DTX Series

DTX-44D ● DTX-88D ● DTX-1616D ● DTX-3232D ● DTX-6464D

Dante Based Audio DSP with Analog I/O

TABLE OF CONTENTS

Technology Overview

Introduction to Dash

The Dash DSP Controller is a Windows-based, full-featured application that configures and controls the DSP hardware. Systems can be designed to perfectly match specifications and conform as requirements change or evolve. All classes of audio processing modules for signal routing and shaping are provided.

Dash provides a revolutionary way of thinking about DSP software. This is a software designed by audio professionals for audio professionals. It uses an interface layout that will be very intuitive to even the most novice integrator yet extremely powerful in its capabilities.

Quick Start

1. Install Dash software.
2. Connect DTX DSP to the same network as the computer running Dash.
3. Use the PoE connection on a DTX-44D, use the LAN connection on all other DTX devices.
4. Open Dash.
5. Click the plus (+) sign in the Device tab row.
6. Select a template that matches the device you are connecting.
7. Click the 'Device List' in the top right corner of Dash.
8. Select the desired device from the list of discovered devices and click 'Set IP'.
9. Change the IP from the default 169.254.10.227 to a valid, unused static address on the active subnet.
10. Click 'Connect'.
11. The green 'LED' next to the device name on the Device tab should now be green.
12. You are all set!

Hardware

Safety Instructions

Please review the following safety precautions. If this is the first time using this model, then read this manual before installing or using the product. If the product is not functioning properly, please contact your local dealer or Aurora for further instructions.

The lightning symbol in the triangle is used to alert you to the presence of dangerous voltage inside the product that may be sufficient to constitute a risk of electric shock to anyone opening the case. It is also used to indicate improper installation or handling of the product that could damage the electrical system in the product or in other equipment attached to the product.

The exclamation point in the triangle is used to alert you to important operating and maintenance instructions. Failure to follow these instructions could result in injury to you or damage to the product.

Be careful with electricity:

- **Power Outlet:** To prevent electric shock, be sure the electrical plug used on the product power cord matches the electrical outlet used to supply power to the Aurora product. Use the power adapter and power connection cables designed for this unit.
- **Power Cord:** Be sure the power cord is routed so that it will not be stepped on or pinched by heavy items.
- **Lightning:** For protection from lightning or when the product is left unattended for a long period, disconnect it from the power source.

Also follow these precautions:

- **Ventilation:** Do not block ventilation slots, if applicable, on the product, or place any heavy object on top of it. Blocking airflow could cause damage. Arrange components so that air can flow freely. Ensure that there is adequate ventilation if the product is placed in a stand or cabinet. Put the product in a properly ventilated area, away from direct sunlight or any source of heat.
- **Overheating:** Avoid stacking the Aurora product on top of a hot component, such as a power amplifier.
- **Risk of Fire:** Do not place unit on top of any easily combustible material, such as carpet or fabric.
- **Proper Connections:** Be sure all cables and equipment are connected to the unit as described in this manual.
- **Object Entry:** To avoid electric shock, never stick anything in the slots on the case, or remove the cover.
- **Water Exposure:** To reduce the risk of fire or electric shock, do not expose to rain or moisture.
- **Cleaning:** Do not use liquid or aerosol cleaners to clean this unit. Always unplug the power to the device before cleaning.
- **ESD:** Handle this unit with proper ESC care. Failure to do so can result in failure.

FCC

This device complies with Part 15 of the FCC Rules. Operation is subject to the following two (2) conditions:

1. This device may not cause harmful interference.

2. This device must accept any interference received, including interference that may cause undesired operation.

Trademarks

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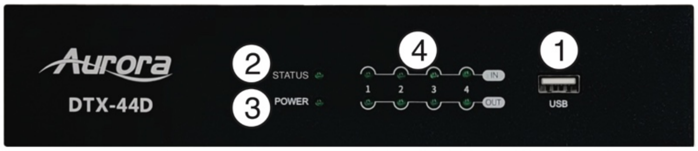
Front Panel Features



DTX-44D

DTX-88D, DTX-1616D, DTX-3232, DTX-6464D

① **USB (Audio)** – USB 2.0 Type A connector. Used in conjunction with the USB Sound card feature. Requires a USB Type-A to USB Type-A cable, included in box. The USB connection can also be used as a sound card with a PC or Mac using a generic HID audio. The driver will appear as a “Crestone USB Soundcard” in your computers sound settings.

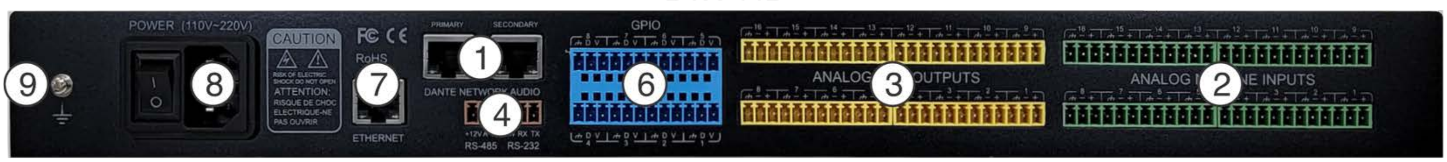
② **Status LED** – A green LED indicates network connectivity on the LAN connection.

③ **Power LED** – A green LED indicates power to the unit is good.

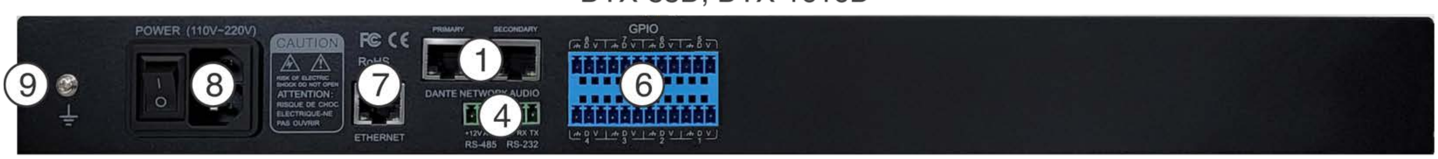
④ **I/O LEDs** – A green LED indicates analog connectivity.

Rear Panel Features

DTX-44D



DTX-88D, DTX-1616D



DTX-3232D, DTX-6464D

① **Dante Ethernet** – Primary and secondary Ethernet connections for Dante audio. Secondary connection can be used for to provide redundancy for the Dante audio network or can be used to daisy chain additional devices. PoE is not passed on.

DTX-44D – 10/100Mbps, PoE PD (15W) on top connector

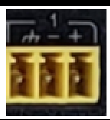
All Others – 10/100/1000Mbps, no PoE

NOTE: This port has different MAC and IP addresses than the LAN Control Interface (not applicable on DTX-44D).

② **Analog Inputs** – Balanced or unbalanced high impedance inputs support Line or Mic level with +48VDC phantom power. Channel count varies per Model. Mating connectors provided in box.

	Signal Label	XLR	TRS
	+	Pin 2	Tip
	-*	Pin 3	Ring
	⏚	Pin 1	Sleeve
	*do not use (-) for unbalanced		

③ **Analog Outputs** – Balanced or unbalanced 600Ω outputs. Channel count varies per Model. Mating connectors provided in box.

	Signal Label	XLR	TRS
	+	Pin 2	Tip
	-*	Pin 3	Ring
	⏚	Pin 1	Sleeve
	*do not use (-) for unbalanced		

④ **Serial Communication** – RS-232 and RS-485 connections. A +12VDC connection can be used with along with the serial connections to power a remote serial device such as a RS-485 control panel. The Ground connection is shared between the RS-232 and RS-485. Mating connectors provided in box.

	Signal Label	Signal
	+12V	Aux Power
	A	RS-485 A(-)
	B	RS-485 B(+)
	⏚	GND
	RX	RS-232 Receive
	TX	RS-232 Transmit

⑤ **DC Power Input** – 12VDC, 10W, center positive power Input. Provides alternate power source to PoE. Can be used simultaneously with PoE for redundant power.

⑥ **GPIO** – General Purpose I/O. The function of the GPIO pins can be configured in the GPIO Settings section of the Dash Software. Mating connectors provided in box.

	Signal Label	Signal
	V	+3.3VDC @ 500mA MAX
	D	GPIO Common
	⏚	GND

Below are general definitions of the GPIO depending on how they are defined in Dash:

GPIO Input: The port value is considered 'High' if the voltage applied to the 'D' pin is greater than 2VDC. The port value is considered 'Low' if the voltage applied to the pin is less than 0.3VDC. Input pins can support up to 12VDC.

GPIO Output: When the port is triggered 'High', the 'D' pin will output 5VDC with a drive current of 100mA; when triggered 'Low' the pin will equal ground. ***The maximum total current if all 8 pins are configured as outputs is 500mA.***

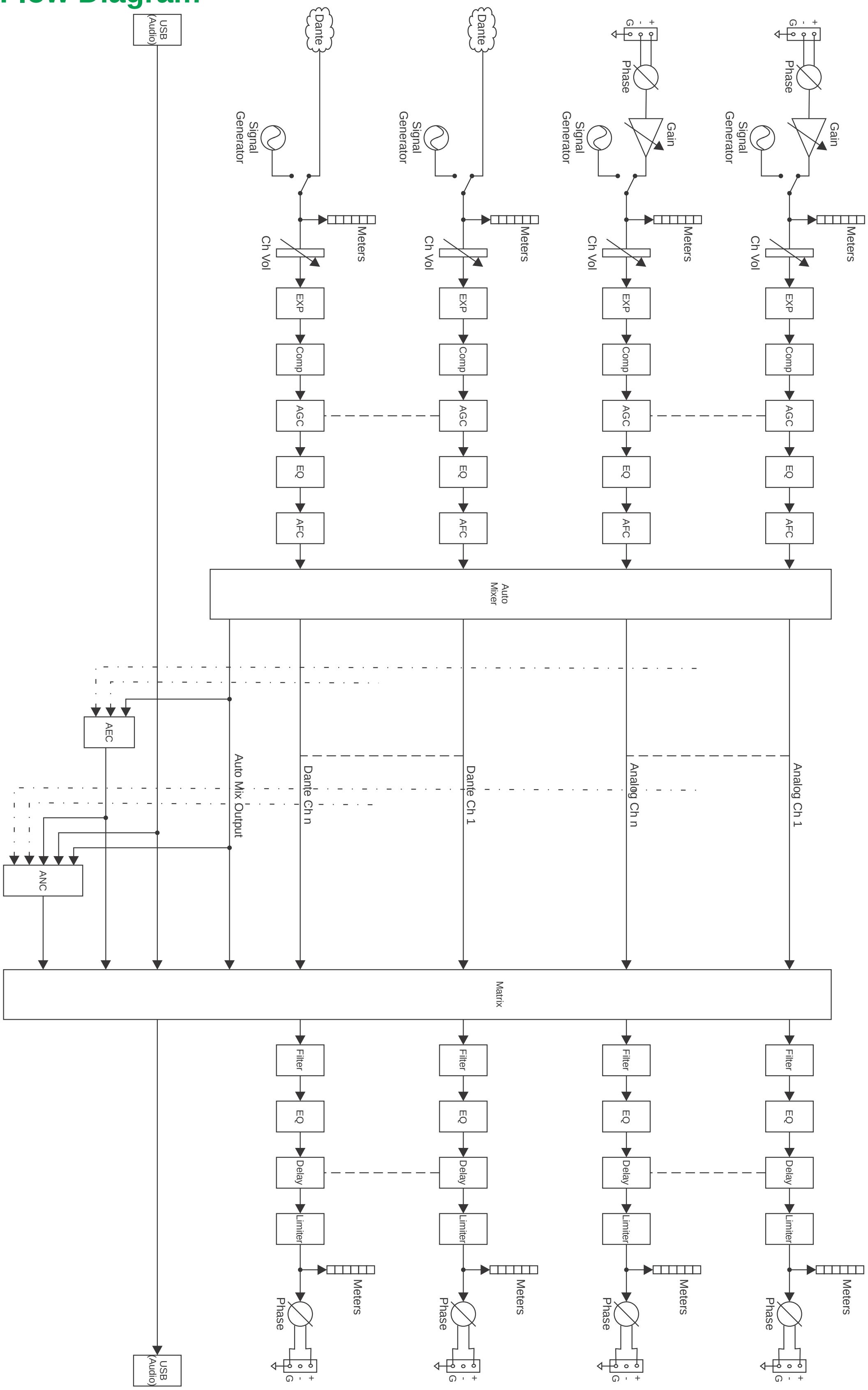
GPIO ADC Input: When the port is configured as an ADC input, an external potentiometer can be connected as a volume control.

⑦ **LAN Control Interface** – 10/100/1000Mbps Ethernet control interface. Use this interface to connect to Dash control software or to use UDP commands from a third-party source.
NOTE: This port has different MAC and IP addresses than the Dante Audio ports.

⑧ **AC Power Input/Switch** – 110-240VAC 50/60Hz, 40W Max power inlet via IEC 320 C14 socket. Main power switch for unit must be in the ON (I) position for normal operation.

⑨ **Ground Lug** – Ground lug to provide chassis ground. Secure to earth ground according to code.

Audio Signal Flow Diagram



Software

Software Installation

A Windows PC with a processor of 1 GHz or higher and:

- Windows 7 or higher version.
- 1 GB free storage space.
- 1024 x 768 resolution.
- 24 bit or higher color.
- 2GB or higher memory.
- Network (Ethernet) port.

CAT5 cable or current Ethernet network

Visit web page address https://support.auroramultimedia.com/dash/

Download software

1. Install files.

After the software is installed, use one of the following methods to enable the software:

- Desktop icons
- Start menu

When starting the software for the first time, it may take some time (1-15s) to start. Please be patient.

Software Operation

After starting the software, the main screen will look like the image below:

File Menu

The File menu is for opening and saving presets to the computer during offline configuration. Presets are stored on the DTX devices by default. In the case where you are configuring a system offline, that configuration can be saved to a local file on the controlling computer. This is important because when a live device is connected to the Dash software, the current active settings of the DTX device will be uploaded to the Dash software, including all stored presets. After the device is active, a saved preset configuration can be loaded from the host computer and pushed to the device.

Here are the steps to load an offline configuration file when NOT connected to an active device.

- Click 'File | Open'
- Select the file from the local storage and click 'OK'

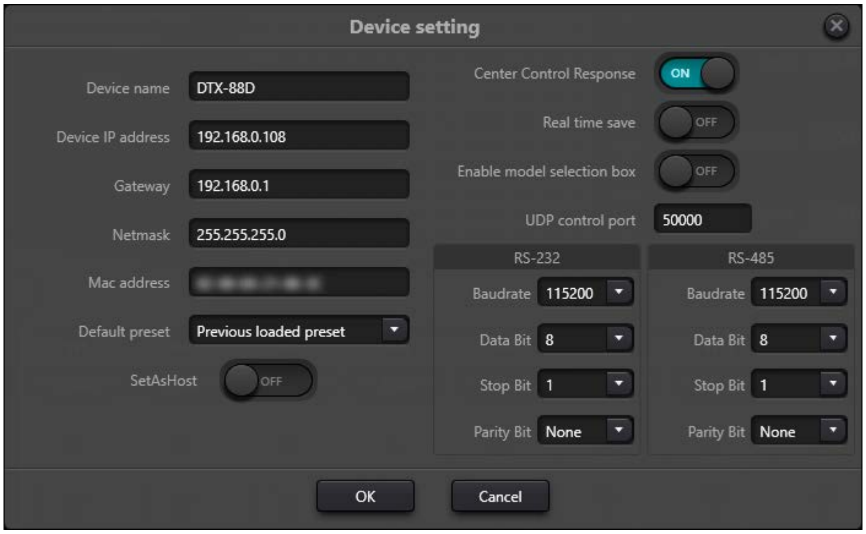
When the active Device tab is connected to a DTX device, the steps to open a saved file are different.

- Click the 'Load Preset' button next to the preset list 
- Select the preset from the list into which you wish to load the saved configuration file.
- Select the file from the local storage and click 'OK'
- The configuration will be uploaded and saved to the device in the selected preset. The preset will NOT be made active automatically.

Settings Menus

Device Setting

NOTE: This menu selection is only active when a DTX device is connected to Dash.



This dialog window provides the basic configuration information about the connected device.

Device Name: Change the name of the device by directly typing in the field. This device name will show up in Dash anywhere the device is addressed. This name is saved to the hardware.

Device IP Address: The IP address of the LAN interface. NOTE this is a different IP address than the Dante interface and is static. Since many control systems use hardcoded IP addresses to send commands to devices, DHCP is not supported for this interface. Be sure to use static addressing on this network or use DHCP reservations to ensure the address is not duplicated on another device on the network.

Gateway: The default gateway for the network.

Netmask: The network mask associated with the desired subnet.

Mac Address: The MAC address of the LAN interface. NOTE this is a different MAC address than the Dante interface.

Default Preset: Determines the default preset that should be made active on the device after a power cycle event. It can be either one of the 16 presets or can be set to load the last active preset before power cycle.

Set As Host: Enables device mirroring. See Appendix A: Device Mirroring for more information.

Backup Device ID: If 'Set As Host' is set to 'On' this sets the IP address of the backup device on the network.

Center Control Response: When set to on, enables control responses on the DTX device. Set this to 'ON' when using an external controller that expects a response to control messages.

Real Time Save: Saves any device changes to the active preset in real time. Setting this to 'On' can ensure that the exact state of the device is restored in the event of a power cycle if 'Previous loaded preset' is selected in the 'Default Preset' parameter. If set to 'Off' and changes are made to the device that are not saved to a preset, those changes will be lost after a power cycle event.

Enable Model Selection Box: If set to 'On', the model selection box will appear as soon the Dash software is loaded. This is set to 'Off' by default.

UDP Control Port: Sets the UDP control port for remote communication with the device.

UART Settings: ***Baud Rate, Data Bits, Stop Bit, and Parity Bit*** can be set for both the RS-232 and RS-485 ports of the device.

OK: Setting are saved to the device and the dialog box is closed.

Cancel: All changes are lost, and the dialog box is closed.

GPIO Setting

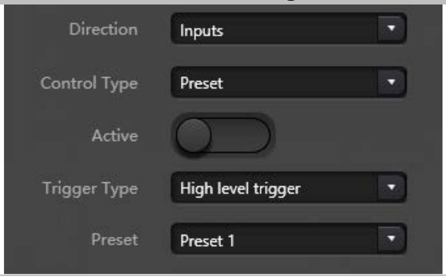
All DTX units above 4x4 channels include 8 GPIO interfaces. The "GPIO Setting" dialog allows for the configuration of this I/O.

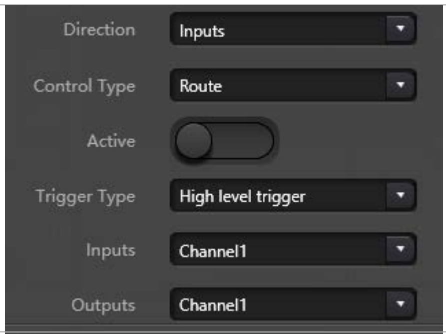
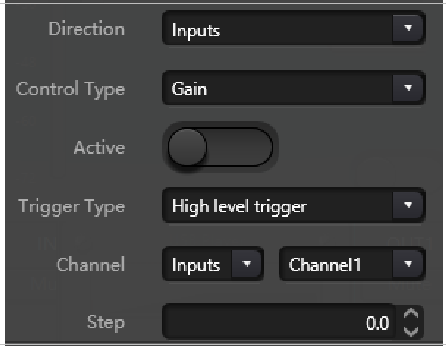
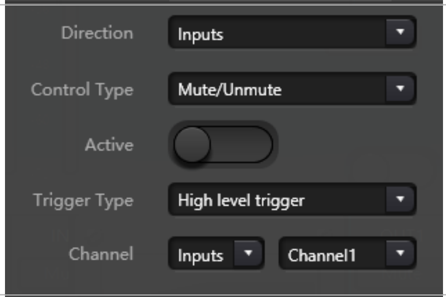
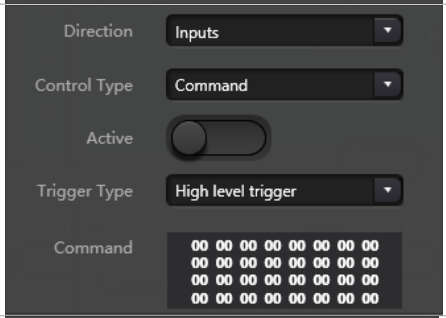
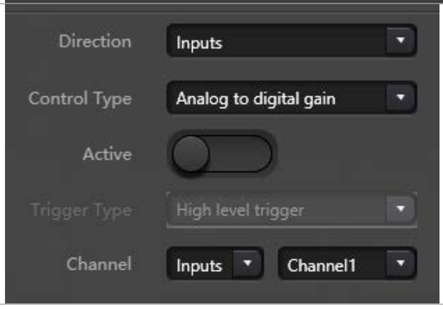
Channel Buttons: 8 buttons across the top of the dialog box selects which GPIO channel is being configured.

Direction: Determines whether the selected channel is an input or an output. Inputs and Outputs will have different value selections for Control Type.

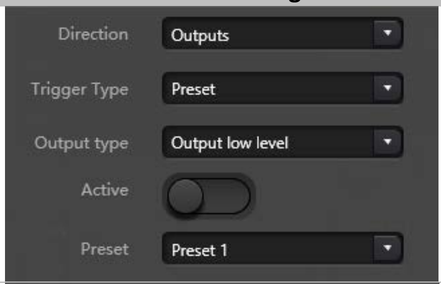
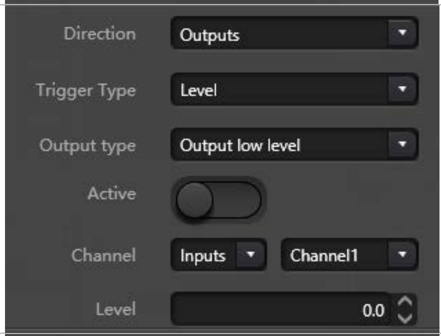
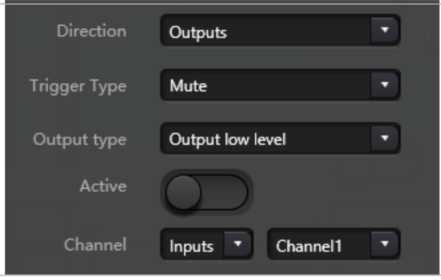
Active: Enables or disables the selected GPIO channel.

Control Type (inputs only): Determines the function which will control or be controlled by the GPIO. Additional parameters will differ depending on the Control Type selected. The following tables define the additional parameters based on the value of this field.

Direction = 'Input'		
Control Type Value	**Associated Dialog Box**	**Additional Parameters**
Preset: Changes the preset to a defined value when a trigger event occurs.		**Trigger Type:** 'High level trigger' will trigger the channel when the voltage on the pin is above the threshold. 'Low level trigger' will trigger the channel with the voltage on the pin is below the threshold. **Preset:** Selects which preset will be applied upon a trigger event.

Routing: Creates a matrix route point when a trigger event occurs.		**Trigger Type:** 'High level trigger' will trigger the channel when the voltage on the pin is above the threshold. 'Low level trigger' will trigger the channel with the voltage on the pin is below the threshold. **Input & Output:** Selects the two points that will be routed together upon a trigger event.
Gain: Changes the gain value on the predetermined channel when a trigger event occurs.		**Trigger Type:** 'High level trigger' will trigger the channel when the voltage on the pin is above the threshold. 'Low level trigger' will trigger the channel with the voltage on the pin is below the threshold. **Channel:** Selects which 'Input', 'Output' or 'System' channel will be affected by the gain change. **Step:** Defines the size of the gain change in dB of the gain change.
Mute/Unmute: Will mute a predetermined channel when the trigger is active and will release the mute when it becomes inactive.		**Trigger Type:** 'High level trigger' will trigger the channel when the voltage on the pin is above the threshold. 'Low level trigger' will trigger the channel with the voltage on the pin is below the threshold. **Channel:** Selects which 'Input', 'Output' or 'System' channel will be affected by the mute/unmute.
Command: Sends a hex command string via RS-232 when a trigger event occurs.		**Trigger Type:** 'High level trigger' will trigger the channel when the voltage on the pin is above the threshold. 'Low level trigger' will trigger the channel with the voltage on the pin is below the threshold. **Command:** String of Hex characters that will be sent over RS-232 upon a trigger event.
Analog-to-digital Gain: When a potentiometer is connected to the GPIO, the selected channel gain will be adjusted.		**Channel:** Selects which 'Input', 'Output' or 'System' channel will be affected by the gain change.

Trigger Type (outputs only): Determines which function will initiate a trigger event on the GPIO channel. Additional parameters will differ depending on the Trigger Type selected. The following tables define the additional parameters based on the value of this field.

Direction = 'Output'		
Trigger Type Value	**Associated Dialog Box**	**Additional Parameters**
Preset: Initiates a trigger event when a predefined preset is selected.		**Output Type:** 'Output low level' will set the channel pin to ground. 'Output high level' will set the channel pin to +5VDC. **Preset:** Defines which preset, when selected, will initiate a trigger event.
Level: Initiates a trigger event when a channel level reaches a defined threshold.		**Output Type:** 'Output low level' will set the channel pin to ground. 'Output high level' will set the channel pin to +5VDC. **Channels:** Identifies which channel will be monitored. **Level:** Sets the gain threshold, which when met, will initiate a trigger event.
Mute: Initiates a trigger event which mirrors the mute state of the predefined channel.		**Output Type:** 'Output low level' will set the channel pin to ground. 'Output high level' will set the channel pin to +5VDC. **Channels:** Identifies which channel will be monitored. When a channel is muted it will set the channel GPIO pin to the state defined in 'Output Type'. When unmuted it will revert back to the default state.

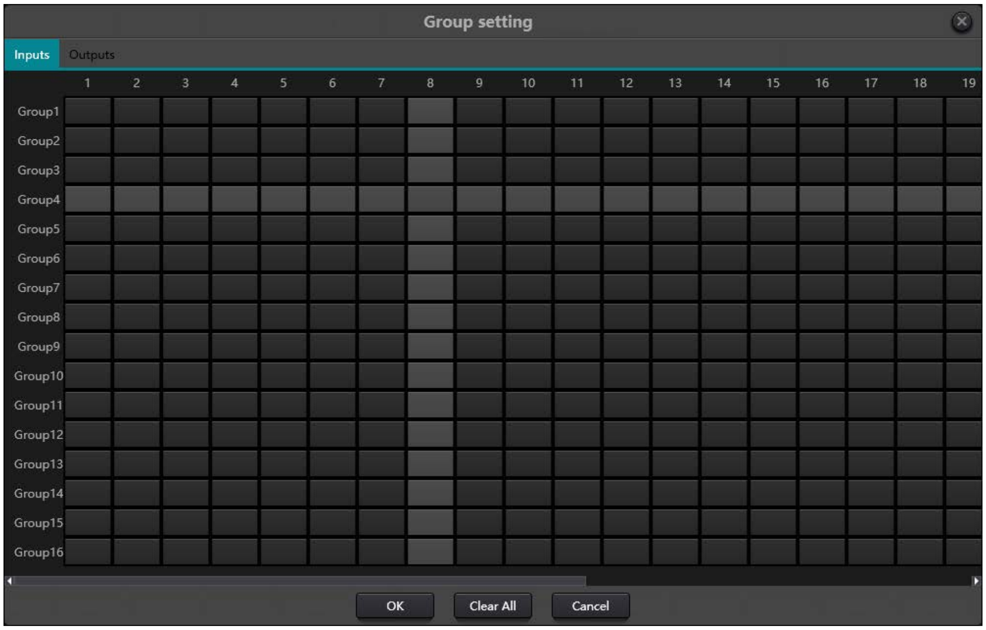
Save as…: Saves the current GPIO configuration to a file. When configuring devices offline, save the GPIO configuration to a file before connecting to a live device. When a live device is first connected, it's onboard settings will overwrite whatever is in the software to ensure proper sync. Once the deice is connected, this file can be applied and saved to the device.

Open…: Opens a dialog box to select a previously saved GPIO configuration file. The configuration will be applied to the GPIO when open.

Save: Saves the configuration to the device. This will be 'grayed out' when the device is not connected.

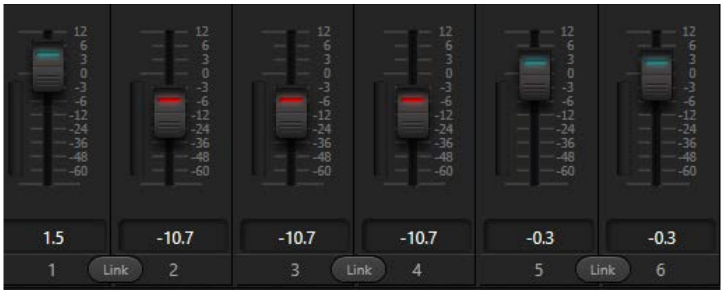
Group Setting

Both input and output channels can be grouped together for the purpose of gain setting and muting, all other parameters remain independent. Grouped channels can be identified by a colored mark on the channel fader, all channels in a particular group will share the same color. Up to 16 input and 16 output groups are supported.



To select channels, simply click in the intersection of the Group name and the channel number. Note that any one channel can only exist in a single group.

Grouped channels only share gain and mute settings, while Linked channels share all parameters. Keep this in mind when creating your groups. Linked channels will still mirror parameters other than gain and mute, however the gain and mute settings of the Group will take precedence over Link.



Inputs: Create Input groups.

Outputs: Create Output groups.

OK: Saves setting to device and closes the dialog box.

Clear All: Clears all group settings.

Cancel: Closes dialog box without saving changes.

Preset Name

NOTE: This menu selection is only active when a DTX device is connected to Dash.

This menu provides the facility to rename the default preset names to more standard language for easier identification. Simply type the new names in the provided field and press ‘OK’. All the preset names will be updated on the device.

Panel Setting

DTX DSPs are designed to work natively with DTX wall panel controllers. This menu item opens a Panel creation application that can be used to assign and customize these DTX wall panels.

User Interface

Dash software can be controlled by up to 8 PC, Android or iOS devices by creating a custom interface. This menu opens a User Interface creation application used to create these custom interfaces.

Help

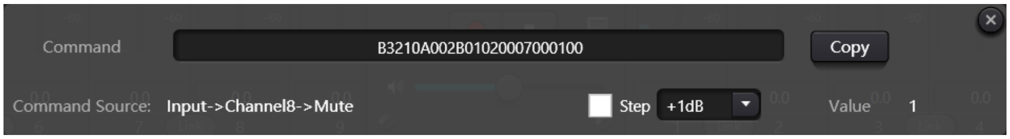
About

Displays software version information.

Document

Help file for Dash software.

Command Capture



The Command Capture window provides an easy capture system for command strings. With the window open, simply click or set any parameter and the command string that corresponds to that command will appear in the window. Using the button, copy the command and use it with a third-party control system to send the command via UDP or RS-232.

Device Tabs

To the right of the Help menu are the Device tabs. Each tab represents a separate DTX DSP device. The Device Name appears as the text in the tab for easy identification. See 'Device Settings' for information on how to change the device name. To the right of the Device name is a small status indicator which will be green if the device is connected, or grey if it is not. A small (x) next to the name can be clicked to close the tab.

New devices can be added by clicking the (+) plus sign to the right of the last device tab. See Device List or more information on adding devices.

Device List

The Device list is the last item on the far right of the top menu line. The device list contains all of the active, discovered DTX devices on the network. Each device on the network will appear in the list with details about the device. Device Name, Device IP and software version. Below this information are two buttons.

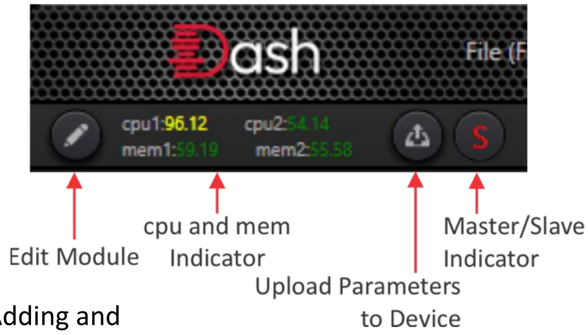
Plus Sign: Click to manually add a device to the list by IP address.

Refresh: Click this to refresh the discovery of new devices on the network.

Connect/Disconnect: Connects the device to the active Device tab. NOTE that the active device tab must match the device type. If the device is already connected, the button will read 'Disconnect'. Clicking the button in this state will disconnect the device from the active Device Tab.

Set IP: This allows the Device IP address to be set. The list is generated by UDP discovery, so it is possible that the IP address is not in the correct subnet for the active network. This is especially true for new DTX devices that still have the default IP address set. The IP address must be set to and address on the active subnet before configuration of the device. NOTE: This IP address is for the LAN interface and is different than the IP address for the Dante Interface.

Edit Module



Below the main software logo is the Edit Module mode button with a pencil icon. Pressing this button enters to module edit mode where individual DSP Modules can be added or removed from channels. Pressing the button again, returns to normal run mode. Next to the Edit Module button is CPU and memory load indicators. These indicate the current load of the connected device and are used when adding modules. NOTE: This mode is only available on certain DTX models and is only active when a device is connected. See the section entitled Adding and Removing DSP Modules for more information about using this mode.

Upload Parameters to Device

This is a small button to the right of the Edit Module button. Pressing this button immediately uploads the current configuration to the connected device. This would typically not be needed in normal operation but is provided as a debugging tool to force immediate synchronization with a device. NOTE: This button is only active when a device is connected.

Master Slave Indicator

To the right of the ‘Upload Parameters to Device’ button is a small circle indicator. This indicates the Backup Device status of the connected DTX device. A red S indicates the device is a slave. This is the normal indication for devices that are not using the Backup Device functionality. When a DTX device is designated as a Host or Master in Device Setting dialog the indicator will display a red M.

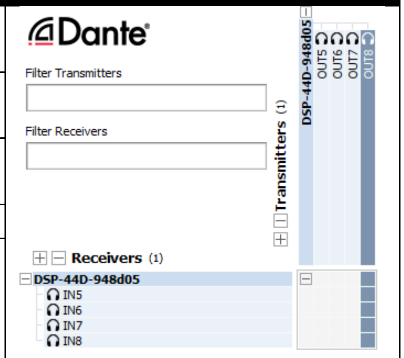
Navigating the Home Tab

The Home tab is the main dashboard for the selected device. It is located below the Device tabs and is the first entry in the line of Module tabs. Click the word 'Home' to enter the Home tab.

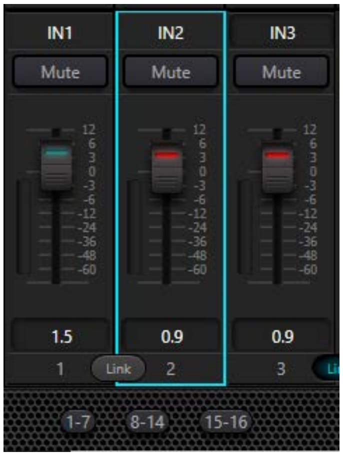
Channel Layout

For both inputs and outputs channels are numbered first with analog followed by Dante. Dante channels can be identified in Dash with the Dante logo above the channel fader. This arrangement flows everywhere in the software – channel layout, Matrix, AEC, etc, everywhere channels are referenced. NOTE Dante channel numbering will follow this numbering scheme. Notice in the image of Dante Controller below that the DTX-44D starts with channel 5.

DTX Model	Channel Layout
DTX-44D	Ch 1-4 Analog Ch 5-8 Dante
DTX-88D	Ch 1-8 Analog Ch 8-16 Dante
DTX-1616D	Ch 1-16 Analog Ch 17-32 Dante
DTX-3232D	Ch 1-32 Dante
DTX-6464D	Ch 1-64 Dante



View of DTX-44D in Dante Controller



Selecting Channels

Clicking the channel label, fader or number will select the active channel. This will be indicated with by teal border around the channel.

Changing the Channel Name

Channel names can be changed by double clicking on the name. Type the new name and hit enter to save.

Setting Channel Gain

Channel Gain can be set by either moving the fader with a mouse, or by typing the value in the gain value box directly below the fader.

Linking Channels

Two adjacent channels can be linked by clicking the 'Link' button between them. All parameter changes to one linked channel will be duplicated on the second.

Channel Pages

At the bottom left and bottom right of the Dash window are buttons that can be used to page through channels. Clicking one of these buttons will scroll the channel bank to view the selected group of channels.

Setting Minimum and Maximum Gain

By default, the minimum and maximum gain levels on the input and output faders are set to their limits of -72dB to +12dB. Right clicking over a fader will bring up a context menu that will allow you to reduce these limits.

Accessing Parameters

There are three ways to access parameters of a channel and its modules.

Right Click: Right clicking over a module will bring up a dialog box for the selected module and channel while staying on the Home tab.

Double Click: Double clicking on a module will bring up either the ‘Inputs’ tab or the ‘Outputs’ tab depending on the type of channel you have clicked. The tab will come up with the module you selected as the active window. This is described in more detail in the next section.

Module Tabs: Channel module parameters can be also accessed by selecting either the ‘Inputs’ or ‘Outputs’ tab from the module tabs. These tabs are described in more detail in the next section.

Module Tabs

Below the main menus and Device tabs are a series of Module tabs. The quantity of these tabs will vary depending on the capabilities of the DTX model you are using.

DTX Model	Available Module Tabs
DTX-44D	Inputs, AutoMixer, AEC, ANS, Matrix, Outputs, Meters
DTX-88D, DTX-1616D, DTX-3232D	Inputs, AutoMixer, AEC, ANS, Matrix, Outputs, Meters, Camera
DTX-6464D	Inputs, Matrix, Outputs, Meters

The detailed functions of these modules are described later in the manual.

Inputs

The Input tab maintains the input fader section across the bottom. Above the fader row is a section which slides right to left and contains all the DSP modules for the selected input channel. Modules can be navigated by sliding the section left to right with the mouse, or by clicking one of the DSP Module names in the button list at the far-right of the screen.

AutoMixer

The AutoMixer tab maintains the Input and Output fader sections divided by the Sound Card as in the Home tab. Above the fader row is the AutoMix System Module.

AEC

The AEC tab maintains the Input and Output fader sections divided by the Sound Card as in the Home tab. Above the fader row is the AEC System Module.

ANS

The ANS tab maintains the Input and Output fader sections divided by the Sound Card as in the Home tab. Above the fader row is the ANS System Module.

Matrix

The Matrix tab maintains the Input and Output fader sections divided by the Sound Card as in the Home tab. Above the fader row is the Matrix System Module.

Outputs

The Output tab maintains the output fader section across the bottom. Above the fader row is a section which slides right to left and contains all the DSP modules for the selected output channel. Modules can be navigated by sliding the section left to right with the mouse, or by clicking one of the DSP Module names in the button list at the far-right of the screen.

Meters

The Meters tab maintains the Input and Output fader sections divided by the Sound Card as in the Home tab. Above the fader row is a set of input and output meters.

Camera

The Camera tab maintains the Input and Output fader sections divided by the Sound Card as in the Home tab. Above the fader row is the Camera Control Module.

System Mute

To the right of the Module tabs is the System Mute and appears as a round button with a speaker. Pressing this button performs a system mute of all channels. The speaker image will turn red when engaged. Click the button again to release the mute.

Presets

The final item to the far right of the Module tabs is the Preset list. Clicking the drop-down list will open it to show the list of 16 presets. Selecting an item from the list will recall the preset. The Settings menu contains the Preset Name functionality which allows for the renaming of the presets.

The save button is used to immediately save the current configuration to the selected preset on the device. This is only available on a connected device.

The Preset Load button loads a saved configuration file (.dsp) from the computer into the selected preset. Simply select the preset from the 'Preset Load' list and an open dialog will appear. Select the file from its location and hit 'Open'. The configuration will be loaded into the preset on the device. See the section titled File Menu for more information about creating a .dsp file. This is only available on a connected device.

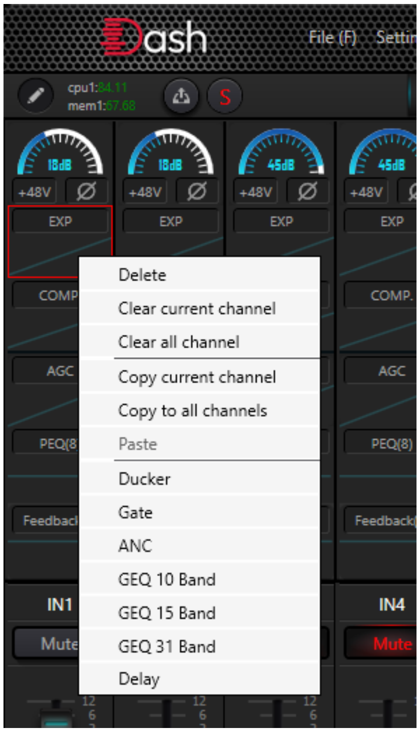
DSP Channel Modules

The Dash DSP Control software has been designed to be incredibly flexible and includes a large variety of DSP Modules, each providing a unique function.

NOTE: The DTX-44D and DTX-6464D have fixed DSP modules, they cannot be changed.

Adding and Removing DSP Modules

DTX-88D, DTX-1616D and DTX-3232D allow for the customization of DSP Modules on a per channel basis. There are five Module Slots on each input channel and four Module slots on each output channel. Modules can be added and removed by entering 'Edit Module' mode.



From the Home tab, click the Edit Mode button which has the pencil icon. Now when you hoover your mouse over one of the module slots, the slot will be outlined in red. Right click over the highlighted module slot to get a context menu. Using the right menu functions configure the device. As you are configuring the device, keep an eye on the 'cpu:' and 'mem:' indicators next to the 'Edit Module' button. These indicate the current load of the device resources based on the current configuration. If they turn yellow, resources are getting thin. If they turn red, you have exceeded the available resources and will not be able to save the configuration.

The following functions can be performed:

Delete: Empties the selected module slot.

Clear Current Channel: Empties all module slots in the selected channel.

Clear All Channel: Empties all input or output channels on the device, depending on whether the selected channel is an input or output.

Copy Current Channel: Copies all the module slots on the current channel.

Copy to All Channels: Copies the current channel modules to all the other channels, either input or output.

Paste: Pastes a previously copied channel module set to the currently selected channel.

Module List: Lists all the available modules that can be used in the selected module slot. Selecting an item from the list will fill the highlighted slot with the selected DSP Module.

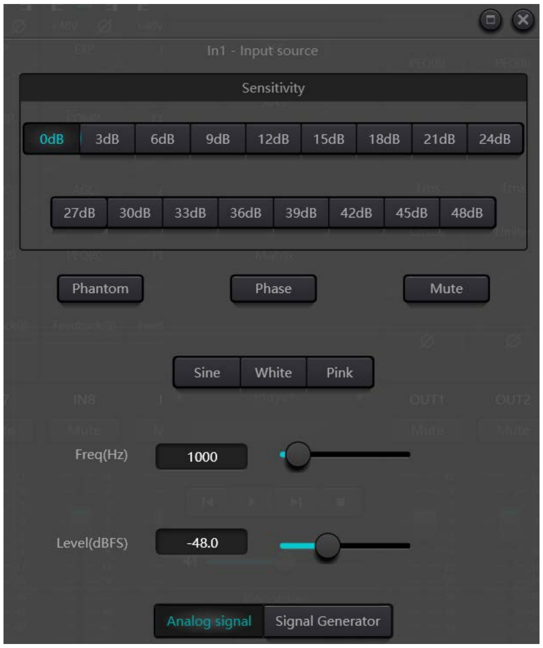
When complete, click the pencil icon again to save your changes to the device. You will be prompted to either Finish, Upload or Cancel.

Finish: Saves the changes locally in Dash but does not upload them to the connected device if one is connected. If there is a device connected it will be discontinued as the device will now be out of sync with the Dash configuration.

Update: Saves the changes to the device. This button will be 'greyed out' if there is no device connected.

Cancel: Exits 'Edit Module' mode and discards changes.

Input Source



The Input Source Module is a fixed module that exists on every input channel of every DTX device. It provides basic input parameters for the selected channel. Since this is a fixed Module, it does not consume one of the five Module Slots available on some DTX models.

Analog Signal: Selects the analog input connected to the external analog connection for the channel source. *(Analog Inputs Only)*

Digital Signal: Selects the Dante input connected to the external analog connection for the channel source. *(Dante Inputs Only)*

Sensitivity: Microphone input gain selectable in 3dB increments from 0 to 48dB. *(Analog Inputs Only)*

Phantom Power: Provide power for external condenser microphone or other phantom powered input peripheral. Do not enable phantom power on a line input source or when the power is not required, so as not to damage the external device. *(Analog Inputs Only)*

Phase: Reverses the phase of the input signal by 180°.

Mute: Mutes the channel.

Signal Generator: Activates the internal signal generator for the selected channel instead of the analog input.

Sine Wave: Sets the output of the Signal Generator to a sine wave with a frequency between 20Hz and 20kHz. You may regulate the output level (unit: dBFS) based on your own needs. Use a fader to adjust or click the text field to designate a value.

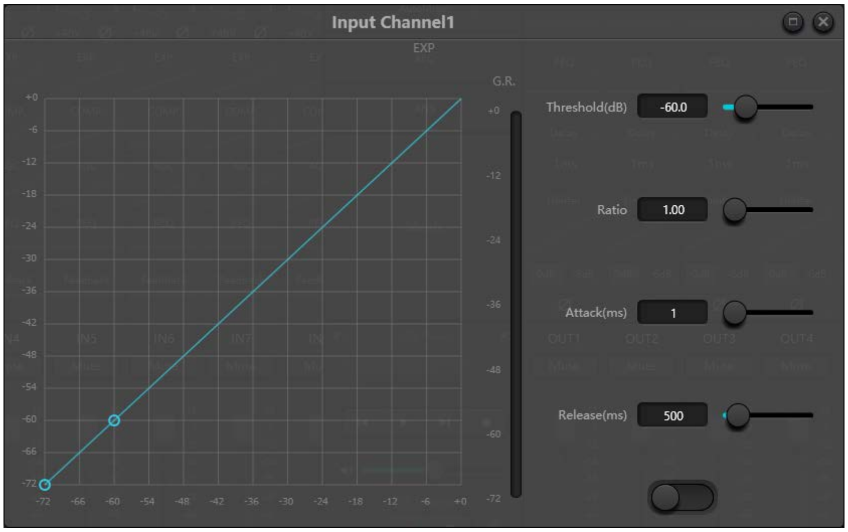
Freq: Use this slider to set the frequency of the sine wave generated. This slider has no effect when white or pink noise is generated.

Level: Sets the level of the Signal Generator. This slider works for all three types of signal generation.

White Noise: Sets the output of the Signal Generator to white noise. White noise is a signal made of uncorrelated samples, such as the numbers produced by a random generator. When such randomness occurs, the signal will contain all frequencies in equal proportion and its spectrum will turn flat. The Freq slider will have no effect on the pink noise, however the Level slider can be used to adjust the amount of signal inserted into the channel.

Pink Noise: Sets the output of the Signal Generator to pink noise. Pink noise is a random signal, filtered to have equal energy per octave. Pink noise has equal power in proportionately wide bandwidths. The Frequency slide will have no effect on the pink noise; however, the level slider can be used to adjust the amount of signal inserted into the channel.

Expander



Expanders are helpful dynamic processing tools that increase the dynamic range of an audio signal. They are essentially the opposite of a compressor, which decreases the dynamic range of an audio signal. The most fundamental difference between and expander and a compressor is that while a compressor engages when a signal is above a defined threshold, an expander engages when a signal drops below a defined threshold. An expander is a more nuanced version of a gate in that it attenuates low level signals below a threshold and leaves those above alone.

Threshold: The expander engages when the signal is below this threshold.

Ratio: Sets the slope below the threshold point on the gain curve. In other words, sets the amount of expansion that is applied.

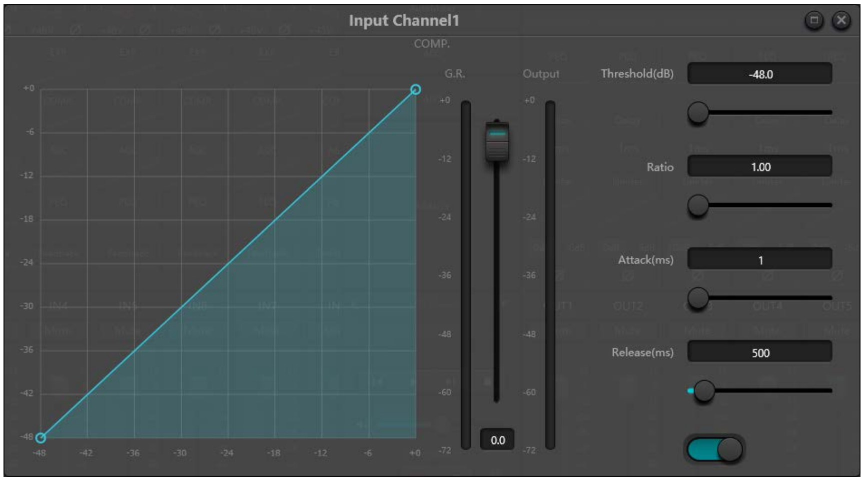
Attack: Determines how fast the expander engages once the threshold is crossed.

Release: Determines how fast the expansion recovers after the signal returns above the threshold.

A combination of Attack and Release provides hysteresis to reduce the changing speed of gain attenuation or what is often referred to as 'breathing'.

ON/OFF: The Expander has a master on/off switch which is used to enable or disable the module. The effected expansion curve will be highlighted in the graph when the module is ON.

Compressor



Audio compression reduces the dynamic range of an audio signal. Dynamic range is the difference between the loudest and quietest parts of a waveform. Compression reduces this range by attenuating the louder signals and boosting the quieter signals. As a result, the loudest and softest parts are closer in volume, creating a more balanced sound. With the extra headroom, you can then increase the overall level of the signal.

Threshold: When the signal level is higher than the threshold, the compressor/limiter begins to reduce the gain. Any signal that exceeds the threshold is regarded as overshoot signal, and its level will be reduced. The more the signal exceeds the threshold, more its level is attenuated.

Ratio: Refers to the compression ratio. The ratio determines how much attenuation to apply to a signal that has overshot the threshold level. A lower threshold will reduce a greater portion of the signal, while a higher threshold will reduce only the loudest peaks and leave the rest untouched. For example, a ratio of 4:1 means that for every 4 dB the signal rises above the threshold, the compressor will increase the output by 1 dB. However, if the ratio is 1:1, no compression will occur. Whereas a ratio of 10:1 or higher will make the compressor effectively act as a limiter.

Attack: Sets the time it takes for the compression to kick in once the signal passes the threshold. Faster attack times will immediately clamp down on the initial transients of a sound. This is good for controlling dynamics and catching transients that cause clipping. Whereas slower attack times allow more of the initial attack to punch through before compression begins.

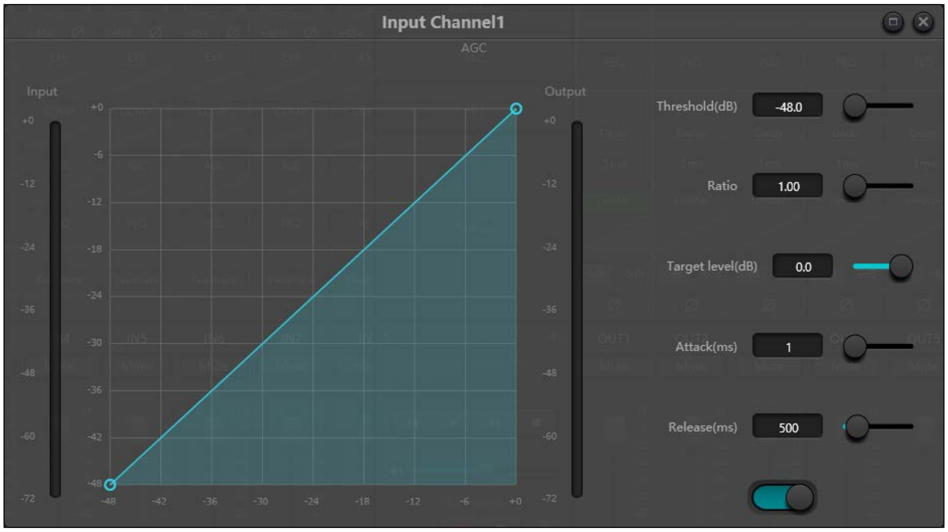
Release: Sets the time it takes for the compression to stop once the signal falls below the threshold. Faster release times sound more natural and transparent. However, setting the release too fast can cause a "pumping" effect and other unnatural sounds. Whereas longer release times are good for smoothing out dynamics.

Output Gain: This is also called gain compensation fader. If the compressor significantly reduces the level of the signal, it may be desirable to boost output gain to compensate.

G.R. and Output Level Meter: G.R. indicates the amount of Gain Reduction (compression) applied; Output refers to the output level of the signal that has passed through the compressor module. The compression amount is displayed in an inverse level meter. If the input signal and threshold are set as -6dB and -30dB, respectively, and the ratio is 2:1, then the compression amount is 12dB; G.R. level meter will indicate around -12dB and output meter around -18dB.

ON/OFF: The Compressor has a master on/off switch which is used to enable or disable the module. The effected compression curve will be highlighted in the graph when the module is ON.

Automatic Gain Control (AGC)



Automatic Gain Control (AGC) is similar to a compressor, except while a compressor reduces the dynamic range of a signal, an AGC maintains the dynamic range while automatically boosting the signal level. The purpose of an AGC is to improve or normalize signal level while maintaining the dynamic range. For example, normalizing the level of audio playback of background music so that it is not drastically different than the level of a paging microphone, or, establishing a consistent audio between varying microphone levels in a conference. The AGC includes silence detection to prevent wild gain swings during silent periods.

Threshold: When the signal level is lower than the threshold, the input-to-output ratio is 1:1. When the signal level is higher than the threshold, the input-to-output ratio changes according to the ratio control settings. The threshold will typically be set just above the noise floor. This assures that the noise floor itself will not trigger the AGC, but any signal above that will.

Ratio: Sets the desired ratio between input signal and output signal.

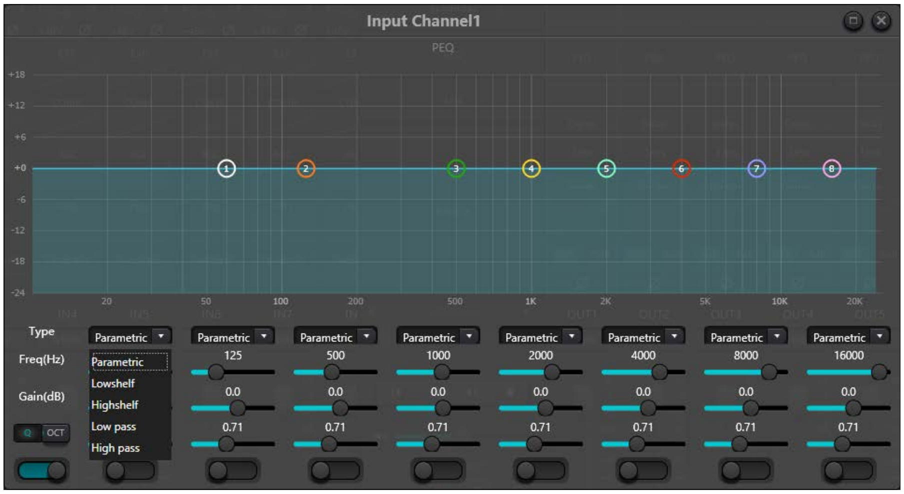
Target Threshold: Sets the desired output signal level. If the signal is higher than the threshold, the controller will compress the signal in proportion.

Attack: Sets the time it takes for the AGC to kick in once the signal passes the threshold.

Release: Sets the time it takes for the AGC to stop once the signal falls below the threshold.

ON/OFF: The AGC has a master on/off switch which is used to enable or disable the module. The effected gain curve will be highlighted in the graph when the module is ON.

Parametric EQ



A Parametric Equalizer is an adjustable filter used to shape a signal's frequency content. The Parametric EQ allows control of multiple frequency bands and has adjustable parameters to control the center frequency, the gain or attenuation of that frequency and the Q (Quotient of Change) aka bandwidth around that frequency that is affected. While it is more complex than other types of equalizers, a Parametric EQ provides surgical precision and fine adjustment. For example, the squeal of a running air conditioner could be eliminated from a microphone signal without drastically effecting the desired audio content by using a very narrow Q with significant attenuation at the exact frequency of the squeal. The Dash software provides incredibly flexibility by offering 5, 8 and 12-band Parametric EQ modules (not available on all hardware).

Type: Parametric EQ is a default. High & low shelf filters and high & low pass filters can also be selected. Each kind of filter has different frequency shapes to achieve different functions.

High & Low Pass Filter: The reference frequency of pass-type filter is called the cut-off frequency. Pass-type filter allows the frequencies on one side of the cut-off frequency to fully pass the filter while the frequencies on another side of the cut-off frequency are attenuated. A high pass filter allows the frequencies above the cut-off frequency to pass and filters the frequencies below the cut-off frequency, while a low pass filter allows the frequencies below the cut-off frequency to pass and filters the frequencies above the cut-off frequency.

High & Low Shelf Filter: High shelf filter means that the gain enhances or attenuates for the frequencies above the set frequency. Low shelf filter means that the gain enhances or attenuates for the frequencies below the set frequency. The set frequency is not the 3dB cut-off frequency but refers to the center of the failing edge or rising edge of the filter. Q value affects the peak and has a mathematical relationship to the peak.

Frequency (Hz): Sets the center frequency of the filter.

Gain (dB): It refers to the amount of gain or attenuation applied at the center frequency.

Q : Q Represents 'Quotient of Change' or the bandwidth. This sets how wide a swath of frequencies are affected surrounding the center frequency. The adjustable range of Q value is 0.02-50.

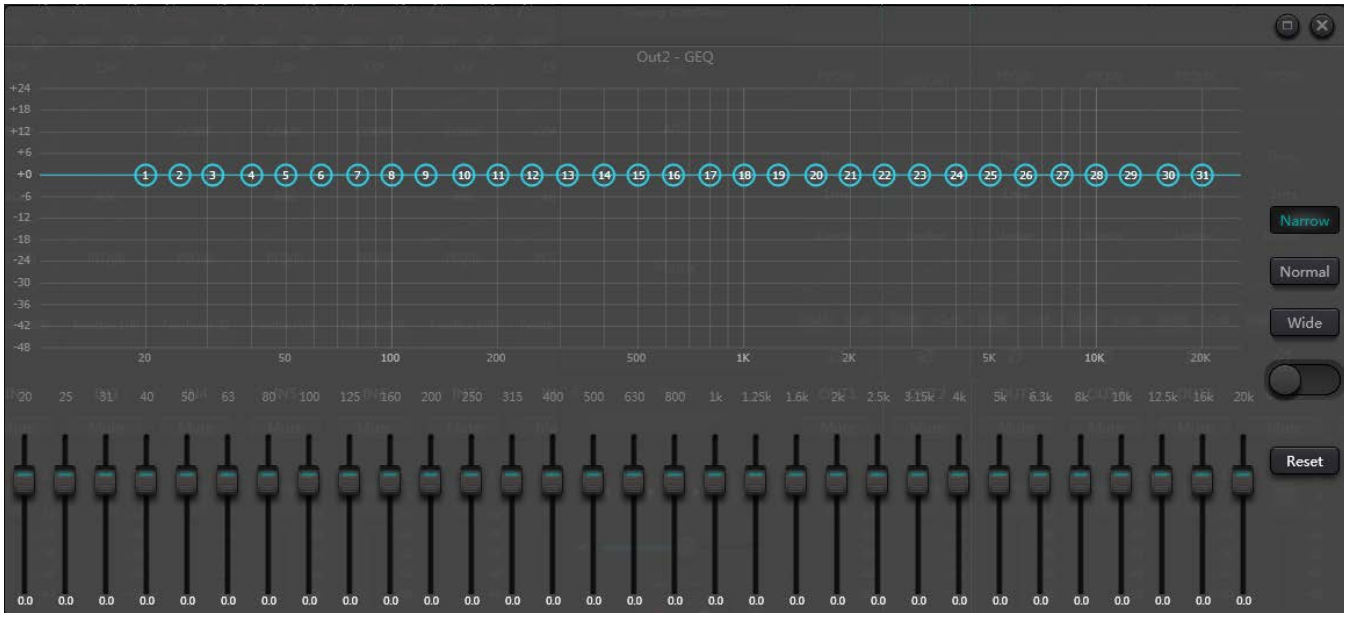
When the filter is a parametric EQ filter, Q value refers to the width of the bell-shaped frequency response curve on both sides of the center frequency.

When the filter is a high & low shelf filter or a high & low pass filter, if Q>0.707, there will be peaks in the filter responses. If Q<0.707, the slope will become flatter, and the roll-off will occur in advance.

OCT: Changes the units from Q-value to octaves. When dealing with music content, measuring bandwidth in terms of octaves can provide more musical results. A wider octave setting will make the content more natural and musical, on narrow settings it allows you to know the musical notes you are affecting as determined by the center frequency.

ON/OFF: Each segment of equalizer has a switch which is used to turn on or turn off the corresponding segment. If set to OFF, the parameter position is disabled. The Parametric EQ has a master switch located on the far left, which is used to enable or disable the entire module. The effected frequency response curve will be highlighted in the graph when the module is ON.

Graphic EQ



Like the Parametric EQ, the Graphic EQ is an adjustable filter used to shape a signal's frequency content. The Graphic EQ, however, is far simpler. Unlike the Parametric EQ, the Graphic EQ has its frequencies, and Q (bandwidth) fixed. Each fixed center frequency has its own gain control. There is one important thing to understand about graphic equalizers. When you adjust a slider, it also affects the neighboring frequencies. The Dash software provides incredibly flexibility by offering 10, 15and 31-band Graphic EQ modules (not available on all hardware).

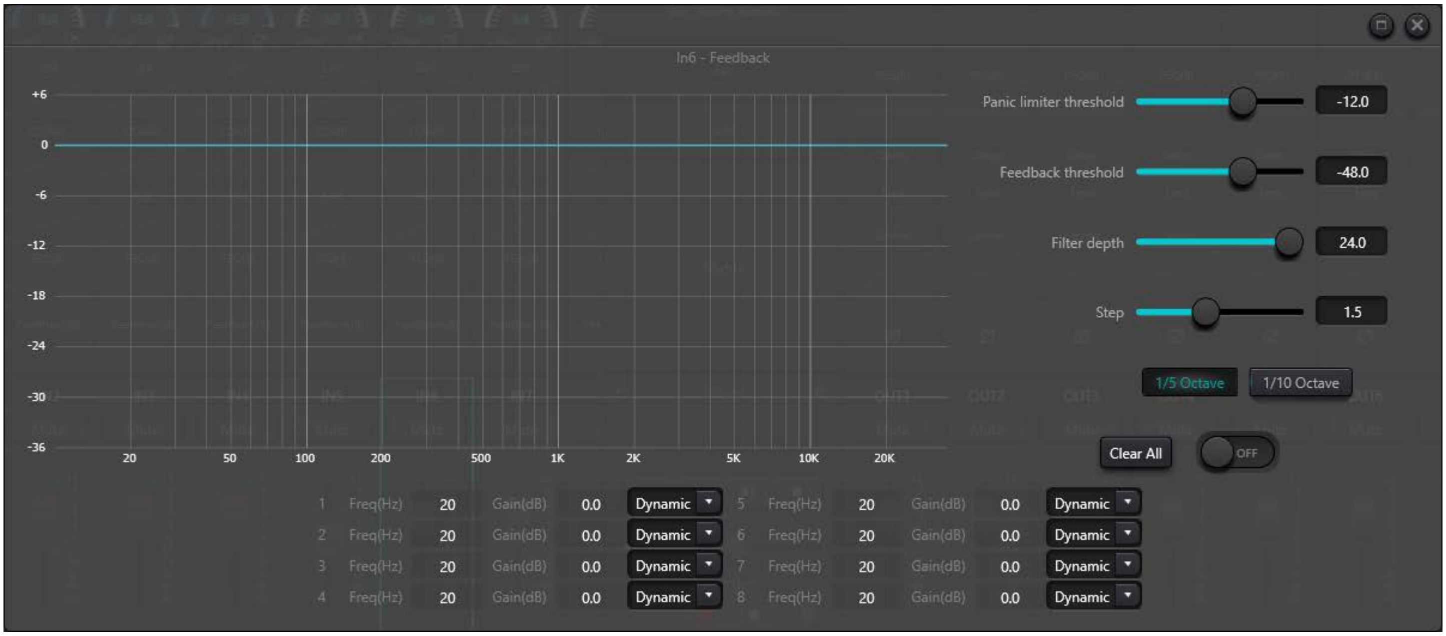
Frequency Bands: Gain control for the specified frequency. Value ranges are from +12 dB to -24 dB (attenuation).

Narrow/Normal/Wide: Selects from one of three preset Q-values. This setting applies to all frequency bands.

Reset: Resets gain values for every frequency band to 0 dB and turns OFF the Graphic EQ module.

ON/OFF: The Graphic EQ has a master on/off switch which is used to enable or disable the module. The effected frequency response curve will be highlighted in the graph when the module is ON.

Adaptive Feedback Control (AFC)



The Adaptive Feedback Control (AFC) module automatically detects and inhibits audio feedback. The module differentiates undesirable feedback from the desired audio content based on specific characteristics of the audio signal. When a feedback instance is detected, a notch filter (narrow Q) will be automatically added at the center frequency of the feedback. The applied attenuation is progressive meaning it starts with minimal attenuation at the feedback frequency, slowly increasing the attenuation, in compliance with the set parameters, until the feedback is eliminated or falls below the Feedback Threshold value.

Once a feedback event has been handled and the notch filter engaged, the filter will be locked to prevent change. These filters can be duplicated in a channel equalizer for a more permanent fixed feedback solution. There are eight (8), twelve (12) or sixteen (16) narrow-band filter slots in an automatic cycle depending on which AFC module was selected. Eight is the default.

Panic Threshold: Positively defines a feedback event. Any signal above the Panic Threshold will be considered feedback. When a signal level exceed the Panic Threshold any of the following will occur to attempt to get the feedback under control:

- Gain will be attenuated to attempt to control the feedback
- The output of the AFC module will be limited to prevent runaway feedback
- The filters sensitivity will be increased to provide faster detection

When the signal returns to below the threshold, all settings are returned to their pre-feedback event state.

Feedback Threshold: Positively defines what is NOT a feedback event. Any signal that is below this threshold is not considered feedback. Setting this threshold too high can diminish the effectiveness of the AFC.

Filter Depth: Sets the maximum attenuation of a single filter. A low setting will minimize the effect on the overall signal; however, it will also diminish the effectiveness of the AFC.

Step: Sets the Notch Depth Step Size (dB). Defines the depth of correction of each step. When a new feedback event occurs, the initial depth of the notch filter is set to twice this value. If the feedback persists, the notch will be deepened be this amount at regular intervals until the feedback is mitigated or the Filter Depth is reached. Larger values can cause 'over-correction' of the problem. In general, larger rooms should use smaller steps as the delay time of the room will cause the feedback to take longer to dissipate.

Bandwidth: 1/10 and 1/5 octave can be selected. A constant Q value applied to the notch filter, the filter will not become wider due to the increase of depth. A value of 1/10 octave should be used by default as it is narrower, it will not affect the desired signal as much as the 1/5 octave bandwidth. If feedback is persistent, the bandwidth can set at 1/5 octave because it has wider bandwidth and can catch more feedback events. The tradeoff, however, is that the notch filter will be more noticeable in the desired signal content.

Dynamic/Manual: Each of the notch filters can be set to either Dynamic or Manual mode. The default should be Dynamic as that will utilize the built-in content analysis and feedback detection algorithm. The Manual setting will effectively turn the related channel into a parametric EQ channel. Center frequency and gain can be manually set, and the Q-value is selectable using the 1/5 or 1/10 octave buttons.

Clear: Click the button to instantly clear all notch filters. This operation is generally done when recommissioning the AFC module.

Commissioning

The AFC module can be used as a tool during system commissioning to identify feedback points or as a preventive measure during normal operations. To maximize channel gain and AFC effect, use the following steps to set up the AFC.

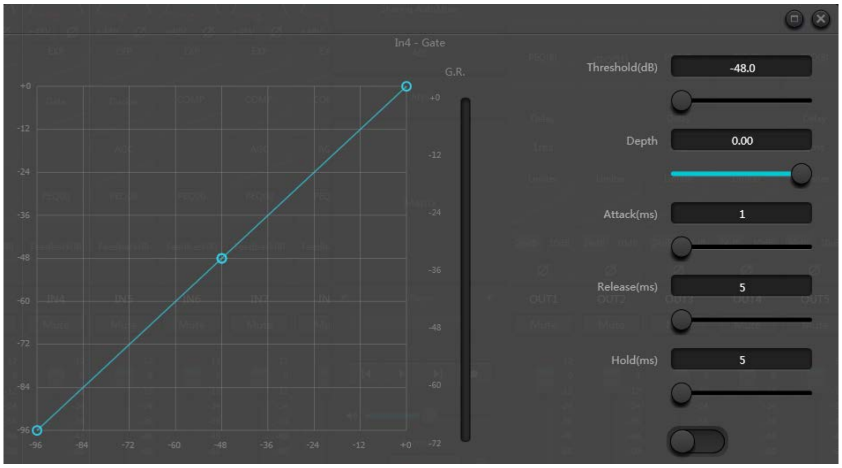
1) Reduce the channel volume, and use the button "Clear All" to reset all notch filter parameters
2) Set up parameters for the feedback inhibition module as desired. Start with a low Panic Threshold to make sure you can identify all feedback events.
3) Open all microphones, and slowly increase channel volume until the feedback occurs. Stop increasing channel volume when the feedback occurs.
4) Wait for the AFC module to take effect; after the feedback disappears, continue to increase gain.
5) Repeat the operation until the system reaches the required gain or until all filters are fully distributed
6) Change the Panic Threshold to a maximum level just higher than the expected non-feedback signal.

After initial setup, notch filter setting can be transferred to a channel Parametric EQ for permanent suppression, freeing up the notch filter for any potential new feedback event that could occur in the future.

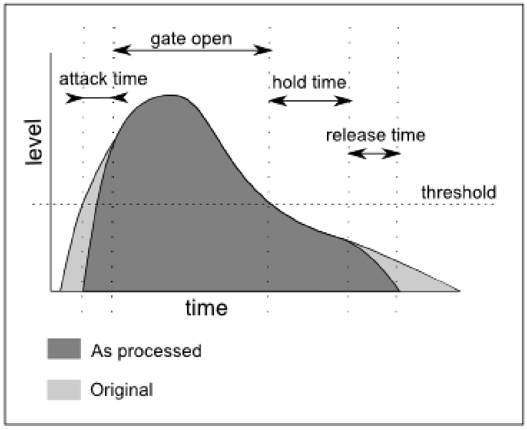
If a speaker is included among the devices used, it is recommended to use a compressor/limiter module on those outputs for additional protection. You should set an appropriate limiter to make sure that the speaker will not get damaged even if all notching filters slots are used or the feedback inhibitor cannot effectively control the feedback, such as in the case of excessive system gain.

ON/OFF: The AFC has a master on/off switch which is used to enable or disable the module. The effected frequency response curve will be highlighted in the graph when the module is ON.

Gate



The primary purpose of a Gate is to clamp off signals below a defined threshold, typically noise. Beyond just controlling IF a signal can pass through a channel, a Gate can also control how quickly the signal is allowed to pass, how long the Gate stays open and how soon the Gate can be fully opened again. Gates can be used in a couple of ways. First, they can be used to reduce unwanted external sounds. Think of someone speaking into a microphone in a busy office environment. You can use a Gate to reduce or eliminate the background noise when they are not speaking. Secondly, they can reduce natural channel noise. All audio equipment has an inherent noise floor. A Gate can be used on a channel where one of these devices is connected, to silence the noise when the device is silent.



Threshold: Any signal greater than the Threshold value will open the Gate, below the Threshold will close the Gate.

Depth: Determines how much attenuation below the Threshold will be applied to gated signal.

Attack: Sets how quickly the Gate will open after the signal exceeds the Threshold.

Release: Sets how quickly the Gate will close after the signal drops below the Threshold and after Hold has expired. Release is the key to a natural sounding audio decay.

Hold: Sets the time the Gate will stay open after the Threshold has been passed. This can be important on signals with fast varying levels. A longer Hold value will keep the Gate from producing an audible 'chatter' in this scenario.

ON/OFF: The Gate has a master on/off switch which is used to enable or disable the module. The effected gain curve will be highlighted in the graph when the module is ON.

Ducker

A Ducker is a device used for lowering the level of one audio source based on the presence of another audio source. Duckers are often used in paging systems to lower the level of background music during a page. In this application, the Ducker restores the background music once the page is completed.

Reference Source: Sets the source channel used to trigger the Ducker. This can be more than one channel.

Threshold: When the reference source exceeds the Threshold, the Ducker engages.

Depth: Determines how much attenuation below the Threshold will be applied to ducked signal.

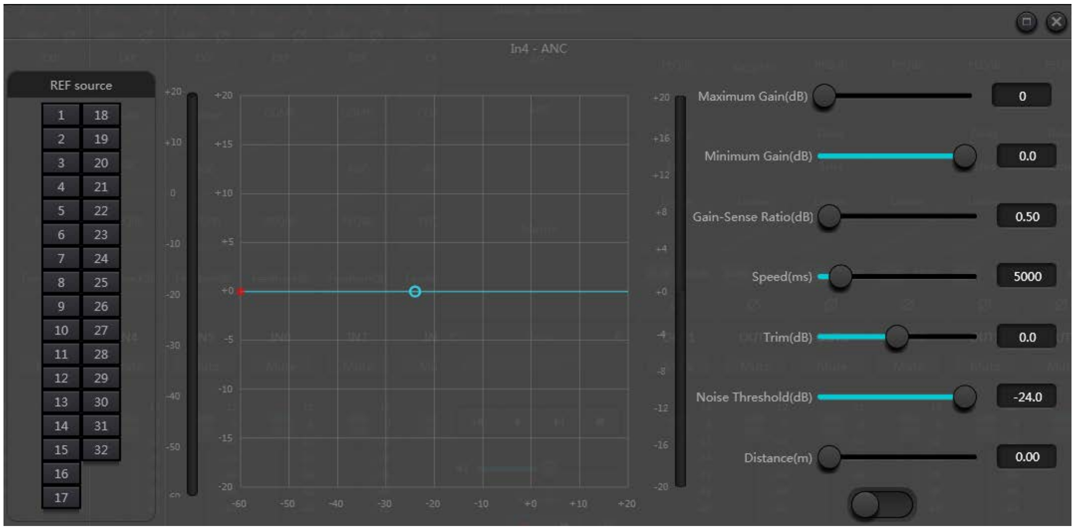
Attack: Sets how quickly the Ducker will engage after the reference source exceeds the Threshold.

Release: Sets how quickly the Ducker will release after the reference source drops below the Threshold and after Hold is expired.

Hold: Sets how long the Ducker will stay engaged after the reference source drops below the Threshold.

ON/OFF: The Ducker has a master on/off switch which is used to enable or disable the module. The effected gain curve will be highlighted in the graph when the module is ON.

Ambient Noise Compensation (ANC)



Ambient Noise Compensation (ANC) automatically adjusts the volume of the channel to compensate for ambient noise in the space. This is achieved through a sophisticated algorithm which listens to the ambient noise via a dedicated microphone. For example, consider music and announcements in a bar or restaurant setting. When the bar is empty, the ambient sound is low and thus, the program (music/announcements) do not to be too loud. As the bar fills, the ambient noise gets louder, and the program will get drowned out in the noise floor of the room. ANC can compensate for this by actively ‘listening’ to the ambient in the room and adjusting the volume accordingly.

Reference Source: Sets the source channel used as the ‘listener’ for ANC. This can be more than one channel, for example multiple microphones in a large space.

Maximum Gain: Sets the maximum amount gain that will be applied to the channel.

Minimum Gain: Sets the minimum gain level for the channel signal.

Gain-Sense Ratio: Sets the ratio of attenuation or gain.

Speed: Sets the speed that the ANC makes a gain correction.

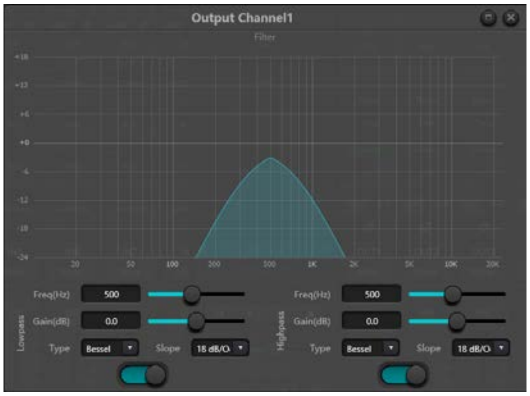
Trim: Adds gain or attenuation to the resulting output of the ANC.

Noise Threshold: Level of the Reference Source above which the ANC is engaged.

Distance: Sets the physical distance between Reference Source and local signals.

ON/OFF: The ANC has a master on/off switch which is used to enable or disable the module. The effected gain curve will be highlighted in the graph when the module is ON.

High and Low Pass Filter



Each output channel has a High and Low Pass Filter (Filter). A pass filter is a useful tool in audio mixing. That’s because, unlike other EQs which turn frequencies up or down, a pass filter allows you to filter out portions of the frequency spectrum. There are two main types of pass filters, high-pass filters (HPF) and low-pass filters (LPF). A high-pass filter, also referred to as a ‘low-cut filter’, does exactly what the name suggests. It allows only frequencies that are higher than a certain point to pass through. Simultaneously, it filters out the frequencies that are lower than that point. A low-pass filter, also referred to as a ‘high-cut filter’, allows only frequencies that are lower than a certain point to pass through. Simultaneously, it filters out the frequencies that are higher than that point.

Frequency: The cutoff frequency of filters. The cutoff frequency of Bessel and Butterworth is defined at -3 dB, and the cutoff frequency of Linkwitz-riley is defined at -6dB.

Gain: Sets the gain or attenuation applied to the signal.

Type: There are three types of filters including Bessel, Butterworth, and Linkwitz-Riley.

The purpose of the Bessel filter is to achieve approximately linear phase, linear phase being equivalent to a time delay. This is the best phase response from an audible standpoint, assuming you don't want to correct an existing phase shift.

The Butterworth filter is a type of signal processing filter designed to have as flat frequency response as possible (no ripples) in the passband and zero roll off response in the stopband. However, one main disadvantage of the Butterworth filter is that it achieves this pass band flatness at the expense of a wide transition band as the filter changes from the pass band to the stop band. It also has poor phase characteristics as well.

The Linkwitz-Riley filter concept came from cascading pairs of identical two-pole Butterworth filters in low pass and high pass configurations. The result is a crossover filter that uses the same corner frequency in high pass and low pass, has no peaks or dips and is phase continuous at the crossover frequency. In other words, at the crossover frequency, both drivers are in phase, each contributing half the energy they would in their passbands.

Slope: It refers to attenuation values of transition zone of filters. There are totally 8 attenuation values including 6, 12, 18, 24, 30, 36, 42 and 48dB/Oct. For example, 24dB/Oct indicates that the attenuation range is 24dB for each octave difference existed in frequency in transition zone.

Users may click bottom activate button to activate high- or low-pass module.

ON/OFF: The Filter has two on/off switches which are used to enable or disable the High Pass and Low Pass Filters independently. The effected frequency response curve will be highlighted in the graph when the module is ON.

Delay

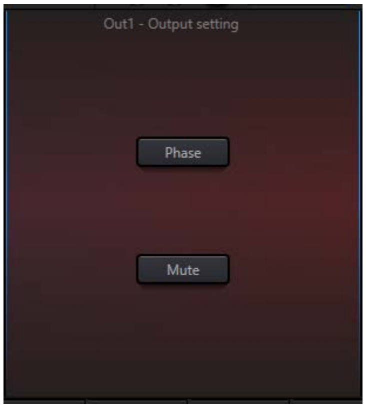
The Delay Module inserts delay into the output of the selected channel. Delay is often used in DSP processors for the purpose of time alignment. This can be either for time alignment of speaker elements if the outputs are used to drive different speaker drivers, or to align an audio output to a video source. In a multi-element loudspeaker or array, the sound from different drivers will take slightly different amounts of time to reach your ears. Since a woofer cone is "deeper" than a tweeter, the acoustic center of the woofer is further away from the listener's ear than the tweeter. The sound from the woofer will therefore arrive at the listener's ear slightly later than the sound from the tweeter. This can have a detrimental effect on the speaker's response around the crossover. The Delay Module can delay the output signal destined for the tweeter by a small amount of time. The result is that the acoustic waveform from both drivers will arrive at the listener's ear at the same time.

Milliseconds, Meters and Feet: All these parameters are linked, they display the same delay time measured in three different units. Entering a value in any one of the parameters will automatically be converted to the other units of measure and filled in.

Delay Slider: Provides a graphical way to enter the delay value.

ON/OFF: The Delay has a master on/off switch which is used to enable or disable the module. The 'Delay' indicator will be green when the module is ON.

Output Setting



The Output Setting Module is a fixed module that exists on every output channel of every DTX device. It provides basic output parameters for the selected channel. Since this is a fixed Module, it does not consume one of the four output Module Slots available on some DTX models.

Phase: Reverses the phase of the output signal by 180°.

Mute: Mutes the output.

DSP System Modules

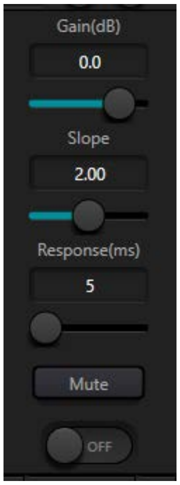
Gain Sharing Auto Mixer

The Dash Gain Sharing AutoMixer is a multichannel mixer that automatically lowers the level of non-active audio sources such as microphones while simultaneously riding the level of active sources to maintain the desired target gain. This eliminates the need for tedious and often impractical live mixing of audio sources. The alternative is to just leave all source channels open, however that can severely degrade the overall quality and user experience allowing hum, background noise, etc. from inactive open microphones to be mixed with the desired signals. The Dash Gain Sharing AutoMixer solves that problem elegantly by using the intelligence of the DSP to optimize the audio mix, prioritizing desired content while diminishing undesired content.

Gain Sharing provides the most natural sounding mix of multiple microphones. Some AutoMixers use gating technology to gate inactive microphones while maintaining a defined maximum Number of Open Mics (NOM). While this technique functionally ‘works’, one can imagine the unnatural sounding result as microphones are being effectively muted and unmuted during conversation. A Gain Sharing Automixer uses mathematics to reduce the individual gain of a microphone as microphones are added. Fundamentally, for every doubling of the NOM, the AutoMixer reduces the overall gain by 3dB. The effect is a seamless, natural, and conversational sounding mix.

There are two groups of control parameters in the Gain Sharing AutoMix module: main control parameters and channel control parameters.

Main Control Parameters



Gain: Sets the desired target gain for the output of the AutoMix.

Slope: Determines how aggressive the gain sharing is. Low Slope settings cause the module to behave just like a conventional mixer. High slope settings cause the inputs to attenuate more deeply for inactive microphones, similar to a gate. A value of around 2.0 provides the most natural sounding effect in most applications.

Response: Sets the speed at which the AutoMix operates. A faster response time may be needed if different speakers are speaking in quick succession however a response value that is too fast could cause unwanted ‘pumping’ of the audio between spoken words. Conversely, too slow of a response could cut off the beginning sounds of words. However, the very nature of the Gain Sharing algorithm provides for a faster on time than off, to help prevent this effect, even at very fast Response settings. Practice shows that the most natural effect will be produced with a Response value between 100ms and 1000ms.

Mute: Mutes the output of the AutoMixer.

ON/OFF: The Auto Mixer has a master on/off switch which is used to enable or disable the module. The ‘Gain Sharing AutoMix’ indicator will be green when the module is ON.

Channel Control Parameters

Fader/Gain: Sets the channel gain. Faders can be used to tell the Gain Sharing algorithm the desired relative level of each channel. NOTE: Channel gain is after the input to the AutoMix engine. Channel level may still influence the gain adjustment of other channels even if the fader is all the way down.

Priority: Sets the priority level of the selected channel. This value feeds into the AutoMix engine and can influence automatic gain adjustment decisions. Priority values range from 0 to 10. The higher the value the higher the priority of the channel.

AutoMixer: Enables AutoMix participation of the selected channel.

Mute: Mutes the channel. NOTE: Mute is after the input to the AutoMix engine. A muted channel may still influence the gain adjustment of other channels in the AutoMix.

AutoMixer Application Notes

Both channel gain and channel mute in the AutoMix module are after the AutoMix engine in the signal chain. Therefore, as long as the channel is participating in the AutoMix it will influence the algorithm – even if the channel is muted or the gain is all the way down.

If two or more channels have the same signal level, the channel with the highest priority will get more auto gain. It should be noted that great care should be given when assigning priority values. It is best to avoid wide differences in between channel priorities for example 0 and 10. If the channel with a value of 10 can recognize even background noise as valid content and then channels with a priority of 0 could be masked all together and never added to the AutoMix. This effect gets worse with higher Slope values. If it is necessary to have such diverse priorities, it is recommended to add a gate or expander to the highest priority channels to eliminate unwanted noise.

Gating AutoMixer

In addition to a Gain Sharing AutoMixer, Dash provides the option of a more traditional Gating AutoMixer (Relative Threshold). When in Edit Module mode, right clicking on the AutoMixer slot in the top center of the Home Tab, provides the option of using either a Gain Sharing AutoMixer or a Gating AutoMixer.

As the name implies, a Gating Auto Mixer employs active noise gates to only allow active channels to go through and prevent inactive channels from contributing to the overall mix. A channel is considered active if it is able to breach the threshold set for the noise gate.

Main Control Parameters

Gain: Sets the desired target gain for the output of the AutoMix.

Hold: Sets the amount of time a channel remains open after the signal on that channel falls below the Noise Threshold. This ensures that a slight pause when speaking does not close the channel.

Off Gain: Sets the amount of attenuation when the channel is inactive/closed. It does not affect the AutoMix logic. Setting this value too high will allow a room noise and echo on the gated channel to come through.

Sensitivity: Determines how far above the Noise Threshold the channel must be to open the gate.

NOM Atten: Sets the logarithmic attenuation of the output based on the Number of Open Mics. The output is attenuated by this value each time the NOM doubles. For example, if the value is set to 3dB, then when there are two open mics (NOM = 2) the attenuation is 3dB. When NOM equals 4 then 6dB of attenuation is applied, NOM equals 8 then 9dB, etc.

NOM Limit: Sets the maximum number of open microphones (channels) allowed at any given time.

Noise Threshold: Noise Floor Tracking follows the steady or slowly changing noise of equipment fans, HVAC, and so on. In order for a channel to open its gate, the signal must exceed the Noise Threshold control.

Last On: Leaves the last channel that was used in the open condition, until another channel signal exceeds the Threshold.

Mute: Mutes the output of the AutoMixer.

ON/OFF: The Auto Mixer has a master on/off switch which is used to enable or disable the module. The ‘Gain Sharing AutoMix’ indicator will be green when the module is ON.

Channel Control Parameters

Fader/Gain: Sets the channel gain. Faders can be used to tell the Gain Sharing algorithm the desired relative level of each channel. NOTE: Channel gain is after the input to the AutoMix engine. Channel level may still influence the gain adjustment of other channels even if the fader is all the way down.

Priority: Sets the priority level of the selected channel. This value feeds into the AutoMix engine and can influence automatic gain adjustment decisions. Priority values range from 0 to 10. The higher the value the higher the priority of the channel.

AutoMixer: Enables AutoMix participation of the selected channel.

Mute: Mutes the channel. NOTE: Mute is after the input to the AutoMix engine. A muted channel may still influence the gain adjustment of other channels in the AutoMix.

Default: Sets the default channel. When all other channels are gated, the single default channel will be opened.

Acoustic Echo Cancellation (AEC)

Acoustic Echo Cancellation (AEC) is important for audio teleconferencing when simultaneous communication (or full-duplex transmission) of speech is necessary. In acoustic echo cancellation, a measured microphone signal contains two signals: the near-end speech signal (program or local) and the far-end echoed speech signal (reference or remote). The goal of AEC is to remove the reference signal from the microphone signal so only the program signal is transmitted.

AEC is not just useful in telephony applications; they can be used anywhere one signal (remote) needs to be removed from a second signal (local). For example, consider a drive-thru ordering system at a restaurant. The person taking the order (remote) speaks into a microphone and that audio is broadcast so the person in the car (local) can hear it. The audio coming from the speaker is in a unique acoustic environment. The audio can reflect off the car, the curb, other building and back into the talk back microphone. The order taker would hear back everything they said as an echo. By implementing AEC, the remote audio from the order taker can be subtracted out of the talk back mic audio, eliminating the echo.

Local Input: Selects the source or sources which contains the desired audio signal as well as unwanted echo from the remote audio signal.

Remote Input: Selects the source or sources for the audio signal which will be subtracted from the Local Input.

Non-Linear Processor (NLP): Sets the ‘tail time’ in the NLP. The ‘tail time’ is the length of time the reference (or remote) signal will be compared with the program (or local) signal and used for echo estimation. If the AEC has an NLP value shorter than the length of the longest echo, echo will still be heard in the program signal. The following preset values can be selected:

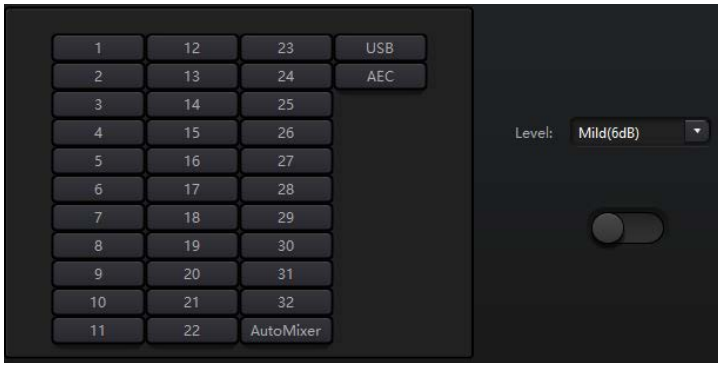
Selection	Tail Time
Conservative	128ms
Moderate	256ms
Aggressive	512ms

ON/OFF: The AEC has a master on/off switch which is used to enable or disable the module. The ‘AEC’ indicator will be green when the module is ON.

Note: Use the output of the AEC to route the resulting audio in the Matrix. Do not add the both the AEC and the local source into the Matrix together or the desired effect will be lost.

Automatic Noise Suppression (ANS)

Automatic Noise Suppression (ANS) module can remove audio that it detects is noise. It utilizes an intelligent algorithm which distinguish human voice from other sounds and treats the latter as noise. The noise is then subtracted from the original audio source.



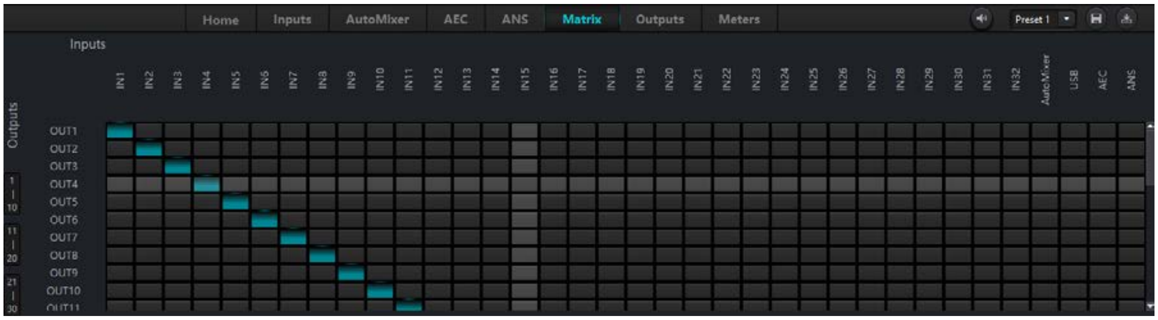
Input Selection: Selects source or sources for ANS.

Level: Sets the attenuation of the non-human voice audio.

Value	Attenuation
Mild	6dB
Medium	10dB
Aggressive	15dB

ON/OFF: The ANS has a master on/off switch which is used to enable or disable the module. The ‘ANS’ indicator will be green when the module is ON.

Matrix



The Matrix in the Dash software is the heart of the DSP. It is a true mixing matrix and has dual functions. At the most basic level it is a signal router. Simply clicking at the intersection of an input and output routes the input to the output. A blue highlight indicates a route point.

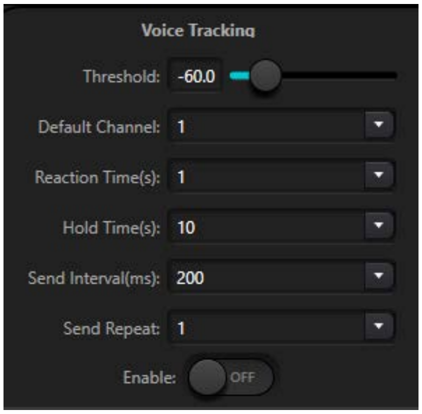
The Matrix can also be used as an audio mixer. If it desired to route two or more inputs to the same output, simply click the intersection of all desire inputs to the desired output. The selected input channels will be mixed and routed to the selected output. Additional balance control of the input channels can be achieved by right clicking the blue intersection point. This will bring up a small fader window where the amount of input signal can be adjusted. The resulting adjustment will be displayed at the intersection. If the intersection has no text, that indicates a gain adjustment of 0dB.

Camera Tracking

The Camera Tracking tab is split into two distinct functional categories. The first two sections are tied together to configure Automatic Voice Tracking Cameral Control. The two sections to the furthest right of the panel function as live, Manual Camera Control.

The Automatic Voice Tracking Camera Control is an advanced and highly flexible feature which allows for seamless automation of camera switching to provide 'broadcast-like' production of meetings and conferences. Every input on the DTX device can be used as an activation trigger. All enabled inputs are monitored for activity. When the signal level exceeds the preset Tracking threshold a camera preset can be changed and/or a custom serial command can be sent. For example, a preset can be sent to change the preset of a camera and simultaneously send a command to an Aurora VTX decoder or a video switcher to change which camera is being displayed.

Voice Tracking



Threshold: Sets tracking threshold. When the signal from an enabled channel exceeds the tracking threshold, the camera tracking settings for that channel are executed.

Default Channel: Sets the default channel. When all other channels are below the threshold, the camera tracking settings for the Default Channel are executed.

Reaction Time: Sets the amount of time a signal must be above the threshold before executing a camera tracking setting.

Hold Time: Sets the shortest length of activation before another trigger is executed.
When a person begins speaking the camera can be repositioned to view them after Reaction has expired and will remain on them at least as long as defined in Scroll Time.

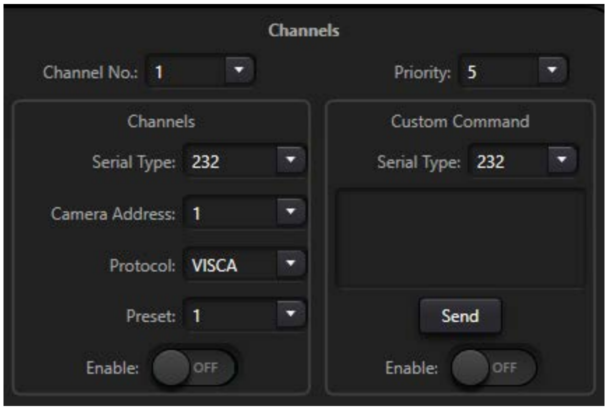
Send Interval: Sets the interval between successive consecutive camera commands.

Send Repeat: The number of consecutive times a camera command is sent.

Enable: Enables Automatic Voice Tracking Camera Control.

Channels

This section is used in conjunction with the Voice Tracking settings. Each input channel in the DSP can be selected and optionally enabled to participate in the Automatic Voice Tracking Camera Control.



Channel No: Selects the current input channel.

Priority: Sets the priority for the selected input channel. If multiple channels exceed the Trigger threshold, the highest priority channel will have its camera control settings executed. A lower number has a higher priority.

Channels

All the following parameters apply to the channel selected above in Channel No. Every channel has its own set of parameters.

Serial Type: Selects either RS-232 or RS-485 for camera communication.

Camera Address: The address of the camera that will be associated with the selected input channel.

Protocol: Communication protocol for the camera associated with the selected input channel.

Preset: Selects which camera preset will be triggered. These are presets on the camera and are defined using the procedure defined by the camera manufacturer.

Enable: This indicates that the selected input channel is participating in Automatic Voice Tracking Camera Control by sending camera presets. This can be used in conjunction with Custom Command.

Custom Command

All the following parameters apply to the channel selected above in Mic No. Every channel has its own set of parameters.

Serial Type: Selects either RS-232 or RS-485 to communicate with an external device.

Command Textbox: Enter the command string to be sent to an external device.

Send: Immediately sends the command when pressed.

Enable: This indicates that the selected input channel is participating in Automatic Voice Tracking Camera Control by sending a custom command. This can be used in conjunction with camera presets.

Save: Saves all the Automatic Voice Tracking Camera Control settings to the connected DTX device.

Live Controls

This section of controls is separate from the Automatic Voice Tracking Camera Control functionality. These parameters allow direct live control of the camera. This can be useful in creating camera presets to be used in Automatic Voice Tracking Camera Control without having to use a third-party application.

Camera Control

Serial Type: Selects either RS-232 or RS-485 for camera communication.

Camera Address: The address of the camera that will be associated with the selected input channel.

Protocol: Communication protocol for the camera associated with the selected input channel.

Camera Speed: Sets the PTZ speed of the camera.

Camera Control Buttons: Provides control capability over the connected camera including pan, tilt, zoom focus and iris settings.

Camera Presets

Preset: Selects the active camera preset.

Call: Send preset change message to the camera.

Save: Saves the current camera settings to the selected preset on the camera.

Clear All: Clears all camera presets.

USB Soundcard

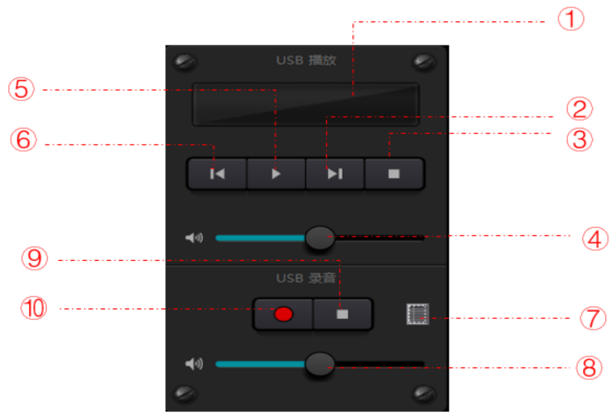
PC or MAC USB Sound Driver

At the most basic level, a USB connection to a DTX DSP can be used as a sound card with a PC or Mac using a generic HID audio. The driver will appear as a “Crestone USB Soundcard” in your computers sound settings. The USB connection shows up in the Matrix as both an input and an output for routing and mixing audio. Connecting a DTX device to a computer via USB requires a USB Type-A to USB Type-A cable which is included in the box. This is the perfect solution for using the DTX as a source for conferencing applications such as Zoom or Teams.

USB Soundcard Module

The USB Soundcard is a system module built into all DTX hardware and can be used for both playback and recording. This module allows playback and recording without the need for external software. When connected to a computer via

USB, audio files stored on the computer can be added to the playlist and played on the connected computer. Similarly, the module can be used to record audio to a storage location on the PC.



① Song playing information, double click to enter playlist
② Next song
③ Pause
④ Song volume adjustment
⑤ Play
⑥ Prev. song
⑦ Sound recording list
⑧ Sound recording volume adjustment
⑨ Stop recording
⑩ Start recording

Appendix A: Device Mirroring

Some DTX models support device mirroring. This provides real-time synchronization of configuration between a Master and Slave DSP. In the event of a catastrophic failure of the Master DSP, the functionality will switch over to the Slave DSP within 3 seconds. This feature is configured in the Device Setting menu.

Appendix B: Control Protocol

External Control

All DTX DSP devices can be controlled through external third-party controllers which support UDP or RS-232. All DSP parameter setting and reporting is supported as well as preset recall.

When using UDP control, the default port is 50000. When using RS-232 control, the default baud rate is 115200, 8 bits, 1 stop bit and no parity. Keep in mind that for stable RS-232 communications, the interval between messages should be greater than 100ms. All port settings can be modified in the ‘Device Setting’ menu to suit your application.

In the event that the external controller expects a reply, turn on ‘Center Control Response’ in the ‘Device Setting’ menu.

Control Protocol

Protocol Header

b[0]	b[1]	b[2]	b[3]	b[4]~b[131]
0xB3	Message Type	Length	Version	Data

Message Start: 0xB3 indicates start of message.

Message Type (b[1]): Defines the type of message to follow.

Value	Function
0x21	Parameter SET
0x22	Parameter GET
0x13	Preset SET
0x74	Other Controls

Length (b[2]): Defines the length of the Data portion of the message.

Version: Provided for legacy purposes. This value should always be 0x01.

Data: The format of the Data bytes will be determined by the Message Type value. These formats are defined below.

Message Type = Parameter SET (0x21)

For Parameter Set operations the Data bytes will be in the following format:

b[4]	b[5]	b[6]	b[7]	b[8]-b[71]
Input/Output	Start Channel	End Channel	Parameter Type	Parameter Value

Input/Output (b[4]): Determines whether to address an input or output channel.

Value	Function
0x02	Input
0x01	Output

Start Channel (b[5]): Zero based start channel number for desired range of channels. For example, channel 1 = 0x00, channel 2 = 0x01, etc.

End Channel (b[6]): Zero based end channel for desired range of channels. If setting a single channel this value will be the same as Start Channel.

Parameter Type (b[7]): Refer to Appendix C: Module Parameter Types for Parameter Type values.

Parameter Value (b[8]-b[71]): Defines parameter value to set. Each value should be represented by two bytes.

Message Type = Parameter GET(0x22)

For Parameter Get operations the Data bytes will be in the same format as Parameter SET operations with the exception that the Parameter Value bytes (9-72) will not be present. The response message from the device will have these bytes filled in with the acquired parameter values.

Message Type = Preset SET(0x13)

For preset SET operations the Data bytes will be in the following format:

b[4]	b[5]	b[6]	b[7]	b[8]~b[71]
Preset	-	-	-	-

Preset (b[4]): Zero based preset number (0x00 – 0x0F).

Message Type = Other Controls(0x74)

Other controls include but are not limited to GPIO, RS-232, RS-485 and Central Control replies.

- For GPIO messages, the Data bytes will be in the following format:

b[4]	b[5]	b[6]	b[7]	b[8]	b[9]	b[10]	b[11]
0x01	0x04	Reserved	Reserved	GPIO Dir	Start GPIO	End GPIO	Value

GPIO Dir (b[8]): Sets input or output.

Value	Function
0x00	Input
0x01	Output

Start GPIO (b[9]): Zero based start channel number for desired range of channels. For example, channel 1 = 0x00, channel 2 = 0x01, etc.

End GPIO (b[10]): Zero based end channel for desired range of channels. If setting a single channel this value will be the same as Start GPIO.

Value (b[11]): When 'GPIO Dir' (b[8]) is set to be an output, Value sets the state of the output GPIO pin. For GPIO inputs, this byte is used as a return field to read the GPIO input pin state.

Value	Function
0x00	Low
0x01	High

- For RS-232 and RS-485 messages, the Data bytes will be in the following format:

b[4]	b[5]	b[6]	b[7]	b[8]~b[131]
Control Type	Message Length	Reserved	Reserved	Message

Control Type (b[4]): Defines the desired serial port to be addressed.

Value	Function
0x02	RS-232
0x03	RS-485

Message Length (b[5]): Length of the message that will be sent via the serial port.

Message (b[8]-b[131]): Message payload to be sent.

- For Control Center messages, the Data bytes will be in the following format:

b[4]	b[5]	b[6]	b[7]	b[8]
0x04	0x01	Reserved	Reserved	Reply Switch

Reply Switch (b[8]): Sets the Command Control Replies switch.

Value	Function
0x00	OFF
0x01	ON

Serial Port to UDP Conversion

DTX DSPs support RS-232 to UDP translation. When a DTX device receives a serial message where the first four bytes are 'UDP:' the message will be translated and sent via the LAN port. The protocol for this function is as follows:

b[0]-b[3] (prefix)	b[4]-b[7]	b[8]-b[9]	b[10]	B[11]	<128bytes
0x3a504455	IP Address	Port	Data Length	Reserved	Data

Prefix (b[0]-b[3]): Four byte prefix that identifies the message as a serial to UDP translation message. Should always be ASCII 'UDP:' or the hex equivalent 0x3A504455.

IP Address (b[4]-b[7]): Destination IP address for the UDP message.

Port (b[8]-b[9]): UDP port on destination device.

Data Length (b[10]): Length of the Data portion of the message, not including prefix, address and port.

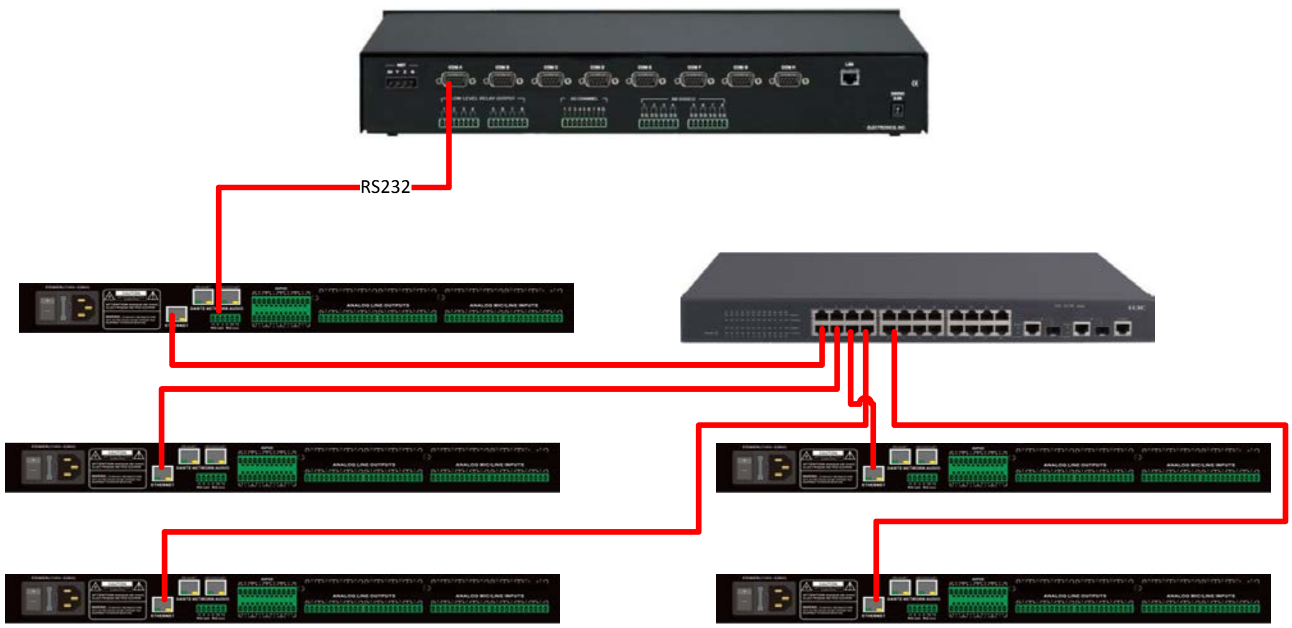
Data: Message payload. Maximum 128 bytes.

For example, when sending a “HELLO DSP” message to a device at 192.168.1.22 port 50000, the protocol would look as follows:

b[0]-b[3] (prefix)	b[4]-b[7]	b[8]-b[9]	b[10]	B[11]	<128bytes
0x3a504455	0x1610A8C0	0xC350	0x09	0x00	0x505344204F4C4C4548

Use Case

When using an external controller which does not have a network port or network access, a DTX device can be used to communicate via UDP over the network. The Controller Host is connected to the serial port of a DTX device, and that device performs the translation into UPD and broadcasts to the network. Other DTX devices or any third-party devices on the network can be controlled via the Controller Host serial port.



Appendix C: Module Parameter Types

Module Name	Parameter Type	Description
Input Source	0x01	Gain
	0x02	Mute
	0x03	Sensitivity
	0x04	Phantom Power Switch
	0x05	Signal Generator Type
	0x06	Signal Generator Frequency
	0x07	Sine Wave Gain Size
	0x08	Channel Name
	0x09	Invert
	0x10	Gain Compensation
	0x11	Link
	0x12	Channel Level
Delay	0x01	Bypass Switch
	0x02	Millisecond
	0x03	Microsecond
Equalizer	0x01	Total Equalizer Switch
	0x02	Child Segment Switch
	0x03	Frequency
	0x04	Gain
	0x05	Q Value
	0x06	Type
Output	0x10	Gain Compensation
	0x11	Link
	0x12	Channel Level
	0x01	Gain
	0x02	Mute
	0x03	Channel Name
	0x04	Invert
	0x05	Sensitivity
	0x06	Gain Compensation
	0x07	Link
	0x08	Channel Level
Expander	0x01	Switch
	0x02	Threshold
	0x03	Ratio
	0x04	Setup Time
	0x05	Release Time
Compressor	0x01	Compressor Switch
	0x02	Compressor Threshold
	0x03	Compressor Ratio
	0x04	Setup Time
	0x05	Recovery Time
	0x06	Gain Compensation
Matrix	0x01	Mixer Switch
	0x02	Mixer Gain
High & Low Pass	0x01	High Pass Switch
	0x02	High Pass Type
	0x03	High Pass Slope
	0x04	High Pass Frequency
	0x05	High Pass Gain
	0x11	Low Pass Switch

Module Name	Parameter Type	Description
	0x12	Low Pass Type
	0x13	Low Pass Slope
	0x14	Low Pass Frequency
	0x15	Low Pass Gain
Auto Mix	0x01	Total Mute
	0x02	Total Gain
	0x03	Slope
	0x04	Response Time
	0x05	Channel Auto Switch
	0x06	Channel Mute
	0x07	Channel Gain
	0x08	Priority
	0x09	Auto Mix Switch
Feedback Control	0x01	Switch
	0x02	Feedback Point Frequency
	0x03	Feedback Point Gain
	0x06	Preset
	0x07	Clear
	0x08	Panic Threshold
	0x09	Feedback
Auto Gain	0x01	Switch
	0x02	Threshold
	0x03	Target Threshold
	0x04	Ratio
	0x05	Setup Time
	0x06	Release Time
Echo Cancellation	0x01	Echo Cancellation Switch
	0x02	Echo Cancellation Mode
Noise Suppression	0x01	Noise Suppression Switch
	0x02	Noise Suppression Mode
System Control	0x01	System Mute
	0x02	System Gain

Aurora Multimedia Corporation

205 Commercial Court | Morganville, NJ 07751

Phone: 732-591-5800 | Fax: 732-591-6801