shows the current gain level.
- Single flash on press: Key is configured for a Preset or Command function.

7.6 Help Menu

7.6.1 Central Control Command Assistant

Opens a command-builder window. Click any parameter on the software interface to display the corresponding ASCII control command. Copy the command for use with UDP or RS-232 central control systems.

7.6.2 Device Upgrade

Initiates a firmware upgrade over UDP. Connect to the device, then navigate to Setting > Help > Device Upgrade. Select the upgrade file (*.bin) from the file browser and follow the prompts.

7.6.3 About

Displays the software version, technical support contact information, and copyright notice.

8. External Control

8.1 Overview

The DSP supports external control via UDP (Ethernet) and RS-232 serial. The ASCII command protocol provides complete control of all device parameters including gain, mute, phantom power, routing, scene recall, and more.

UDP Control Port	50000 (default; configurable in Device Setting)
RS-232 Baud Rate	115200 (default; configurable in Device Setting)
RS-232 Format	8 data bits, 1 stop bit, no parity (8N1)
RS-232 Message Interval	Minimum 100 ms between consecutive RS-232 commands
Central Control Reply	Disabled by default. Enable in Device Setting to receive acknowledgement responses.

8.2 Command Format

Set command: set:<module>#<item>#<start_ch>-<end_ch>#<value>

Get command: get:<module>#<item>#<start_ch>-<end_ch>

Note: *Channel numbering is zero-indexed. Channel 0 corresponds to channel 1 as displayed in DSP Controller. In the examples below, channels 0–3 represent DSP Controller channels 1–4. Use the actual channel range for your device model.*

8.3 Command Reference

Function	Command	Example / Notes
Input Gain (set)	set:input#gain#0-3#1	Sets input channels 1–4 to +1 dB
Input Gain (get)	get:input#gain#0-3	Returns: get:input#gain#0-3#1#1#1#1
Output Gain (set)	set:output#gain#0-3#1	Sets output channels 1–4 to +1 dB
Output Gain (get)	get:output#gain#0-3	Returns: get:output#gain#0-3#1#1#1#1
Phantom Power (set)	set:input#phant#0-3#1	1 = on, 0 = off
Phantom Power (get)	get:input#phant#0-3	Returns per-channel state
Input Mute (set)	set:input#mute#0-3#1	1 = mute on, 0 = mute off
Input Mute (get)	get:input#mute#0-3	Returns per-channel mute state
Output Mute (set)	set:output#mute#0-3#1	1 = mute on, 0 = mute off
Output Mute (get)	get:output#mute#0-3	Returns per-channel mute state
Sensitivity (set)	set:input#sens#0-3#1	Sets gain step (1 = second step = 3 dB)
Sensitivity (get)	get:input#sens#0-3	Returns per-channel sensitivity step
Matrix Route (set)	set:mixer#switch#0#0-3#1	Routes input 1 to outputs 1–4
Matrix Route (get)	get:mixer#switch#0-3#0	Returns route states for inputs 1–4 to output 1
Matrix Gain (set)	set:mixer#gain#0-3#0#1	Sets inputs 1–4 to output 1 route gain to 1 dB
Scene Recall	scene:toggle#3	Recalls preset 4 (zero-indexed; displays as scene 4)
Scene Save	scene:save#3	Saves current state to preset 4
Scene Name (set)	set:scene#name#0-3#pre1	Names presets 1–4 'pre1'
Scene Name (get)	get:scene#name#0-3	Returns names for presets 1–4
Input Level (get)	get:input#level#0-3	Returns dBFS level for input channels 1–4
Output Level (get)	get:output#level#0-3	Returns dBFS level for output channels 1–4
System Mute (set)	set:sysctl#mute#1	1 = system mute on, 0 = off
System Mute (get)	get:sysctl#mute	Returns: get:sysctl#mute#1
Phase Invert Input	set:input#phase#0-3#1	1 = inverted, 0 = normal
Phase Invert Output	set:output#phase#0-3#1	1 = inverted, 0 = normal
Channel Name (set)	set:input#name#0#MyMic	Names input channel 1 'MyMic'
Channel Name (get)	get:input#name#0-3	Returns: get:input#name#0-3#IN1#IN2#IN3#IN4
Input Step (set)	set:input#step#0-3#10	Sets step size for input channels 1–4 to 10
Link Channels	set:input#link#0-3#1	Links input channels 1–4
Signal Gen Type	set:input#type#0-3#1	Sets signal generator type for input channels 1–4
Factory Reset	set:refactory	Restores factory defaults
Scene Reset	set:rescene	Resets all scene data

9. Frequently Asked Questions

9.1 How do I restore factory settings?

There are two methods:

21. In DSP Controller: Click the Factory Reset button on the main toolbar. This clears all presets and configuration data.
22. Via RS-232 serial terminal (e.g., SecureCRT): Connect to the device serial port at 115200 baud, 8N1. Long-press Enter at the terminal interface during boot to enter the bootloader. Available bootloader commands:

- del config — Deletes network configuration. The device returns to its default IP address (169.254.10.227).
- del scenes — Deletes all 16 presets. All presets are restored to factory defaults.
- del all — Deletes all configuration and presets (firmware is preserved).

Note: *If SecureCRT shows no local echo, enable it via Options > Session Options > Local Echo.*

9.2 The software cannot find the device on the network. What should I check?

23. Confirm the PC is connected to the same network segment as the device. Default device address: 169.254.10.227 / 255.255.0.0.
24. Temporarily disable any firewall or antivirus software that may be blocking UDP broadcast traffic on the PC.
25. Verify the Ethernet cable is securely connected to both the device and the PC or switch.
26. Check that the device POWER LED is illuminated and the STATUS LED is active.
27. Try entering the device IP address directly into DSP Controller's connection field rather than using auto-discovery.

9.3 I cannot download the DSP Controller software from the device.

Try the following:

- Use Internet Explorer or Google Chrome.
- Disable any browser pop-up blocker or download prompt settings that may prevent the file from downloading.
- Verify that Microsoft .NET Framework (latest version) is installed on the PC before running the installer.

9.4 Phantom power was accidentally enabled on a dynamic microphone. What should I do?

Immediately disable phantom power for that channel in DSP Controller. Disconnect the microphone and inspect the connector for signs of damage. Many dynamic microphones will withstand brief exposure to phantom power without damage, but ribbon microphones in particular may be permanently damaged. Consult the microphone manufacturer's documentation if in doubt.

9.5 The device is producing feedback even with the Feedback Inhibition module active.

The Feedback Inhibition module has a finite number of notch filters (8 per channel). If all filters have been consumed and feedback continues, consider:

- Using the Clear button to reset filters and recommissioning the feedback module from scratch.
- Reviewing the system gain structure — if the system gain significantly exceeds what the room can support acoustically, no number of filters will fully suppress feedback.
- Reviewing loudspeaker and microphone placement to minimise direct coupling between them.
- Adding a Compressor/Limiter module downstream of the feedback-prone microphone channel for additional protection.

10. Appendix A: Module ID Reference

The following table lists the module IDs used in external control commands.

Module Name	ID	Module Name	ID
Input Source	299	Output Ch 1–32 High/Low Pass	167–198
Input Ch 1–32 Expander	1–32	Output Ch 1–32 Equalizer	199–230
Input Ch 1–32 Compressor	33–64	Output Ch 1–32 Delayer	231–262
Input Ch 1–32 Auto Gain	65–95	Output Ch 1–32 Limiter	263–294
Input Ch 1–32 Equalizer	97–128	AutoMixer	161
Input Ch 1–32 Feedback Inhibition	129–160	Echo Canceller	162
Echo Cancellation (module)	163	Noise Suppressor	164
Noise Suppression (module)	165	Mixer (Matrix)	166
Output	295	System Control	296

11. Appendix B: Module Parameter Type Reference

The following tables list the parameter type codes used when constructing external control commands for individual processing modules.

Parameter	Description
Input Source — 0x1	Gain
Input Source — 0x2	Mute

Parameter	Description
Input Source — 0x3	Sensitivity (preamp gain step)
Input Source — 0x4	Phantom Power Switch
Input Source — 0x5	Signal Generator Type
Input Source — 0x6	Signal Generator Frequency
Input Source — 0x7	Sine Wave Gain
Input Source — 0x8	Channel Name
Input Source — 0x9	Phase Invert
Input Source — 0x10	Gain Compensation
Input Source — 0x11	Link
Input Source — 0x12	Channel Level
Expander — 0x1	Switch
Expander — 0x2	Threshold
Expander — 0x3	Ratio
Expander — 0x4	Attack Time
Expander — 0x5	Release Time
Compressor — 0x1	Compressor Switch
Compressor — 0x2	Threshold
Compressor — 0x3	Ratio
Compressor — 0x4	Attack Time
Compressor — 0x5	Recovery Time
Compressor — 0x6	Gain Compensation
Equalizer — 0x1	Master Equalizer Switch
Equalizer — 0x2	Individual Band Switch
Equalizer — 0x3	Frequency
Equalizer — 0x4	Gain
Equalizer — 0x5	Q Value
Equalizer — 0x6	Filter Type
Delayer — 0x1	Bypass Switch
Delayer — 0x2	Delay (milliseconds)
Delayer — 0x3	Delay (microseconds)
Mixer — 0x1	Mixer Switch (route on/off)
Mixer — 0x2	Mixer Gain
High/Low Pass — 0x1 / 0x11	High Pass / Low Pass Switch
High/Low Pass — 0x2 / 0x12	High Pass / Low Pass Filter Type
High/Low Pass — 0x3 / 0x13	High Pass / Low Pass Slope
High/Low Pass — 0x4 / 0x14	High Pass / Low Pass Frequency
High/Low Pass — 0x5 / 0x15	High Pass / Low Pass Gain
Feedback Inhibition — 0x1	Switch
Feedback Inhibition — 0x2	Feedback Point Frequency
Feedback Inhibition — 0x3	Feedback Point Gain
Feedback Inhibition — 0x6	Preset
Feedback Inhibition — 0x7	Clear
Feedback Inhibition — 0x8	Panic Threshold
Feedback Inhibition — 0x9	Feedback Threshold
Auto Gain — 0x1	Switch
Auto Gain — 0x2	Threshold
Auto Gain — 0x3	Target Threshold
Auto Gain — 0x4	Ratio

Parameter	Description
Auto Gain — 0x5	Attack Time
Auto Gain — 0x6	Release Time
Auto Mix — 0x1	Master Mute
Auto Mix — 0x2	Master Gain
Auto Mix — 0x3	Slope
Auto Mix — 0x4	Response Time
Auto Mix — 0x5	Channel Auto Mix Switch
Auto Mix — 0x6	Channel Mute
Auto Mix — 0x7	Channel Gain
Auto Mix — 0x8	Channel Priority
Auto Mix — 0x9	Auto Mix Switch
Echo Cancellation — 0x1	Echo Cancellation Switch
Echo Cancellation — 0x2	Echo Cancellation Mode (NLP)
Noise Suppression — 0x1	Noise Suppression Switch
Noise Suppression — 0x2	Noise Suppression Level
Output — 0x1	Gain
Output — 0x2	Mute
Output — 0x3	Channel Name
Output — 0x4	Phase Invert
Output — 0x5	Sensitivity
Output — 0x6	Gain Compensation
Output — 0x7	Link
Output — 0x8	Channel Level
System Control — 0x1	System Mute
System Control — 0x2	System Gain