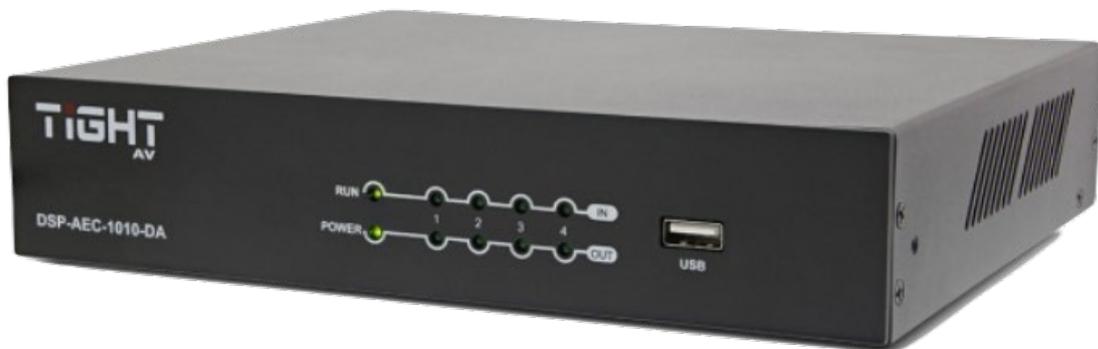# User Manual

DSP-AEC-1010-DA

Audio DSP with AEC
4-in/4-out analog, 4-in/4-out Dante and USB

Version: DSP-AEC-1010-DA_2022V1.2

Table of Contents

Quick Start Guide

1. Install the TiGHT DSP software found on the product page at www.TiGHTAV.com

2. Connect the PC running the software directly to the DSP via network cable, or connect to the same network.

3. Open software and click "Device List" in the top right corner and connect to the DSP.

4. Make sure your computer and TiGHT DSP are connected on the same network.

5. Click on "Connect", it will change to "Disconnect" when your device is online.

You"re all set, enjoy your TiGHTAV product!

Hardware

Safety Instructions

Please read this manual before installing or using the product. Review the following safety precautions.

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

Be careful with electricity:

- **Power Outlet:** To avoid electric shock, make sure that the electrical plug used on the product power cord matches the electrical outlet used to supply power to the TiGHT AV product. Use the power adapter and power connection cables designed for this unit.
- **Power Cord:** Make sure the power cord is routed so that it will not be walked on or pinched by heavy objects.
- **Lightning:** To protect against lightning or when the product is left unattended for a long period of time, unplug it from the power source.

Also follow these precautions:

- **Ventilation:** Do not block the ventilation slots, if any, on the product or place heavy objects on it. Blocking the airflow could cause damage. Arrange the components so that air can circulate freely. Ensure adequate ventilation if the product is placed on a stand or cabinet. Place the product in a well-ventilated place, away from direct sunlight or any heat source.
- **Overheating:** Avoid stacking the DSP-AEC-1010-DA on top of a hot component, such as a power amplifier.
- **Risk of Fire:** Do not place the unit on top of easily combustible materials, such as rugs or fabrics.
- **Proper Connections:** Make 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 when connected to a power supply.
- **Water Exposure:** To reduce the risk of electric shock or fire, never expose the unit to rain or moisture.
- **Cleaning:** Do not use aerosol cleaners or liquid to clean this unit. Always unplug the power source to the device before cleaning the unit.
- **ESD:** Handle this unit with proper ESC care. Failure to do so can result in failure.

Mechanical Data

Space required:
1U (W x D x H: 215 x 162 x 44mm).

At least 70mm should be reserved for the connections on the rear panel. Reserved depth depends on the wire used and the connection mode.

Ventilation:

The recommended highest operating ambient temperature is 30℃ / 86℉.

Make sure that there is no blockage of the venting holes on both sides of the unit. A gap of at least 5cm shall be reserved).

Electrical Property:

The unit can be powered by PoE (802.3af) or with a 12V power adapter.

Universal input power adapter: AC110~240V 5Hz-60Hz, 12V DC / 2A output.

Shipping Weight:

(2 kg)

FCC Statement

This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. It has been tested and found to comply with the limits for a Class A digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a commercial installation.

Operation of this equipment in a residential area is likely to cause interference, in which case the user at their own expense will be required to take whatever measures may be necessary to correct the interference.

Any changes or modifications not expressly approved by the manufacture would void the user's authority to operate the equipment.

Product Introduction

Thanks for choosing the TiGHT AV DSP-AEC-1010-DA. The DSP-AEC-1010-DA is an audio signal processing unit with analog and Dante interface, as well as a USB interface for computer audio. The unit is configured and controlled from a Windows application that presents an easy to use GUI. Powerful processing with eq, dynamics, mixing and matrix combined with advanced acoustic echo cancellation (AEC) makes this unit suitable for a wide range of use cases.

A single network cable connected to a PoE network switch allows for both power, control and Dante audio. Analog audio inputs support both microphone and line level sources. Phantom power is selectable individually for each channel.

USB connection to host computer works as a sound card for easy computer audio connectivity for both playback and recording.

Features

- 4 analog inputs and outputs
- 4 Dante inputs and outputs
- USB for computer audio playback and recording
- Powerful DSP processing with matrix and mixer
- Advanced AEC (Acoustic Echo Cancellation)
- External control from custom software or 3rd party control systems
- External control from configurable PoE wall panel
- Open API for external control via ethernet or RS-232/485

Package List

- 1 x DSP-AEC-1010-DA
- 2 x Mounting Brackets with 6x Mounting Screws
- 1 x 1,5m USB 2.0 cable (Type A to Type A)
- 1 x Screwdriver
- 4 x Rubber Feet

Front Panel Features
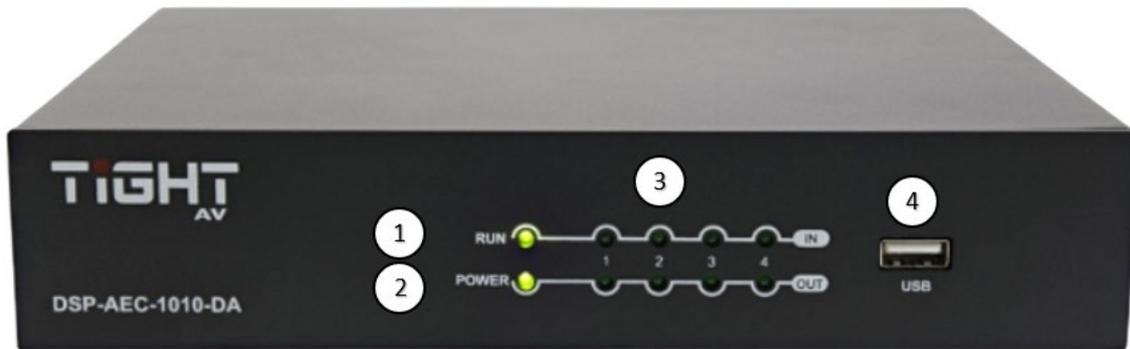
1. **Status LED** – A green LED indicates network connection on the Local Area Network.
2. **Power LED** – A green LED indicates the unit is receiving power.
3. **I/O LEDs** – A green LED indicates the unit receiving analog signal.
4. **USB Audio** – USB soundcard connection for Audio over Type-A USB cable.

Rear Panel Features
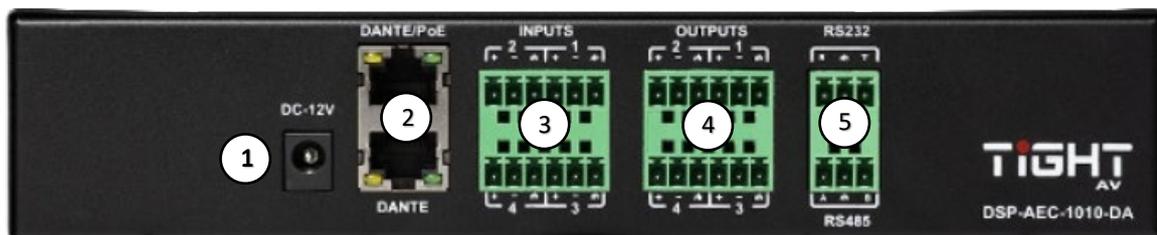
1. **DC Power Input** – 12VDC, 10W, center positive power input. Acts as alternate power source to PoE. For redundancy power, use simultaneous DC Power Input and PoE.

2. **DANTE / PoE Port**- connector, provide up to 8 (4x4) channel Dante network audio. Features the UltimoX4 chip. This interface is also used for communication with PC software and any third-party external controller.

   DANTE Port- Secondary 100 Base-T Ethernet port is Dante only.

3. **Analog Inputs** – Balanced or Unbalanced high impedance inputs. Line or Mic Level supported with +48VDC phantom power.

Signal Label	XLR	TRS
+	Pin 2	Tip
-*	Pin 3	Ring
⏚	Pin 1	Sleeve

	*used with "jumper" when using unbalanced

4. **Analog Outputs –** Balanced or Unbalanced outputs. 24bit DA Conversion.

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

5. **Serial Communication** – RS-232 connections for remote control devices, such as a control panel.

	Signal Label	Signal
	R	RS-232 Receive
	⏚	Ground
	T	RS-232 Transmit

Signal Flow Chart
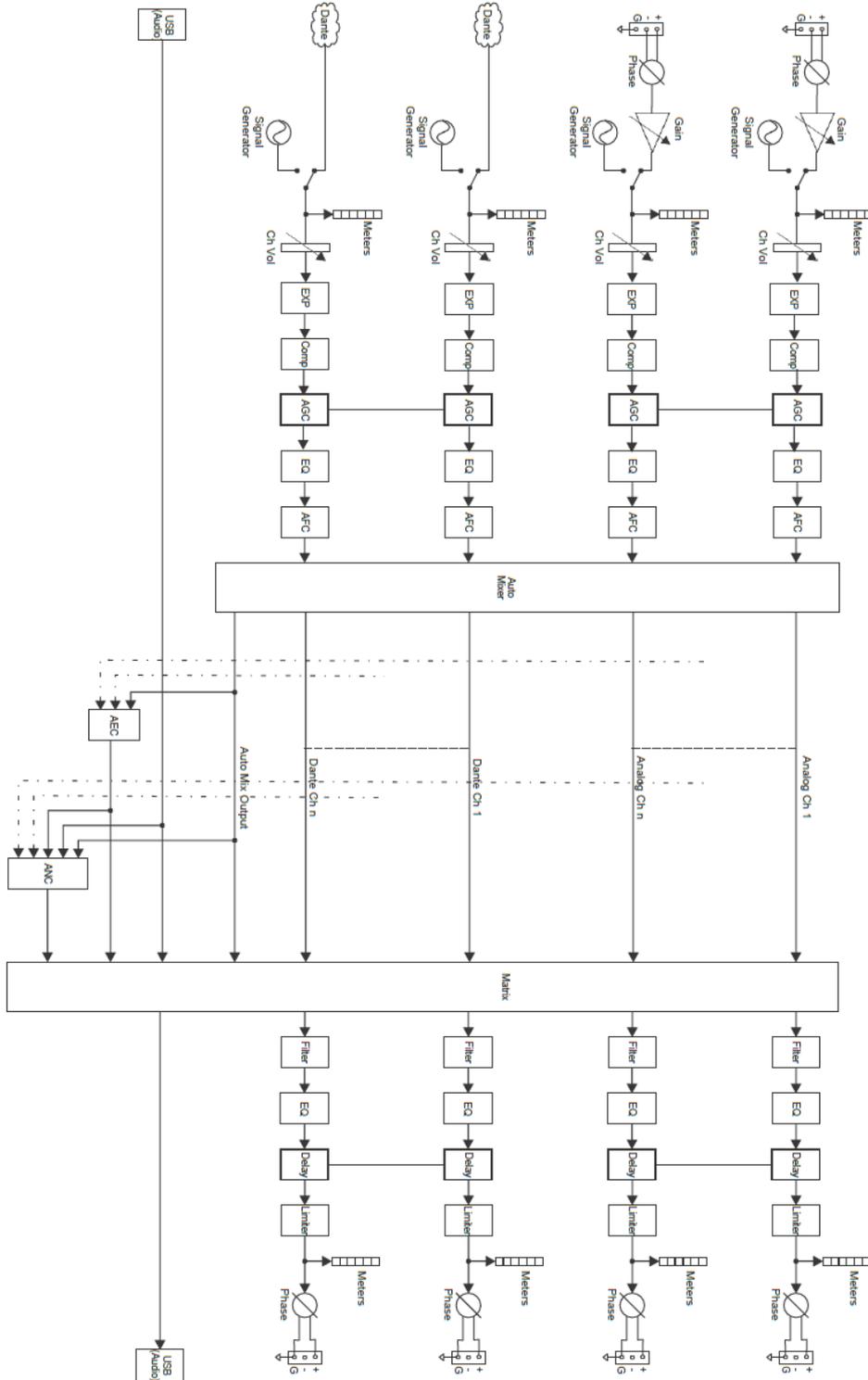
Software

Software Requirements

TiGHT AV DSP software:

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.

Audinate software's:

Refer to Audinate minimum requirements recommendations

Software Installation

1. Download software from the DSP-AEC-1010-DA product page at *https://www.TiGHTAV.com/*. Or open a web browser and type in the IP address of the DSP´s control interface and download the software stored on the device.

2. Double click the downloaded file and install by following the instructions on the screen.

3. After installation is complete, launch the software by clicking the following icon:

4. Wait for the software to launch, may take 1-15s to start.

Software Operation

After the software has opened, click "OK" on your selected device.

To connect to the DSP click on "Device List" and choose your DSP from the list. Then click "Connect".
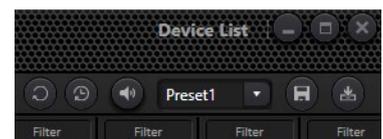
The following window is the Home Tab view of the interface.
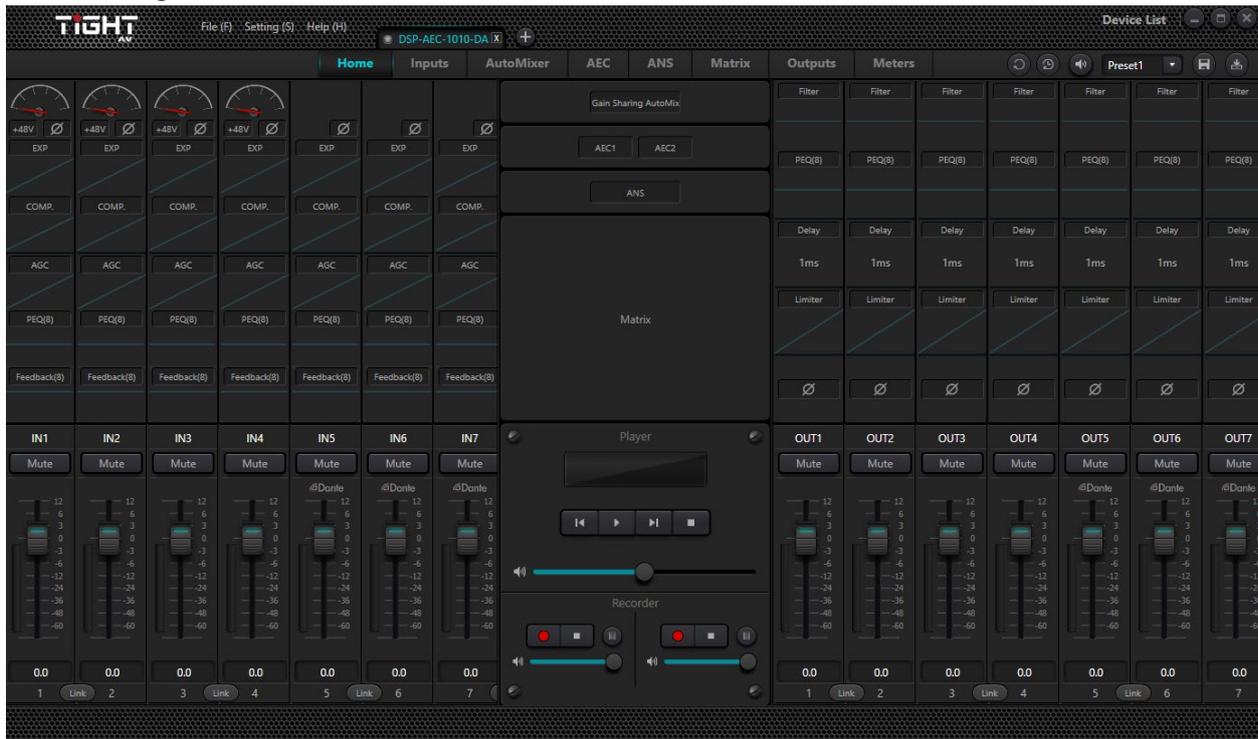
File Menu

Open

The File Menu gives you the option of opening presets from your computer or saving presets to your computer.

Save As

Presets are stored locally on the DSP-AEC-1010-DA. When operating Offline, the configurations can be saved locally to your computer for being uploaded at a later stage to your DSP-AEC-1010-DA.

Settings

For the Device Settings Menu to be accessible the software and DSP-AEC-1010-DA must be connected.

Device Settings

Device Name: Changes the name of the device and the name it will display anywhere the device is addressed. The name is saved into the hardware.

Device IP Address: Changes the IP address of the LAN interface. It doesn´t change the Dante interface IP Address this is done in Audinate's Dante Controller. The unit is set to DHCP IP addressing mode by default. If no DHCP server is present on the network the control interface will use automatic IP addressing protocol (APIPA) in the IP range 169.254.0.1 – 169.254.255.254 with subnet mask 255.255.0.0.

11
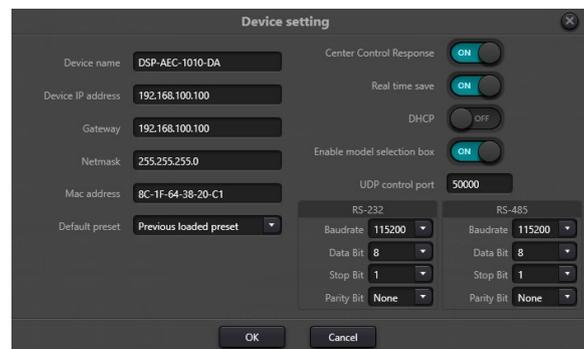
Recommendations if IP-based control systems are being utilized to control the DSP, is to either use static IP address or make DHCP IP reservations in the DHCP server for the DSP control interface.

Gateway: Sets the default gateway for the network.

Netmask: Sets the network mask associated with the desired subnet.

Mac Address: Shows the MAC address of the control interface.
Note: The MAC address for Dante interface is different and can be found by using for example Dante Controller.

Default Preset: Sets the default preset the device loads upon a power cycle event. Can be set to one of the 16 presets or set to last active preset prior to the power cycle.

Central Control Response: Enable this when using an external controller that expects a response to control messages (see Appendix C, Command Structure Examples).

Real Time Save: When enabled, active preset changes are saved in real time. Enable this will ensure the exact sate of the device to be restored upon a power cycle if "Previous Loaded Preset" is selected in the "Default Preset" parameter. If set to "Off", those changes will be lost after the power cycle.

Enable Model Selection Box: When enabled, upon starting the software the model selection window will appear.

UDP Control Port: Sets the UDP control port for remote communication with the DSP-AEC-1010-DA.

OK: All settings done in the Device Setting window are saved to the device and the window will close.

Cancel: Will close the window and no changes are saved.

RS-232/RS-485: Set the baud rate, Data bit, Stop bit and Parity bit for the ports of the device.

Group Settings

The group setting window have separate tabs for input and output groups. 4 groups can be set in each tab. A channel can only participate in one group. In the same group, the channel volume control and mute functions are linked. Other functions are not linked. For pairwise linking of all channel parameters, use the link button below the channel fader.

The relationship between groups and link: The channel participating in a group will not participate in LINK, which means group's priority is higher than LINK. The difference between groups and LINK is that groups can only control channel gain and mute, while LINK includes all parameters in the channel.

Preset Name

In this window, all 16 preset names can be changed in one window.
All changes done here will correspond to other features in the software were Preset Names are available.
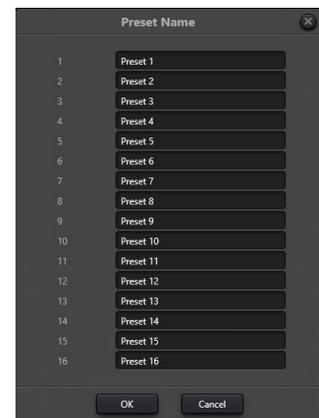
Panel Setting

Panel configuration page for button and OLED panels. Use network cables to connect multiple physical panels with DSP device via the network.
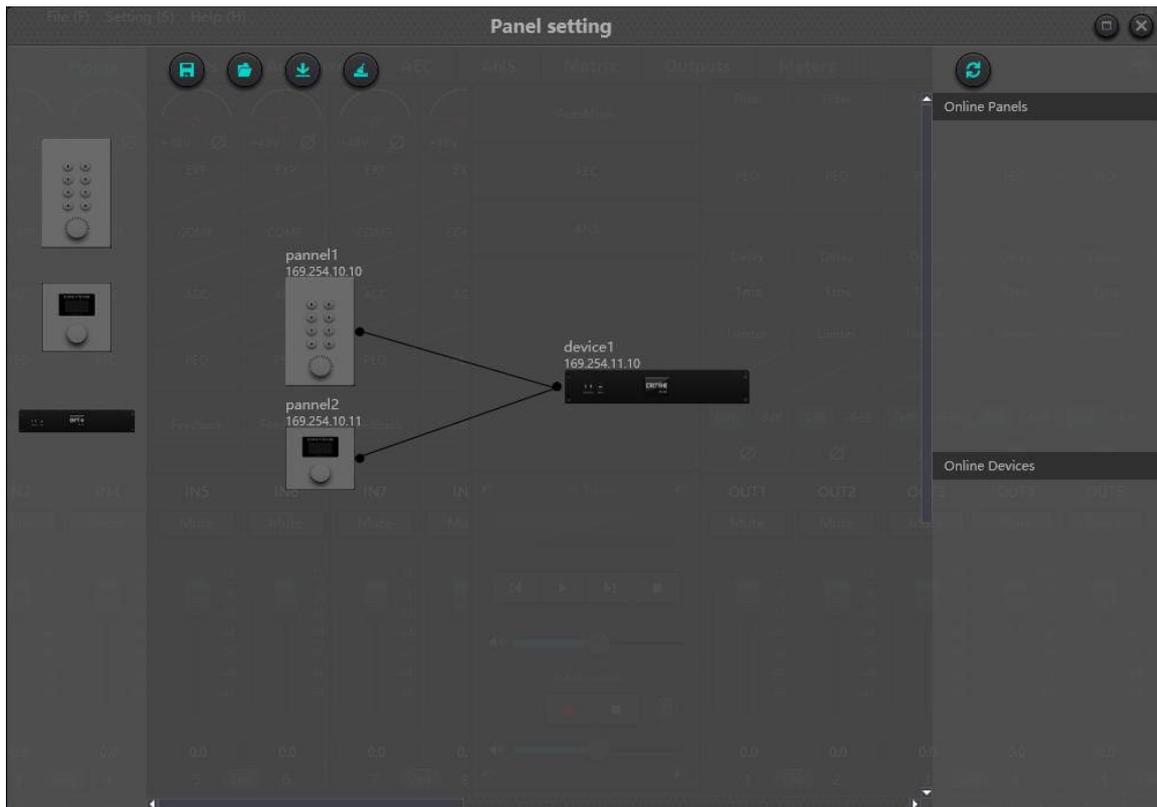
Offline device: Panels can be configured offline for later upload on-site.

Drag offline device from the sidebar to panel design area and double click to edit it.

Note the small circle on both panel and device. Click the circle and draw a line between the units to establish the connection between panel and the DSP device to be controlled.

Double click the panel in design area to enter panel configuration interface.

After finishing configuration, click the toolbar download icon to download the panel configuration to hardware.

OLED panel:

The OLED panel consists of a 1.3" OLED screen and a knob. The panel functions are added in a list, and a single encoder is used for selecting the function and parameter adjustment.

Click "add menu" to show menu selection box. Choose corresponding menu item and configure it. After finishing the setup, click toolbar download icon to download the configuration to panel hardware.

Panel menu navigation:
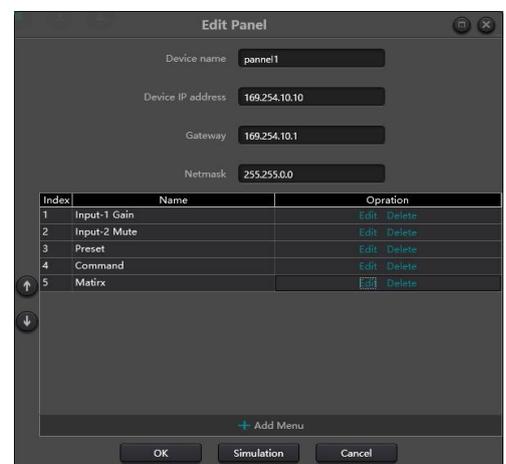
1. Display panel name and IP address on main interface and turn the knob left or right to select function.

2. Press the button on the knob, and the second row on menu interface starts to flash which indicates edit mode.

3. Turn the knob left or right to change value.

4. Press the button on the knob again to exit edit mode and go back to menu mode.

Key Panel:

There are 8 keys and one knob on the key panel. The knob is used to adjust gain and 8 keys can be used for different functions. There are four types of key functions, including volume adjustment, mute, preset and command. Drag an item in function area to appropriate key to program the key.

Similarly, after finishing all programming, users may use the emulation button to check whether the configuration is correct.

Instructions for panel operation indicators

1. Key indicator stays on, which indicates the key is configured with mute function.

2. Key indicator keeps flashing, which indicate the key is configured with gain function. The configured knob is used to adjust gain of the channel. 13 indicators around the knob indicate volume level.

3. A short flash when pressing the key indicates the key is configured with preset or command function.

User Interface

The User Interface menu opens up the User Interface Window. This feature allows the user to fully customize a control panel for hand-held devices such as smartphones, designed to suit the needs of the end user.

The hand-held device must have the app DSP Pro 2 installed in order to use the designed control panel. The app is available on the App Store for apple devices. Android devices is not supported at this stage.
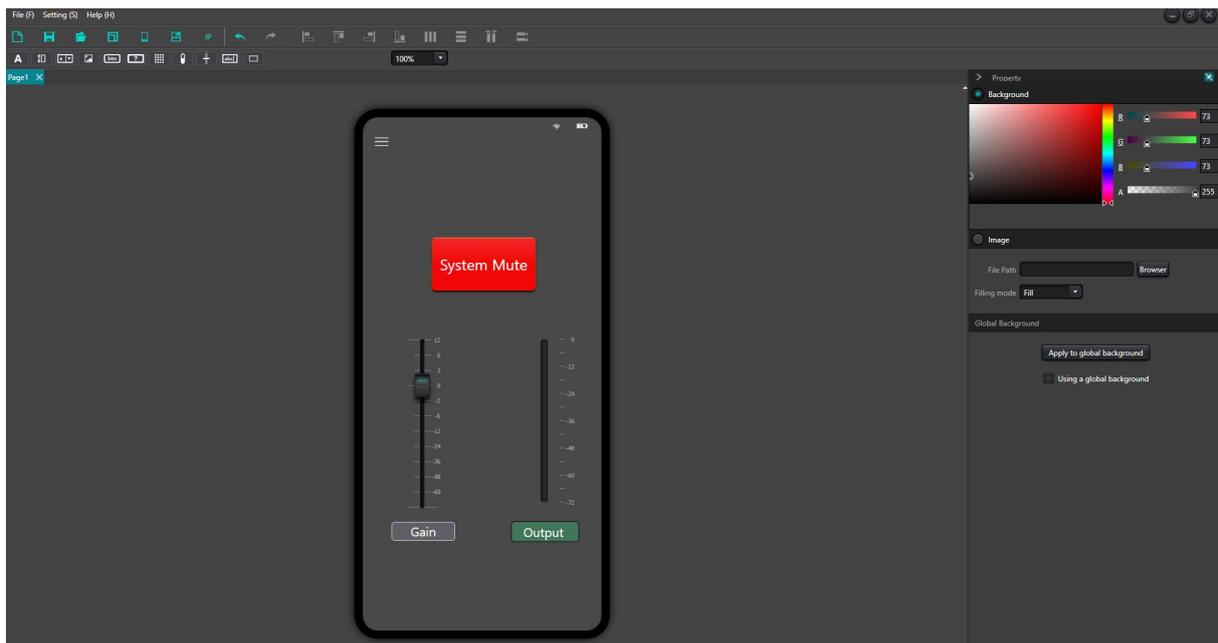
New Page – Opens up a new empty design page.

Save – Saves the user´s current design.

Open – Allows the user to open a previously created panel design.

Template – Allows the user to choose from specific templates based on common hand-held devices.

Sync – This will open up the Sync Window. In this window all discovered devices will appear. Click on your desired device and click sync to connect the control panel and the hand-held device. Using the DSP Pro 2 app the control panel will then be available in the app.

Rotate – This will rotate the chosen template. Note that all current objects in the layout will not be adjusted. Therefore, decide first if the panel will be horizontal or vertically before designed the layout.

Quickly Change IP – This will open up a window for the user to quickly change the source address and destination address, and also the range of pages the changes will apply to.

Designing the Control Panel Layout

By simply drag-and-drop different objects on to the chosen template, each object will be added to the design.

Each object will have its customizable features available on the right side of the window. In this case, the button has been labeled "System Mute", and by opening the "Command" pane the type of command
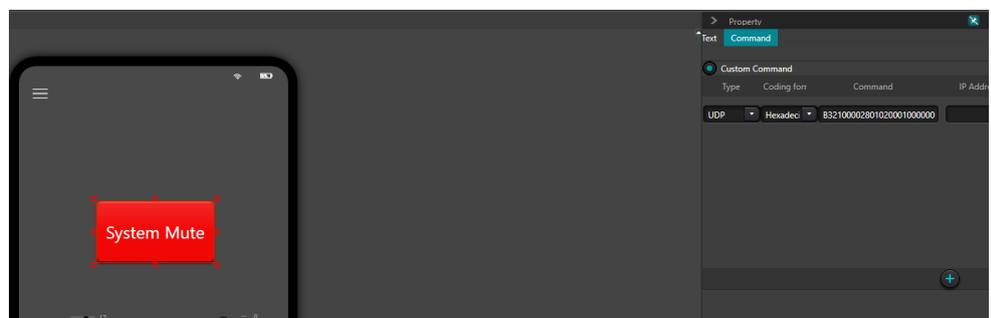
string can be added. Every command is available by using the "Central Control Command" option in the "Help Menu".

For more information and details of the User Interface, please watch the User Interface tutorial video series on www.TiGHTAV.com or the TiGHTAV Youtube Channel.

Help Menu

About
In this window the device Main Version number, the Version number and model type is displayed.
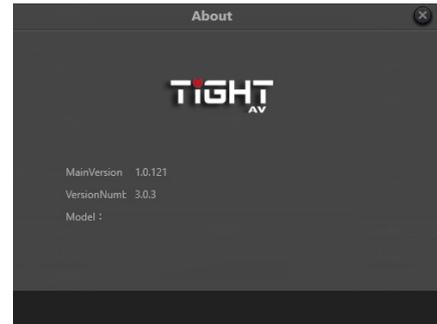
Document
Brings up the device manual pdf.

Central Control Command
Bring up a window showing the last or current change as a command string. When open, click on any parameter and the corresponding command string will be displayed in the window.
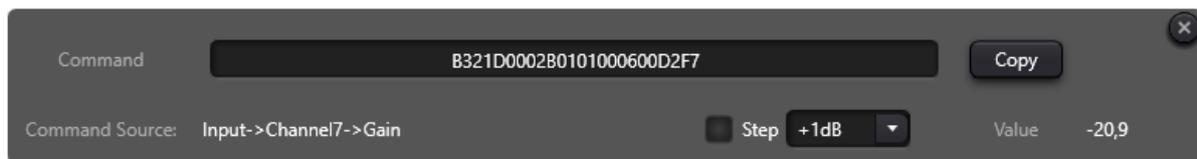
Check for Updates
Click on the option to update your device to the latest version.

Device List
Click on the "Device List" to bring up the pane for your active devices.

Click the ➕ to manually add devices based on IP address.

Click the 🔁 to refresh the discovery of devices on the network.

Device Name: Displays the units device name, this can be changed in the Settings Menus, Device Setting option.

Device IP address: Displays the units IP address, this can be changed in the Settings Menus, Device Settings option.

Version: Displays the units current version.
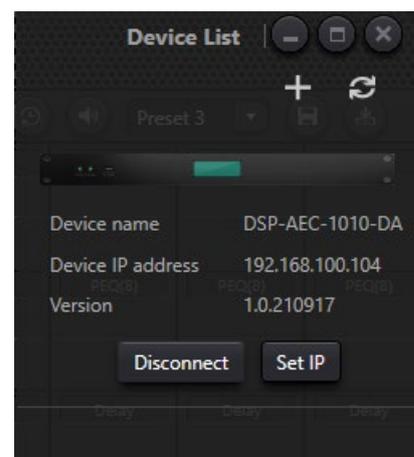
Connect / Disconnect: When the DSP-AEC-1010-DA is on the same network as your computer

running the software the [Connect] button will be available. Click on the button to connect to your
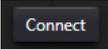
device. As the button changes to [Disconnect] the unit is online and connected.
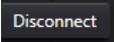
Preset Settings

1. **Reset to default**: Resets the configuration to the chosen default preset set in the Settings Menu, Device Setting window.
2. **Restore Factory Settings:** Reverts the configuration to the factory settings.
3. **System Mute:** When enabled, mutes all audio outputs of the device.
4. **Load Preset Menu:** Allows the user to choose a specific preset to load. Be aware, when clicking on a selected preset the system will load it immediately.
5. **Save Preset:** Allows the user to choose a specific preset slot to save current configuration in.

Home Tab

The Home Tab is the main view of the interface for the selected device. Click the button "Home" to access this view.
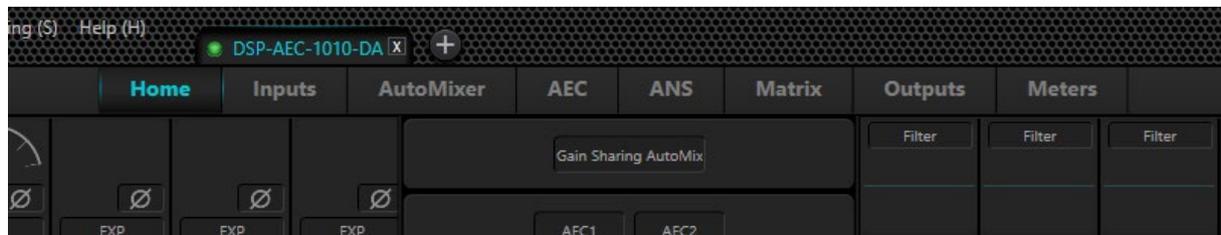
Channel Layout

Both inputs and output channels are numbered 1 to 8. The first four channels are the analog inputs and outputs. The remaining channels 5-8, are Dante channels.

If the user is not in full screen mode, some channels may be hidden from view.

Navigate through the channels by hovering the mouse over the channels and using the scroll wheel up or down.

Selecting Channels

Click on the channel label, fader or number. This will select the active channel, the channel will get a teal border around the fader to indicate it is now selected.

Set Channel Name

Channel names can be set by double clicking the name plate of the channel above the fader. Type the name and hit enter to save the name.

Set Channel Gain

The Channel Gain can be adjusted by either moving the fader or by typing the specific value in the box directly under the fader.
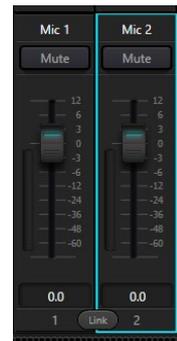
Link Button

Toggle Channel Link by clicking the "Link" button. Each pair of channels can be directly linked to operate accordingly to the adjacent channels settings.

Setting Minimum and Maximum Gain

The default minimum and maximum gain levels on the input and output faders are set to their limits of -72dB to +12dB. By right clicking a channel fader a context menu will show which allows the user to reduce these limits.
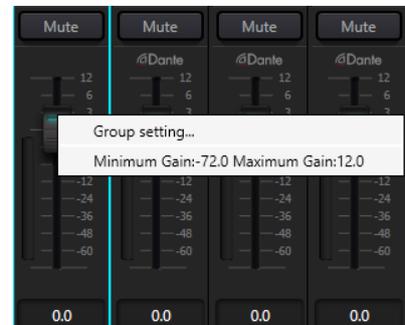
Accessing Parameters

To access channel parameters there are multiple options.

Right Click: By right clicking on a module on channel the selected parameter or module will open in a window while the main interface stays on the home tab.

Double Click: By double clicking on a module on either an input or output channel, the selected module will appear as the active window.

Module Tabs: Modules can be accessed by clicking the specific module tab next to the Home Tab.

Module Tabs

There are multiple Module Tabs showing on the interface. These are Home, Inputs, Automixer, AEC, ANS, Matrix, Outputs, Meters.

Inputs / Outputs

The input /outputs tabs displays the fader section across the bottom. Above the faders are all the modules available for the channel. Scrolling up or down will navigate you through the modules.

AutoMixer

In the AutoMixer tab all input channels are shown with a respective fader control and mute option and routing option. Clicking on "Automixer" for a certain channel will route the audio through the Automixer.

The Automixer is a multichannel mixer which automatically lowers the level of non-active audio signal sources, such as a microphone, while simultaneously adjusting the level of the active sources accordingly to maintain a desired target gain. This eliminates the need for live mixing of audio sources.

Gain sharing allows for the most natural sounding mix of multiple microphones. A Gain Sharing Automixer uses mathematics to reduce the individual gain of a microphone as more microphones are added. For every doubling of the number of open mics, the Automixer reduces the overall gain by 3dB.

Gain: Determines the desired target gain for the output of the Auto Mix.

Slope: Sets the aggression of the gain sharing- Low Slope settings cause the module to act just like a conventional mixer. High slope causes the inputs to attenuate more deeply for inactive sources, very much like a gate module. A value of 2.0 give a natural sounding effect in most applications.

Response: Adjusts the speed at which the AutoMix operates. A quicker response time can be needed if different speakers are speaking in rapid succession, but a response value that is too fast could cause unwanted effects on the overall volume, such as a "pumping" sounding mix. Also, a response time set too slow could cut away the beginning of certain words. The Gain Sharing Algorithm provides for a quicker on time than off to prevent this effect. A Response value between 100ms and 1000ms will provide the most natural effect.

Mute: Mutes the output of the AutoMixer.

Fader/Gain: Adjusts the channel gain. Channel gain is <u>after</u> the input to the AutoMix engine. Channel level can still affect the gain adjustment of other channels even if the fader is all the way down.
Priority: Adjusts the level of priority for the selected channel. This value feeds into the AutoMix engine and may influence the automatic gain adjustments decisions.
AutoMixer: Enables the selected channel to participate in the AutoMix.
Mute: Mute the selected channel. The Mute function on the channel is after the input to the AutoMix engines. A muted channel can still affect the gain adjustments of other channel in the AutoMix.

AEC

The Audio Echo Cancellation tab allows the user to apply acoustic echo cancellation. For audio conferencing when simultaneous or full-duplex transmission of speech is necessary, echo cancellation is an important tool.
Local Input: Selects the audio source or sources that contains the desired audio signal as well as unwanted echoes from the remote signal.
Remote Input: Selects the audio source or sources for the audio signal that will be subtracted from the Local Input.
NLP: Non Linear Processing defines 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 length of the echo is greater than the NLP value the echo will still be heard in the program signal. The following presets corresponds to the milliseconds in the columns below:

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

ANS

The Automatic Noise Suppression can remove noise from the audio when detected. Because of an algorithm that distinguishes human voice from other sounds, such as noise, it can detect the noise and subtract it from the audio signal.
Input Selection: Determines what source or sources the ANS will be applied to.
Level: Determines the attenuation of the non-human sounds or noise.
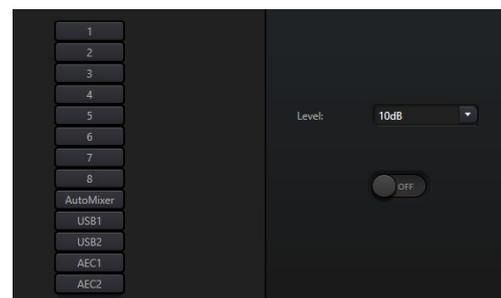
Matrix

The Matrix tab allows you to route inputs and modules to outputs on the device.
When clicking on an intersection, the box will turn into a teal color indicating that those two channels are now connected.
By right clicking on the teal intersection a small fader will appear. This gives the user the option to adjust the amount of input signal going through that specific intersection.
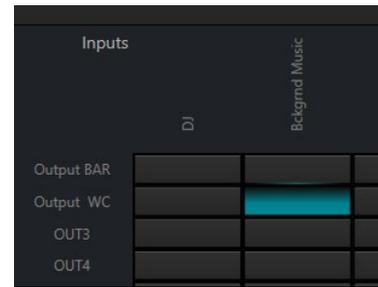
Meters

The Meters tab shows the input and output levels when audio is being routed through the respective channels.
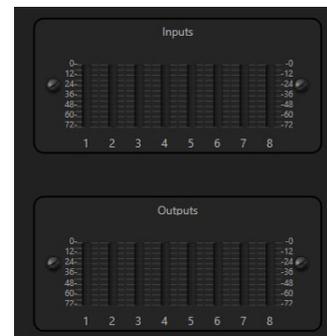
Channel Modules

The DSP-AEC-1010-DA software is designed to be flexible and includes a wide range of modules, each providing a unique function.

Input Source

The Input Source module is a fixed module on every input channel. From here the user can set the input sensitivity, enable 48v phantom power, phase shift or mute.
Below is also a Signal Generator with either sinewave, white or pink noise as sound source.

Expander

The Expander module is a helpful dynamic processing tool to increase the dynamic range of an audio signal. An expander engages when the signal drops below a defined threshold. An expander can be seen as a Noise Gate but with more nuanced settings.

Threshold: The expander engages the signal when the signal falls below the threshold value.

Ratio: The gain curve, or amount of expansion, is defined by the ratio value.

Attack: The attack determines how fast the expander engages as the audio crosses the threshold.

Release: The release determines how fast the expander recovers after the audio returns above the threshold limit.
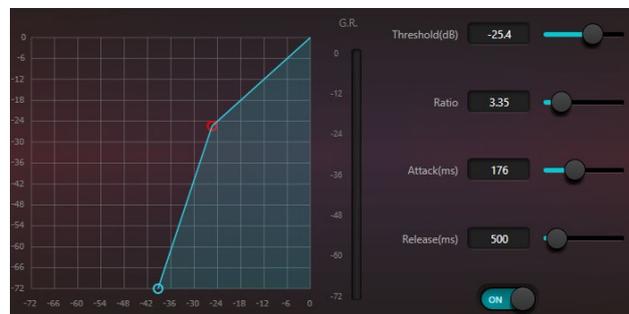
Compressor

A compressor reduces the amount of dynamic range in an audio signal. Dynamic range is the difference between the loudest and the quietest part of a signal. A compressor reduces this range by attenuating the louder parts of the signal and boosting the quieter parts of the signal.

Threshold: If a signal crosses above the threshold of the compressor the compressor will be engaged.

Ratio: The amount of attenuation applied to the audio signal when it crosses the threshold is set by the ratio value.

Attack: Defines the time it takes for the compressor to attenuate a signal that crosses the threshold.

Release: Defines the time it takes for the compressor to stop attenuating when the signal falls below the threshold.

G.R: Displays the amount of gain reduction applied to the signal as it passes above the threshold.

Output: The fader sets the amount of gain applied to the signal after the compressor. As a signal is attenuated it may be quieter, adjust the output gain to compensate for this loss of level in the audio signal.
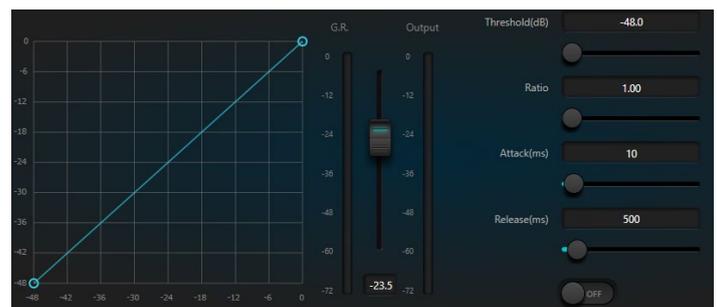
AGC

Automatic Gain Control maintains the dynamic range while automatically adjusting the audio signal level. The goal of an AGC is to normalize the signal level while maintaining dynamic range. When configured correctly, a silent speaking person will sound the same as a loud speaking person without having to manually adjust the microphone gain or mixer level.
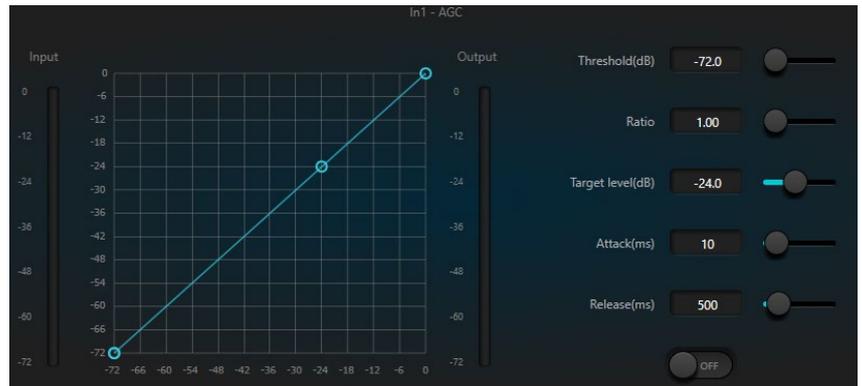
Threshold: When the audio signal level is below the threshold, the input-to-output ratio is 1:1. If the signal level is above the threshold level, the input-to-output ratio changes according to the assigned ratio value.

Ratio: Adjusts the desired ration between input/output signal.

Target level: Defines the desired output signal level.

Attack: Defines the time it takes for the AGC to engage as the signal crosses above the threshold.

Release: Defines the time it takes for the AGC to disengage as the signal falls below the threshold.

PEQ

A Parametric Equalizer is an adjustable filter to attenuate or boost a signal frequency content of an audio signal. The PEQ allows for multiple frequency bands, or filters, with adjustable parameters.

Type: Parametric Equalizer is the default option. Lowshelf adds a filter controlling frequency below a certain point, highshelf does the opposite. Lowpass adds a cut off point where all frequency content above the cut off point will be removed. The highpass does the opposite to a lowpass filter.

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

Gain (dB): Determines the amount of attenuation or boosting applied at the center frequency.

Q: Represents the bandwidth of the filter. This sets how wide or narrow the filter will affect frequencies surrounding the center frequency.

Q/OCT: Changes the units from Q-value to octaves. Measuring bandwidth in octaves can provide a more musical result.

Adaptive Feedback Suppression

The Feedback Suppression is a module on each input channel. The module detects fundamentals in the audio signal that causes feedback and adds notch filters to respected parts of the signal that causes feedback.

Panic Threshold: Any signal crossing above the Panic Threshold will be detected as feedback. When a signal crosses above any of the following will occur to attenuate the feedback:

- The gain of the signal will be attenuated to prevent the continuous feedback
- The module will limit the output signal to prevent continuous feedback
- The amount of sensitivity on each filter will be increased to provide faster detection

When the level of the audio signal falls below the threshold the settings will return to their pre-feedback event state.

Feedback Threshold: Any signal crossing below the Feedback Threshold will not be detected as feedback.

Filter Depth: Adjusts the maximum attenuation of a single filter.

Preset: Give the user preconfigured settings for common scenarios and venues.

Octave Setting: Determines the steepness of the filter notch.

Clear All: In a feedback event, various notch filters will be applied. To remove these filters click "Clear All". Note that turning the module **ON/OFF** does not remove the filters.

Bandwidth: Each filter behavior can be adjusted, either as a dynamically moving filter, a fixed filter or a manually adjusted filter.

Delay

The delay module applies delay to a selected output channel. Its main purpose is to adjust time alignment.

Milliseconds, Meters and Feet: These parameters are interlinked, the display the same delay time measured in three distinct units. Adjusting a value in any one of the parameters will automatically be converted to the other units of measure and filled in.

Delay Slider: Displays a graphical way to enter the delay value.
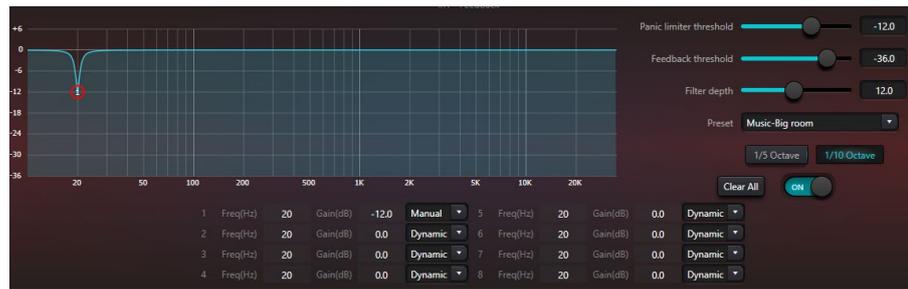
USB Soundcard

PC or MAC USB Sound Driver

By connecting a USB type A cable from the DSP to a computer, the DSP-AEC-1010 DA can be used as a sound card with a PC or MAC using generic HID audio.

When connected to a computer via USB, the driver will appear as a "Cretone USB Soundcard" in your computer settings. The USB Soundcard is built into the hardware and can be used for playback or recording. When connected to a computer, all recordings done with the DSP are stored locally on the computer, and playlists added to the player for playback are stored in the computer as well.
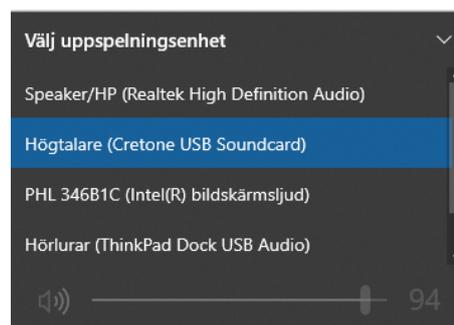
Dante Setting

Use the Dante Controller software for routing and configuration of the Dante inputs and outputs.

The TiGHT AV DSP can interface with other Dante devices operating at 48kHz sampling frequency. Click the crosspoint between transmitting and receiving device to establish a subscription of the audio stream on the network.

Please refer to the Audinate Dante Controller user guide for more information

https://dev.audinate.com/GA/dante-controller/userguide/webhelp/content/front_page.htm

Control

External Control Programmer

External control programmer supports UDP and RS232 and controls protocols covering all controls parameters of processor, including parameter controls, parameter acquisition and preset calling.

When UDP controls are used, the default port is 50000. Ports can be set in "Device Setting" via computer software.

When using RS232 controls are used, the default baud rate is 115200, digit bits is 8, and stop bit is 1, no parity bit. Similarly, they can be set in "Device Setting". **The interval between messages shall keep more than 100mS for RS232 sending.**
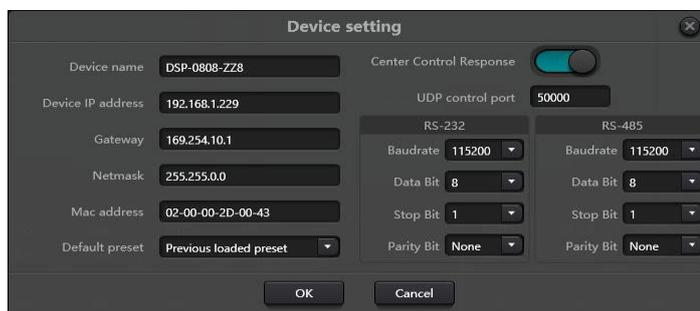
If central controller needs a response, please turn on central control reply switch in "Device Setting".

Control Protocol

Because of historical reasons there is two versions of control protocols V1 and V2. The latest control protocol ,V2, adopts variable-length and is fully compatible with the legacy fixed-length V1 control

protocol. In the V1 and V2 protocols, the fourth byte is used to distinguish between versions. 0x00- indicates V1 version (previous version) and 0x01- indicates V2 version (current protocol version).

The difference between V1 and V2 is that V1 may control all processing module parameters, but one command can only control one parameter. If a parameter is needed to control continuous multiple channels, then V2 version should be used. Both versions are active and available for usage.

V1 Protocol:

Structure

Byte 1 Fixed	Byte 2	Byte 3 Fixed	Byte 4	Byte 5~12
0xB3	Message Type: SET = 0x21 GET = 0x22 PRESET = 0x13	0xAB	Version No: V1 = 0x00 V2 = 0x01	Data

Data Bytes 5-12: All using 2 bytes in little endian format.

Byte 5~6	Byte 7~8	Byte 9~10	Byte 11~12
Module ID	Parameter Type	Parameter 1	Parameter 2

Byte 1 - Fixed start: 0xB3.

Byte 2 - Message type: SET= 0x21, GET =0x22 and PRESET =0x13.

Byte 3 - Fixed to 0xAB.

Byte 4 - Version number: V1 = 0x00 and V2 =0x01.

Byte 5-6 - Module ID: See Appendix A: Module ID.

Byte 7-8 - Parameter type: Appendix B: Module Parameters.

Byte 9-10 - Parameter 1.

Byte 11-12 - Parameter 2.

Note: "Central Control Command" can acquire V1 version code through software menu bar of PC. For customized development, please use this protocol.

V2 Protocol:

Structure

Byte 1 Fixed	Byte 2	Byte 3 Fixed	Byte 4	Byte 5~72
0xB3	Message Type:	0xAB	Version No: V1 = 0x00 V2 = 0x01	Data

Byte 1 - Start of command: 0xB3

Byte 2 - Message types: There are various message types including:

0x21: SET.

0x22: GET, no parameter values should be filled in.

0x13: PRESET.

0x74: Other controls.

0x6E: Dante routing.

Byte 3 – Fixed: 0xAB

Byte 4 - Version number: V1 = 0x00 and V2 = 0x01.

Data bytes 5-72

Byte 5	Byte 6	Byte 7	Byte 8	Byte 9~72
Input/Output Output = 0x01 Input = 0x02	Start Channel	End Channel	Parameter Type	Parameter Value

Byte 5 – Set control of input or output channel: 0x01 for output and 0x02 for input channels.

Byte 6-7 - Indicate start and end channel numbers: Channel count starts at 0 (Channel 1 = 0).

Byte 8 - Parameter type: Please refer to Appendix B.

Byte 9-72 - Fill in parameter values of start to end channels. Each parameter value shall take two bytes.

Preset Structure (V1 and V2 protocol)

Byte 1 Fixed	Byte 2	Byte 3 Fixed	Byte 4	Byte 5~12
0xB3	Message Type: PRESET = 0x13	0xAB	Version No: V1 = 0x00 V2 = 0x01	Preset No 0-15

UDP to RS232/RS485 (V2 protocol only)

Byte 1 Fixed	Byte 2	Byte 3 Fixed	Byte 4	Byte 5~72
0xB3	Message Type: Other = 0x74	0xAB	Version No: V2 = 0x01	Data

Byte 5	Byte 6	Byte 7	Byte 8	Byte 9-132
Control Type: RS232 = 0x02 RS485 =0x03	Data Length	Reserved 0x00	Reserved 0x00	Data Bytes

Byte 5 - Control Type: 0x02 = RS232, and 0x03= RS485.

Byte 6 - The data length of byte 6 refers to the data length in bytes that shall be sent via RS232/485.

Byte 7-8 – Reserved for future use. Fill out with 0x00.

Byte 9-132 - Command data to be sent via RS232/485.

RS232/RS485 to UDP (V2 protocol only)

DSP devices support RS232 translating into UDP. The protocol formats are as follows.

Byte 1 Fixed	Byte 2	Byte 3 Fixed	Byte 4	Byte 5~128
0xB3	Message Type: Other = 0x74	0xAB	Version No: V2 = 0x01	Data

Byte 5~8 Prefix	Byte 9~12	Byte 13~14	Byte 15	Byte 16	Byte 17~128
0x08000000	IP address of receiving device	Port Number	Data length	Reserved	Data

Application scenario: The function can be applied in scenarios when a central controller have no network control ability. One of the DSPs can then be utilized as a RS232/RS485 to UDP translator and control multiple DSP or third-party devices over IP.
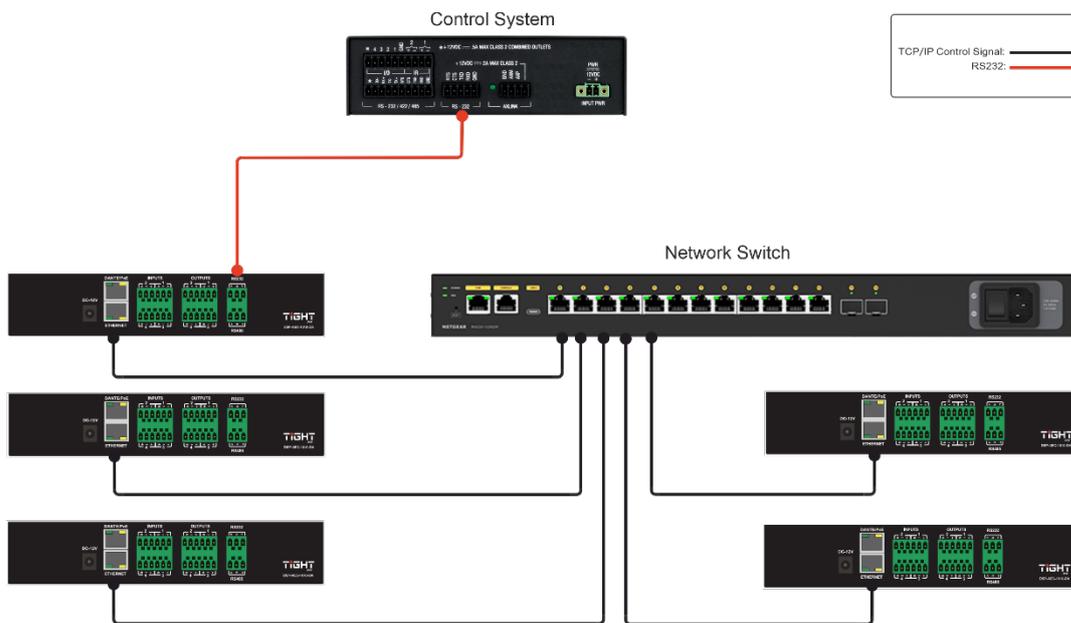
Note: Baudrate for RS232 and RS485 is set globally by software under "Device Settings".

Central Control Replies control ON/OFF (V2 protocol only):

Byte 1 Fixed	Byte 2	Byte 3 Fixed	Byte 4	Byte 5~9
0xB3	Message Type: Other = 0x74	0xAB	Version No: V2 = 0x01	Data

Byte 5	Byte 6	Byte 7	Byte 8	Byte 9
Control Type: 0x04	Data Length 0x01	Reserved 0x00	Reserved 0x00	Data Bytes ON = 0x01 OFF = 0x00

Feedback to a SET command or response to GET command (V1 and V2)

Response to a SET command will mirror the sent command with the exception of Byte 4 (Version No) where the response will have "E0" for V1 and "E1" for V2.

Response to a GET command will have Byte 4 (Version No) "E0" for V1 and "E1" for V2 and the Parameters filled in with the requested information.

Appendix A: Module ID

Module:	ID
Input source	299
Input Channel 1-32 Expander	1~32
Input Channel 1-32 Compressor	33~64
Input Channel 1-32 Auto Gain Compensation (AGC)	65~95
Input Channel 1-32 Equalizer	97~128
Input Channel 1-32 Feedback Inhibition	129~160
AutoMixer	161
Aucoustic Echo Cancellation AEC 1	163
Aucoustic Echo Cancellation AEC 2	301
AEC Matrix	162
Automatic Noise Suppression (ANS)	165
ANS Channel Selector	164
Matrix	166
Output	295
System Control	296
Output Channel 1-32 Filter	167~198
Output Channel 1-32 Equalizer	199~230
Output Channel 1-32 Delay	231~262
Output Channel 1-32 Limiter	263~294

Appendix B: Module Parameters

Module:	Parameter Type	Description	Parameter 1 (Value)	Parameter 2 (Value)
Input	0x1	Gain	Input channel 1 ~ 8 => Values: 0 ~ 7	(-)72dB to 12dB in steps of 0,01dB => Values: -7200 ~ 1200
	0x2	Mute	Input channel 1 ~ 8 => Values: 0 ~ 7	ON = 1 OFF=0
	0x3	Sensitivity	Input channel 1 ~ 8 => Values: 0 ~ 7	0dB to 48dB in 6dB steps => Values: 0~8
	0x4	Phantom Power Switch	Input channel 1 ~ 8 => Values: 0 ~ 7	ON = 1 OFF=0
	0x5	Signal Generator	Input channel 1 ~ 8 => Values: 0 ~ 7	OFF (Analog signal)=0 Sine =1 White=2 Pink=3
	0x6	Signal Generator Frequency	Input channel 1 ~ 8 => Values: 0 ~ 7	20Hz to 20 000Hz in steps of 1Hz => Values 20 ~ 20 000
	0x7	Level (dBFS)	Input channel 1 ~ 8 => Values: 0 ~ 7	(-)72dBFS to 0dBFS in steps of 0,01dBFS => Values -7200 ~ 0
	0x8	Channel Name	Input channel 1 ~ 8 => Values: 0 ~ 7	SET: Only supporting 2 bytes GET: Fill out command past 12th byte with 00 to get full name
	0x9	Invert Phase	Input channel 1 ~ 8 => Values: 0 ~ 7	ON = 1 OFF= 0
	0x11	Link	Link CH1 and CH2 = 1 Link CH3 and CH4 = 3 Link CH5 and CH6 = 5 Link CH7 and CH8 = 7	ON = 1 OFF= 0
	0x12	Channel Level (Meter)		Input channel 1 ~ 8 => Values: 0 ~ 7
Delay	0x1	Bypass Switch	ON = 0 OFF=1	
	0x2	In Milliseconds	0ms to 1200ms in steps of 1ms => Values 0 ~ 1200	
Equalizer	0x1	Total Equalizer Switch	ON = 0 OFF=1	

	0x2	Child Segment Switch	8 Bands => Values: 0 ~ 7	ON = 0 OFF=1
	0x3	Frequency	8 Bands => Values: 0 ~ 7	20Hz to 20 000Hz in steps of 1Hz => Values 20 ~ 20 000
	0x4	Gain	8 Bands => Values: 0 ~ 7	(-)24dB to 18dB in steps of 0,01dB => Values: -2400 ~ 1800
	0x5	Q Value	8 Bands => Values: 0 ~ 7	0,02 to 50 in steps of 0,01 => Values 2 ~ 5000
	0x6	Type	8 Bands => Values: 0 ~ 7	Low pass = 0 High pass = 1 Parametric = 6 Lowshelf = 7 Highshelf = 8
Output	0x1	Gain	Output channel 1 ~ 8 => Values: 0 ~ 7	(-)72dB to 12dB in steps of 0,01dB => Values: -7200 ~ 1200
	0x2	Mute	Output channel 1 ~ 8 => Values: 0 ~ 7	ON = 1 OFF= 0
	0x3	Channel Name	Output channel 1 ~ 8 => Values: 0 ~ 7	SET: Only supporting 2 bytes GET: Fill out command past 12th byte with 00 to get full name
	0x4	Invert Phase	Output channel 1 ~ 8 => Values: 0 ~ 7	ON = 1 OFF= 0
	0x7	Link	Link CH1 and CH2 = 1 Link CH3 and CH4 = 3 Link CH5 and CH6 = 5 Link CH7 and CH8 = 7	ON = 1 OFF= 0
	0x8	Channel Level (Meter)	Output channel 1 ~ 8 => Values: 0 ~ 7	Only available as GET command (-)72dB to 0dB in steps of 0,01dB => Values: -7200 ~ 0
Expander	0x1	Switch	ON = 0 OFF=1	
	0x2	Threshold (dB)	(-)72dB to 0dB in steps of 0,01dB => Values -7200 ~ 0	
	0x3	Ratio	1dB to 20dB in steps of 0,01dB => Values 100 ~ 2000	

	0x4	Attack Time (ms)	1ms to 500ms in steps of 1ms => Values 1 ~ 500	
	0x5	Release Time (ms)	1ms to 10 000ms in steps of 1ms => Values 1 ~ 10 000	
Compressor	0x1	Compressor Switch	ON = 0 OFF=1	
	0x2	Compressor Threshold (dB)	(-)48dB to 0dB in steps of 0,01dB => Values -4800 ~ 0	
	0x3	Compressor Ratio	1dB to 20dB in steps of 0,01dB => Values 100 ~ 2000	
	0x4	Attack Time (ms)	1ms to 500ms in steps of 1ms => Values 1 ~ 500	
	0x5	Release Time (ms)	1ms to 10 000ms in steps of 1ms => Values 1 ~ 10 000	
	0x6	Gain Compensation (dB)	(-)72dB to 6dB in steps of 0,01dB => Values -7200 ~ 600	
Matrix	0x1	Matrix Switch	First byte input ch 0 ~ 13 Second byte output ch 0 ~ 9 Ex IN1 OUT4=> 00 03	ON = 1 OFF= 0
	0x2	Matrix Gain	First byte input ch 0 ~ 13 Second byte output ch 0 ~ 9 Ex IN1 OUT4=> 00 03	(-)72dB to 12dB in steps of 0,01dB => Values: -7200 ~ 1200
Filter	0x1	High Pass Switch	ON = 1 OFF= 0	
	0x2	High Pass Type	Bessel = 0 Butterworth = 1 Likwitz-Riley = 2	
	0x3	High Pass Slope	Values: 6, 12, 18, 24, 30, 36, 42 and 48	
	0x4	High Pass Frequency	20Hz to 20 000Hz in steps of 1Hz => Values: 20 ~ 20 000	
	0x5	High Pass Gain	(-)15dB to 15dB in steps of 0,01dB => Values: -1500 ~ 1500	

	0x11	Low Pass Switch	ON = 1OFF= 0	
	0x12	Low Pass Type	Bessel = 0 Butterworth = 1 Likwitz-Riley = 2	
	0x13	Low Pass Slope	Values: 6, 12, 18, 24, 30, 36, 42 and 48	
	0x14	Low Pass Frequency	20Hz to 20 000Hz in steps of 1Hz => Values: 20 ~ 20 000	
	0x15	Low Pass Gain	(-)15dB to 15dB in steps of 0,01dB => Values: -1500 ~ 1500	
Auto Mixer	0x1	Total Mute	ON = 1 OFF= 0	
	0x2	Total Gain (dB)	(-)72dB to 12dB in steps of 0,01dB => Values: -7200 ~ 1200	
	0x3	Slope	1 to 3 in 0,01 steps => Value: 100 ~ 300	
	0x4	Response Time	5ms to 5 000ms in steps of 1ms => Values: 5 ~ 5 000	
	0x5	Channel Auto Mixer	Channel 1 ~ 8 => Values: 0 ~ 7	ON = 1 OFF= 0
	0x6	Channel Mute	Channel 1 ~ 8 => Values: 0 ~ 7	ON = 1 OFF= 0
	0x7	Channel Gain	Channel 1 ~ 8 => Values: 0 ~ 7	(-)72dB to 12dB in steps of 0,01dB => Values: -7200 ~ 1200
	0x8	Priority	Channel 1 ~ 8 => Values: 0 ~ 7	Values: 1 ~ 10
	0x9	Auto Mix Switch	ON = 0 OFF=1	
Feedback Inhibition	0x1	Switch	ON = 0 OFF=1	
	0x2	Feedback Point Frequency (Hz)	Freq 1 ~ 8 => Values: 0 ~ 7	20Hz to 20 000Hz in steps of 1Hz => Values 20 ~ 20 000

	0x3	Feedback Point Gain (dB)	Freq 1 ~ 8 => Values: 0 ~ 7	(-)24dB to 0dB in steps of 0,01dB => Values: -2400 ~ 0
	0x4	Octave Value	1/5 Octave = 20 1/10 Octave = 10	
	0x5	Type	Freq 1 ~ 8 => Values: 0 ~ 7	Dynamic = 0 Fixed = 1 Manual = 2
	0x8	Panic Threshold (dB)	(-)36dB to 0dB in steps of 0,01dB => Values: -3600 ~ 0	
	0x9	Feedback Threshold (dB)	(-)96dB to -24dB in steps of 0,01dB => Values: -9600 ~ -2400	
Auto Gain Compensation (AGC)	0x1	Switch	ON = 0 OFF=1	
	0x2	Threshold (dB)	(-)72dB to -20dB in steps of 0,01dB => Values -7200 ~ -2000	
	0x3	Target Level (dB)	(-)40dB to 0dB in steps of 0,01dB => Values -4000 ~ 0	
	0x4	Ratio	1dB to 20dB in steps of 0,01dB => Values 100 ~ 2000	
	0x5	Attack Time (ms)	1ms to 500ms in steps of 1ms => Values 1 ~ 500	
	0x6	Release Time (ms)	1ms to 10 000ms in steps of 1ms => Values 1 ~ 10 000	
AEC	0x1	Echo Cancellation Switch	ON = 0 OFF=1	
	0x2	Echo Cancellation Mode	Conservative (128ms) = 0 Moderate (256ms) = 1 Aggressive (512ms) = 2	
Automatic Noise Suppression (ANS)	0x1	ANS Switch	ON = 0 OFF=1	
	0x2	ANS Level (dB)	6dB = 0 10dB =1	

			15dB = 2 18dB = 3	
ANS Channel	0x1	Channel	Channel Values: 0 ~ 13	ON = 1 OFF=0
System Control	0x2	System Mute	ON = 1 OFF=0	
Limiter	0x1	Limiter Switch	ON = 0 OFF=1	
	0x2	Limiter Threshold (dB)	(-)48dB to 0dB in steps of 0,01dB => Values -4800 ~ 0	
	0x5	Release Time (ms)	1ms to 10 000ms in steps of 1ms => Values 1 ~ 10 000	

Appendix C, Command Structure Examples
V1 examples

SET examples

1. Set input Gain to -3dB for input 4.		
Structure	**Command**	**Comments**
Byte 1 : Fixed start B3	B3	Fixed
Byte 2: Message type: SET= 21, GET =22 and PRESET =13	21	SET
Byte 3: Fixed to AB	AB	Fixed
Byte 4: Version number: V1 = 01 and V2 =02.	00	Version 1
Byte 5-6: Module ID: See Appendix A: Module ID in manual.	2B 01	Appendix A, Input source ID = 299 gets 2B 01 in little endian format
Byte 7-8: Parameter type: Appendix B: Module Parameter Types (1).	01 00	Appendix B Input source Gain = 1 gets 01 00 in little endian format
Byte 9-10: Parameter 1	03 00	Inputs starts counting at 0 = input 1 so input 4 gets value 3 and in little endian 03 00
Byte 11-12: Parameter 2	D4 FE	(-)3db in hundreds gets value -300 which corresponds to D4 FE in little endian format.
SET command: B3 21 AB 00 2B 01 01 00 03 00 D4 FE	Response: B3 21 AB **E0** 2B 01 01 03 00 00 D4 FE	

2. Set Input 8 to Mute.		
Structure	**Command**	**Comments**
Byte 1 : Fixed start B3	B3	Fixed
Byte 2: Message type: SET= 21, GET =22 and PRESET =13	21	SET
Byte 3: Fixed to AB	AB	Fixed
Byte 4: Version number: V1 = 01 and V2 =02.	00	Version 1
Byte 5-6: Module ID: See Appendix A: Module ID in manual.	2B 01	Appendix A, Input source ID = 299 gets 2B 01 in little endian format
Byte 7-8: Parameter type: Appendix B: Module Parameter Types (1).	02 00	Appendix B Input source Mute = 2 gets 02 00 in little endian format
Byte 9-10: Parameter 1	07 00	Inputs starts counting at 0 = input 1 so input 8 gets value 7 and in little endian 07 00
Byte 11-12: Parameter 2	01 00	Mute On = 1 which corresponds to 01 00 in little endian format.
SET command: B3 21 AB 00 2B 01 02 00 07 00 01 00	Response: B3 21 AB **E0** 2B 01 02 00 07 00 01 00	

3. Set AEC Switch 1 to ON		
Structure	**Command**	**Comments**
Byte 1 : Fixed start B3	B3	Fixed
Byte 2: Message type: SET= 21, GET =22 and PRESET =13	21	SET
Byte 3: Fixed to AB	AB	Fixed
Byte 4: Version number: V1 = 01 and V2 =02.	00	Version 1
Byte 5-6: Module ID: See Appendix A: Module ID in manual.	A3 00	Appendix A, Echo Cancellation = 163 gets A3 00 in little endian format
Byte 7-8: Parameter type: Appendix B: Module Parameter Types (1).	01 00	Appendix B Echo Cancellation Switch = 1 gets 01 00 in little endian format
Byte 9-10: Parameter 1	00 00	Value 0 = AEC on which corresponds to 00 00 in little endian format.
Byte 11-12: Parameter 2	00 00	Fill up with 00 00
SET command: B3 21 AB 00 A3 00 01 00 00 00 00 00	Response: B3 21 AB **E0** A3 00 01 00 00 00 00 00	

GET examples

1. Get Input Gain for input 2		
Structure	**Command**	**Comments**
Byte 1 : Fixed start B3	B3	Fixed
Byte 2: Message type: SET= 21, GET =22 and PRESET =13	22	GET
Byte 3: Fixed to AB	AB	Fixed
Byte 4: Version number: V1 = 01 and V2 =02.	00	Version 1
Byte 5-6: Module ID: See Appendix A: Module ID in manual.	2B 01	Appendix A, Input source ID = 299 gets 2B 01 in little endian format
Byte 7-8: Parameter type: Appendix B: Module Parameter Types (1).	01 00	Appendix B Input source Gain = 1 gets 01 00 in little endian format
Byte 9-10: Parameter 1	01 00	Inputs starts counting at 0 = input 1 so input 2 gets value 1 and in little endian 01 00
Byte 11-12: Parameter 2	00 00	Fill up with 00 00
GET command: B3 22 AB 00 2B 01 01 00 01 00 00 00	Response: B3 23 AB **E0** 2B 01 01 00 01 00 **24 FF** (24 FF = -220 => -2,2dB)	

2. Get Mute status for output 5		
Structure	**Command**	**Comments**
Byte 1 : Fixed start B3	B3	Fixed
Byte 2: Message type: SET= 21, GET =22 and PRESET =13	22	GET
Byte 3: Fixed to AB	AB	Fixed
Byte 4: Version number: V1 = 01 and V2 =02.	00	Version 1
Byte 5-6: Module ID: See Appendix A: Module ID in manual.	27 01	Appendix A, Output ID = 295 gets 27 01 in little endian format
Byte 7-8: Parameter type: Appendix B: Module Parameter Types (1).	02 00	Appendix B Output Mute = 2 gets 02 00 in little endian format
Byte 9-10: Parameter 1	04 00	Inputs starts counting at 0 = output 1 so output 5 gets value 4 and in little endian 04 00
Byte 11-12: Parameter 2	00 00	Fill up with 00 00
GET command: B3 22 AB 00 27 01 02 00 04 00 00 00	Response: B3 21 AB **E0** 27 01 02 00 04 00 **01** 00 (01 00 = Mute On)	

PRESET example

1. Get Input Gain for input 2		
Structure	**Command**	**Comments**
Byte 1 : Fixed start B3	B3	Fixed
Byte 2: Message type: SET= 21, GET =22 and PRESET =13	13	PRESET
Byte 3: Fixed to AB	AB	Fixed
Byte 4: Version number: V1 = 01 and V2 =02.	00	Version 1
Byte 5-6: Preset number 1-16	02 00	Preset 3 is then value 3 which corresponds to 02 00 in little endian format.
Byte 7-8: Not used	00 00	Fill up with 00 00
Byte 9-10: Not used	00 00	Fill up with 00 00
Byte 11-12: Not used	00 00	Fill up with 00 00
PRESET command: B3 13 AB 00 02 00 00 00 00 00 00 00 00	Response: B3 13 AB **E0** 02 00 00 00 00 00 00 00 00	

V2 examples

SET example:

1. SET Input Mute CH1-CH4		
Structure	**Command**	**Comments**
Byte 1 : Fixed start B3	B3	Fixed
Byte 2: Message type: SET= 21, GET =22 and PRESET =13	21	SET
Byte 3: Fixed to AB (any will actually do but we set it to AB)	AB	Fixed
Byte 4: Version number: V1 = 01 and V2 =02.	01	Version 2
Byte 5: Input or Output	02	Input = 02
Byte 6: Start Channel	00	CH1 = 00
Byte 7: End Channel	03	CH4 = 03
Byte 8: Parameter type	02	Input Mute = 02
Byte 9: Parameter value	01	Mute = 01
SER command: B3 21 AB 01 02 00 03 02 01	Response: B3 21 AB **E1** 02 00 03 02 01	

GET example:

1. GET output Gain CH2-CH5		
Structure	**Command**	**Comments**
Byte 1 : Fixed start B3	B3	Fixed
Byte 2: Message type: SET= 21, GET =22 and PRESET =13	22	PRESET
Byte 3: Fixed to AB (any will actually do but we set it to AB)	AB	Fixed
Byte 4: Version number: V1 = 01 and V2 =02.	01	Version 2
Byte 5: Input or Output	01	Output = 01
Byte 6: Start Channel	01	CH2 = 01
Byte 7: End Channel	05	CH5 = 04
Byte 8: Parameter type	01	Input Gain = 01
Byte 9: Parameter value	01	Fill up with: 00 00 00 00 00 00 00 00
GET command: B3 22 AB 01 01 01 04 01 00 00 00 00 00 00 00 00	Response: B3 22 AB **E1** 01 01 04 01 **C8 00 18 FC 18 FC 50 FB** CH2 = C8 00 (little endian) = 200 > Gain = 2dB CH2 = 18 FC (little endian) = -1000 > Gain = -10dB CH2 = 18 FC (little endian) = -1000 > Gain = -10dB CH2 = 50 FB (little endian) = -1200 > Gain = -12dB	

RS232 to UDP example:

1. Send command "HELLO DSP" to 192.168.40.159 on UDP port 50 000.		
Structure	**Command**	**Comments**
Byte 1 : Fixed start B3	B3	Fixed
Byte 2: Message type: SET= 21, GET =22 and PRESET =13	74	Other controls
Byte 3: Fixed to AB (any will actually do but we set it to AB)	AB	Fixed
Byte 4: Version number: V1 = 01 and V2 =02.	01	Version 2
Byte 5~8: Prefix	08 00 00 00	Prefix =08 00 00 00
Byte 9~12: IP address of receiving device	C0 A8 28 9F	192.168.40.159 = C0 A8 28 9F
Byte 13~14: Port number	50 C3	50 000 = 50 C3
Byte 15: Data length	09	9 Bytes = 09
Byte 16: Reserved	00	00
Byte 17~128: Data	48 45 4C 4C 4F 20 44 53 50	"HELLO DSP" = 48 45 4C 4C 4F 20 44 53 50
Command: B3 74 AB 01 08 00 00 00 C0 A8 28 9F 50 C3 09 00 48 45 4C 4C 4F 20 44 53 50	Response: B3 74 AB **E1** 08 00 00 00 C0 A8 28 9F 50 C3 09 00 48 45 4C 4C 4F 20 44 53 50	

Customer Service

The return of a product to our Customer Service implies the full agreement of the terms and conditions hereinafter. There terms and conditions may be changed without prior notice.

1) **Warranty**

    We provide limited warranty for the product within **five years** counting from date of purchase (The purchase invoice shall prevail).

2) **Scope**

    These terms and conditions of Customer Service apply to the customer service provided for the products or any other items sold by authorized distributor only.

3) **Warranty Exclusion**
    - Warranty expiration.
    - Factory applied serial number has been altered or removed from the product.
    - Damage, deterioration or malfunction caused by:
        - Normal wear and tear.
        - Use of supplies or parts not meeting our specifications.
        - No certificate or invoice as the proof of warranty.
        - The product model showed on the warranty card does not match with the model of the product for repairing or had been altered.
        - Damage caused by force majeure.
        - Servicing not authorized by distributor.
        - Any other causes which don"t relate to a product defect.
    - Shipping fees, installation or labor charges for installation or setup of the product.

4) **Documentation:**

    Customer Service will accept defective product(s) in the scope of warranty coverage at the sole condition that the malfunction has been clearly defined, and upon reception of the documents or copy of invoice, indicating the date of purchase, the type of product, the serial number, and the name of distributor.

Remarks: Please contact your local distributor for further assistance or solutions.