

# **DCP-555 Help File**

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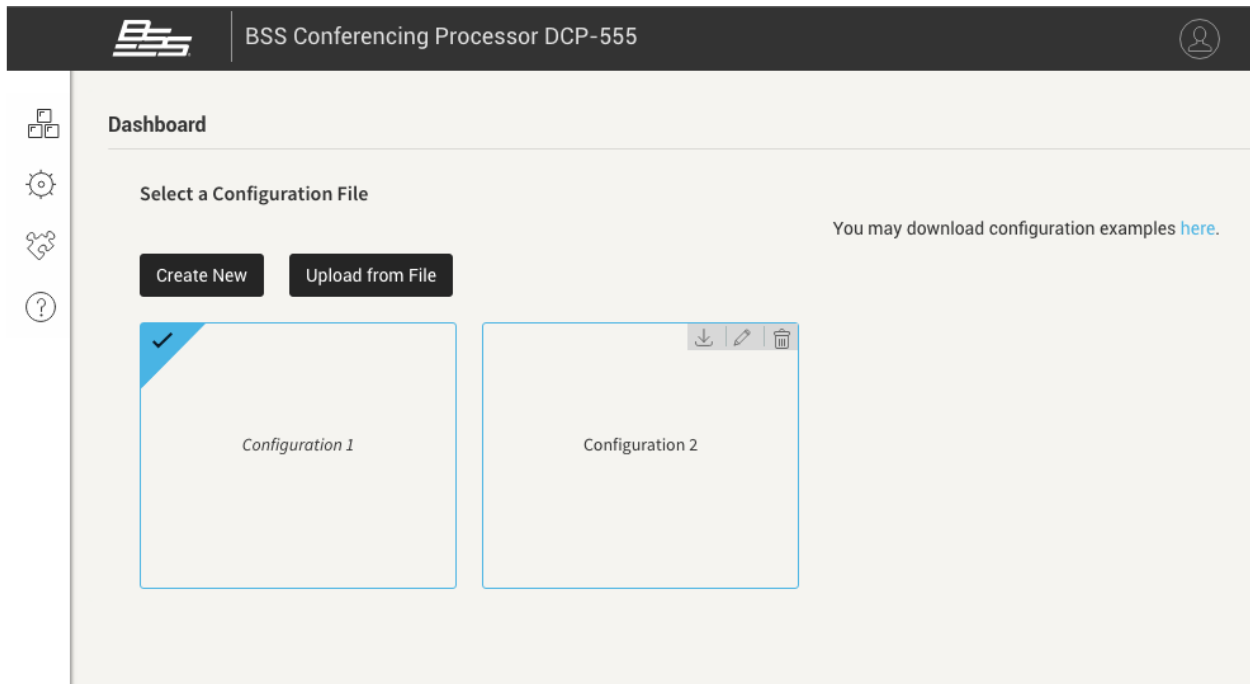
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# Dashboard

The Dashboard is the first page displayed when configuring a DCP-555. Press the **Create New** button to start from scratch or **Upload from File** button to load an existing configuration file.



## Menu Icons

Use these icons to navigate the app.

## User Icon

Select this icon to change the login password or sign out.

## Create New

Press this button to create a new configuration and then edit to taste.

## Upload From File

Press this button to load a configuration template file located on the control device.

## Configurations

Saved configurations will reside in boxes on this page. The checkmark indicates the currently loaded configuration. Click an inactive configuration to load it. Click an active configuration box to edit the configuration.

### Download Configuration

This button appears when hovering over a configuration. Press this button to download the configuration to the control device.

### Edit Configuration Name

This button appears when hovering over a configuration. Press this button to rename the configuration.

### Delete Configuration

This button appears when hovering over a configuration that is inactive. Press this button to delete the configuration.

**Note:** Deleting a configuration is irreversible. To retain a copy of the configuration, download it before deleting it.

# Change Password

The screenshot shows a web interface for changing a password. At the top, there is a dark header with the BSS logo on the left, the text 'BSS Conferencing Processor DCP-555' in the center, and a calendar icon and a user profile icon on the right. Below the header, the main content area is titled 'Change Password'. On the left side of this area, there is a vertical sidebar with four icons: a grid of four squares, a gear, a puzzle piece, and a question mark. The main form contains three input fields: 'Current Password\*' with a placeholder 'Existing Password', 'New Password\*' with a placeholder 'New Password', and 'Confirm New Password\*' with a placeholder 'New Password'. Below these fields is a note: 'Note: Password should be at least 5 letters long'. In the bottom right corner of the form area, it says 'Version - 0.0.30 (15th May 2018)'. At the bottom center, there is a dark button labeled 'Update Password'.

## Current Password\*

Enter the current password into this field. The asterisk indicates that this is a required field.

## New Password\*

Enter the new password into this field. The asterisk indicates that this is a required field.

## Confirm New Password\*

Enter the new password again into this field for confirmation. The asterisk indicates that this is a required field.

## Update Password Button

When done, press this button to update the password.

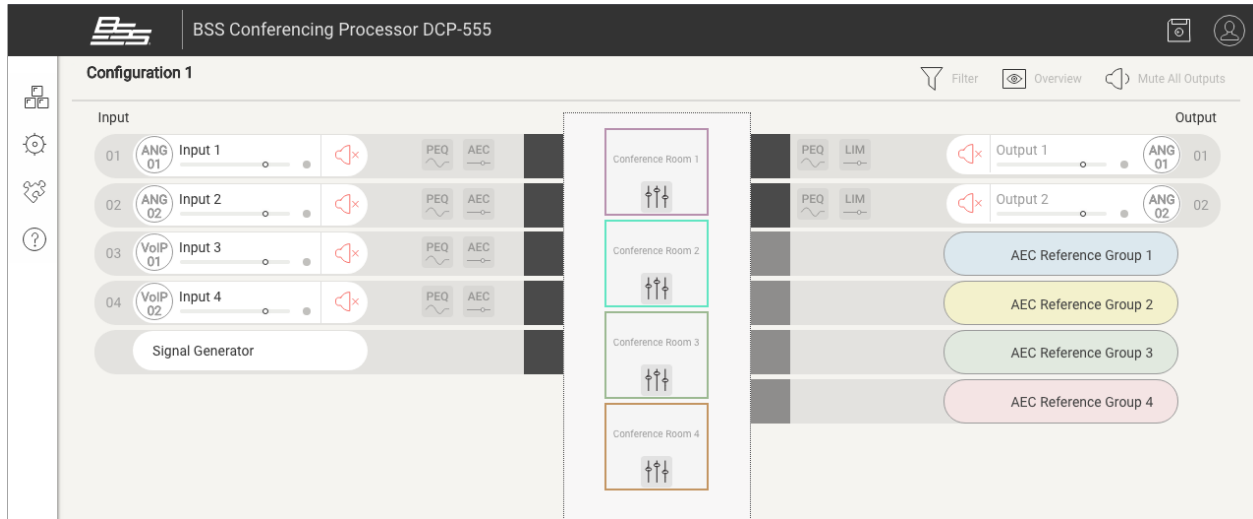
**Important:** Take note of the password you create and keep it in a safe place. There is no way to retrieve the password, and the unit will require a Factory Reset if the password is forgotten.

# Audio Settings

The Audio Settings page allows you to route audio and adjust DSP settings.

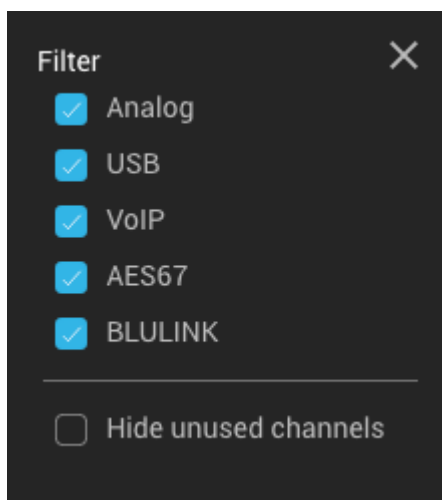
There are a total of 28 input slots and 12 output slots. Any combination of input types (i.e., analog, VoIP, USB, etc.) can be configured, and same-type input channels can be stereo ganged if required for the application.

Any combination of output types can also be configured (i.e., analog, VoIP, USB, etc.). Same-type outputs can be stereo ganged, or each output can be duplicated (mirrored) to up to 7 additional outputs for applications requiring multed outputs.



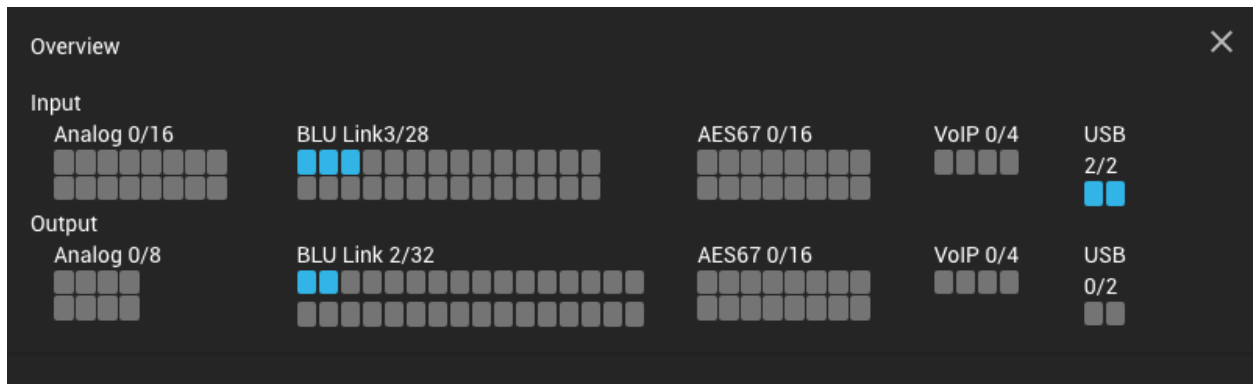
## Filter

Press this button to select the type of channels that are visible on the page. Use the **Hide unused channels** checkbox to hide channels that are not configured.



## Overview

Press this button to see an overview of all input and output channel assignments in the current configuration.



## Mute All Outputs

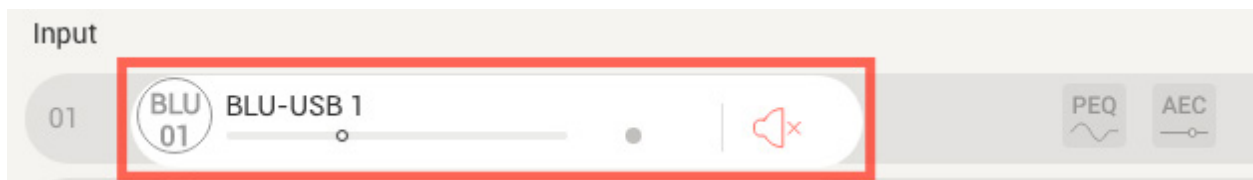
Press this button to mute all outputs.

## Save Icon

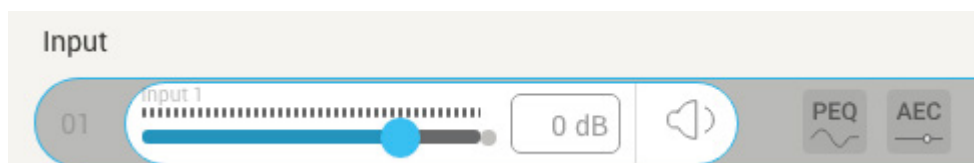
Click this icon and select "Save" to save audio setting changes to the current configuration. Select "Save As" to save the changes as a new configuration.

## Input Channel Selection, Gain, and Mute

Select the white area of an input channel to view the [Input Channel Configuration](#) settings. The dot to the right of the gain slider lights green to indicate signal presence or red to indicate clipping. Press the **Input Mute** button to mute the input channel.

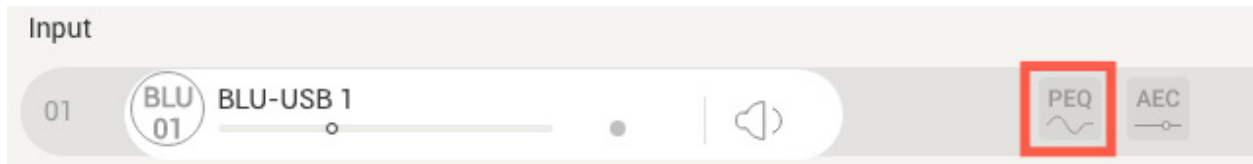


Press the Slider of an input channel to view and adjust the input gain without opening the [Input Channel Configuration](#) page.



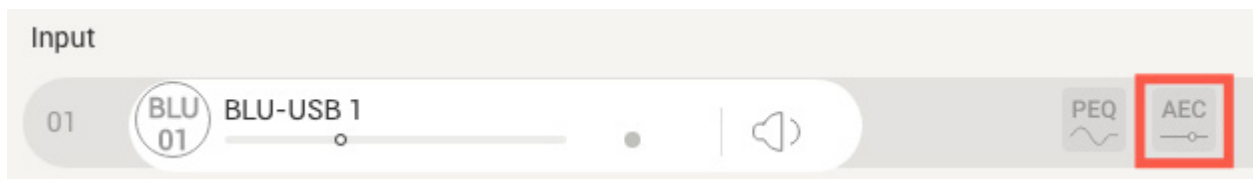
## Input PEQ

Press the Input PEQ button to view and edit [PEQ](#) settings for an input channel.



## AEC

Press the **AEC** button to view and edit [AEC](#) (Acoustic Echo Cancellation) settings for an input channel. The first 12 input channels have AEC processing available.



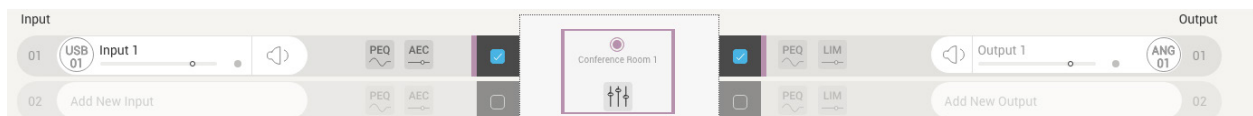
## Compressor

Press the **Compressor** button to view and edit [Compressor](#) settings for an input channel. Input channels 13-28 have a Compressor available.



## AutoMixers

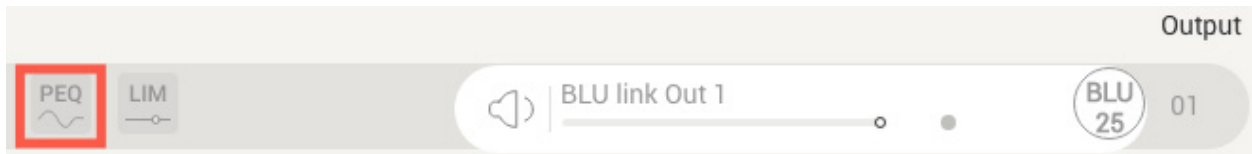
There are a total of 4 AutoMixers located in the center of the Audio Settings page. To route channels through an AutoMixer, select the AutoMixer then check the boxes for the input and output channels that you wish to assign. The colored strips on each input/output channel strip correspond with the colored border of each AutoMixer to provide at-a-glance identification of AutoMixer routing assignments. Double-click an [AutoMixer](#) to access its settings.





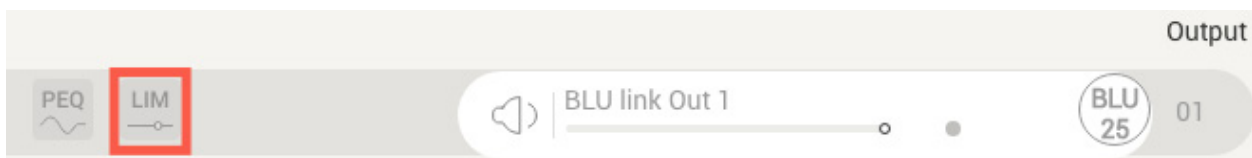
## Output PEQ

Press the Output PEQ button to view and edit [PEQ](#) settings for an output channel.



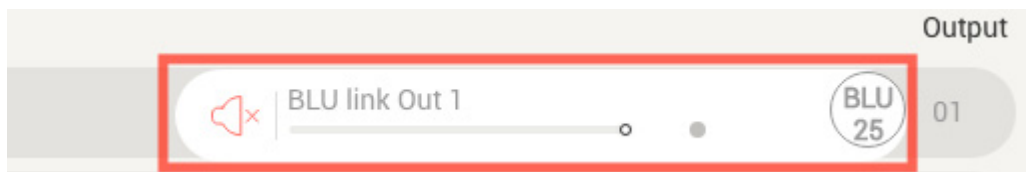
## Limiter

Press the Limiter button to view and edit [Limiter](#) settings for an output channel.

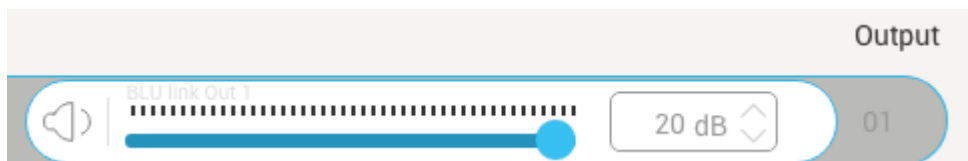


## Output Channel Selection, Gain, and Mute

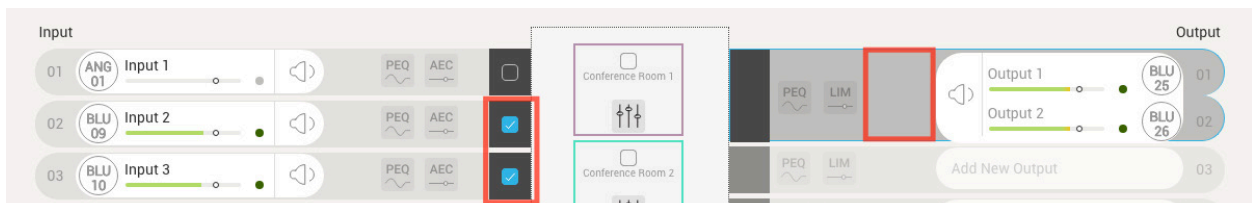
Select the white area of an output channel to view the [Output Channel Configuration](#) settings. The dot to the right of the gain slider lights red to indicate clipping. Press the **Output Mute** button to mute the output channel.



Select the Slider of an output channel to view and adjust the output gain.



To assign inputs to an output, first select the gray area of the output. Now, check the checkboxes for each input you wish to assign.



## Signal Generator

Select the white area of the Signal Generator channel to view the [Signal Generator](#) settings.

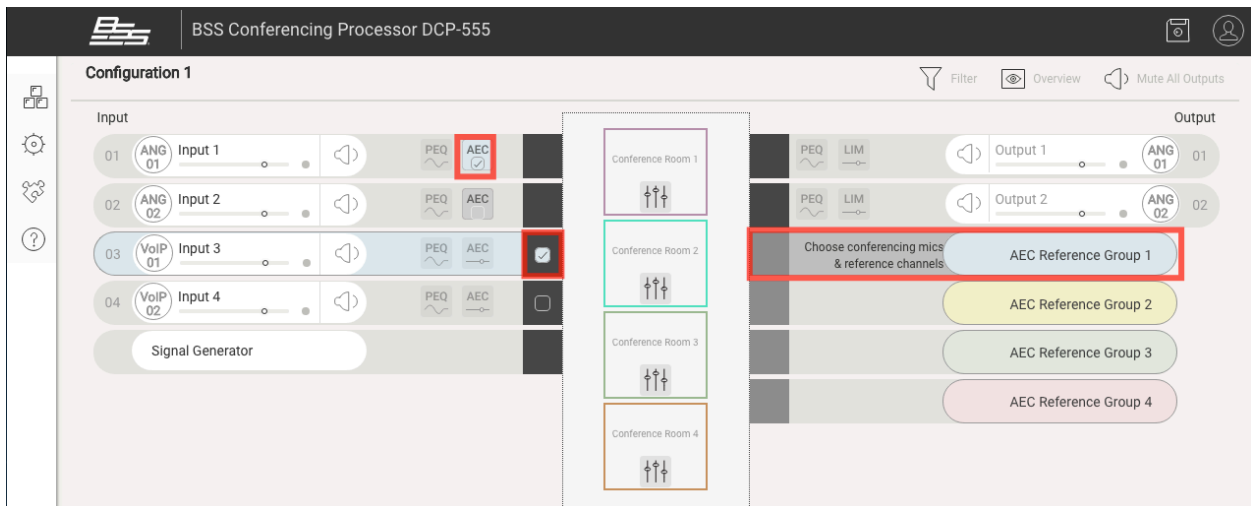


## AEC Reference Groups

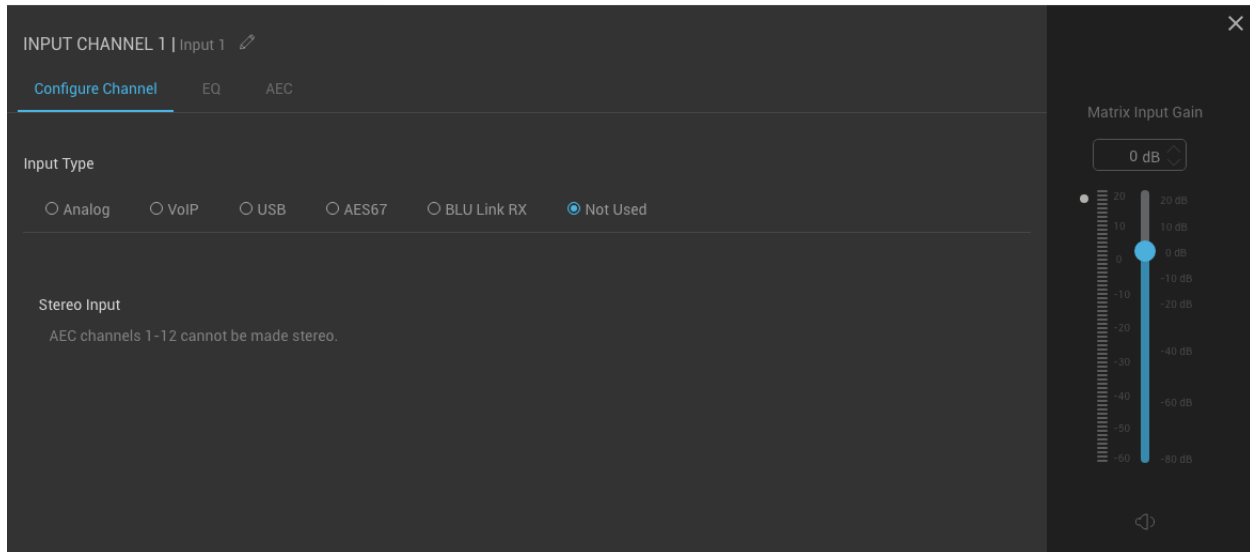
The AEC Reference Groups are used to inform the AEC algorithm which signals will be referenced for processing, which includes the conference room mics (the outgoing signal) and the incoming call signal (the signal to be cancelled so that it isn't transmitted as an echo). There are a total of 4 AEC Reference Groups, one for each conference room.

Before assigning any AEC Reference Groups, you must first [configure the input channels](#) and then enable [AEC](#) for all inputs that will use AEC.

To assign an AEC Reference Group, select the group and then check the checkbox in the AEC icon for all input channels that you wish to assign to the group. You must then check the checkbox to the right of the input that carries the incoming call signal.



# Input Channel Configuration



## Input Channel Name

Click the **Pencil** icon to rename the input channel.

## EQ, AEC, and Compressor Tabs

Use these tabs to navigate to the input channel's [PEQ](#), [AEC](#), or [Compressor](#) page. Input channels 1-12 have AEC processing while input channels 13-28 have a Compressor instead.

## Input Type

Select the desired input type using these radio buttons. Options include **Analog**, **VoIP**, **USB**, **AES67**, **BLU Link**, and Not Used.

**Note:** To configure a channel, you must select an **Input Type** and then an **Input** channel number before closing the window.

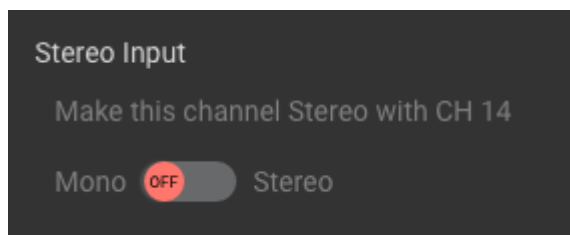
## Matrix Input Gain, Meter, and Mute

This meter displays signal level (in dBu) post input processing. The dot to the left of the meter lights green to indicate signal presence or red to indicate clipping. Use the **Matrix**

**Input Gain** field or slider to adjust the signal level feeding the output(s). Use the **Mute** button to mute the input channel.

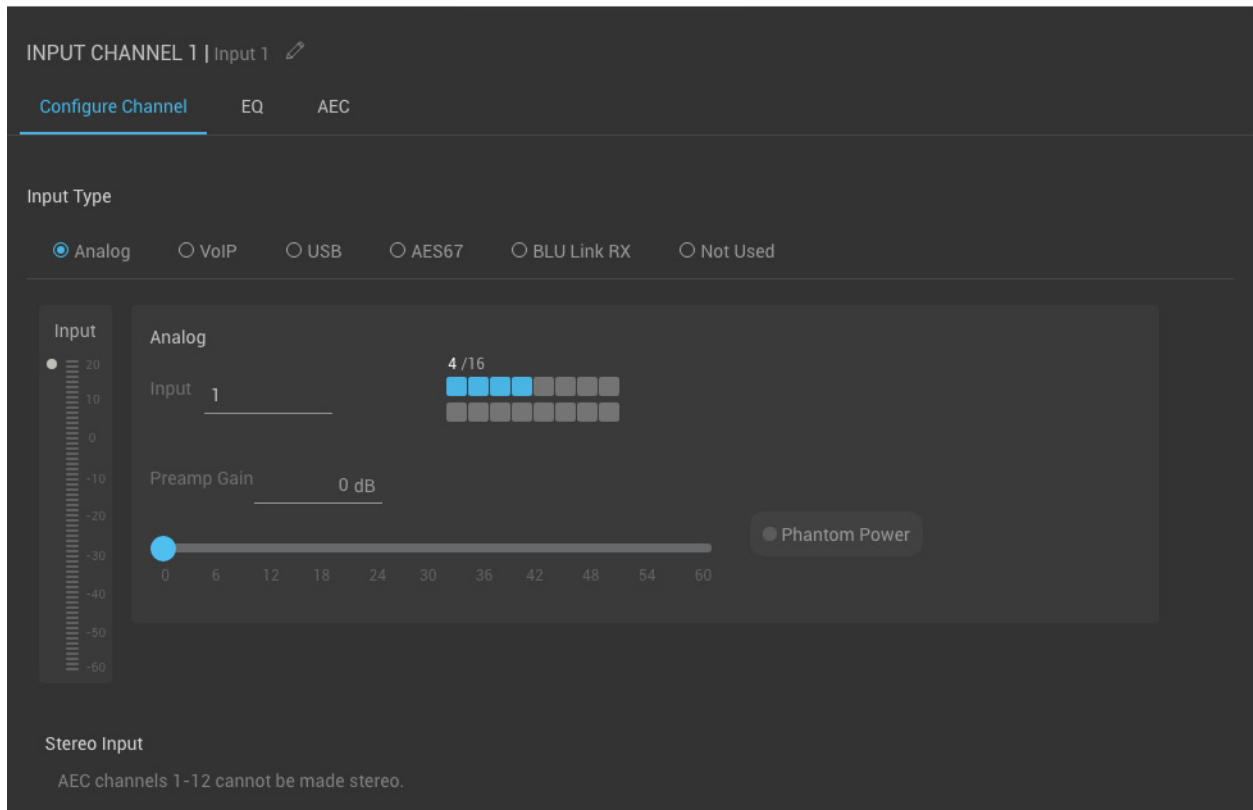
## Stereo Input Switch

Use this switch to stereo gang input channels. To stereo gang two channels, you must first configure both with matching **Input Type** selections (e.g., analog, VoIP, USB, etc.). The **Stereo Input** switch will then be visible on the lower-numbered input channel and can be engaged if required for the application.



**Note:** Only non-AEC input channels (13-28) can be stereo ganged.

# Analog Input Type



## Input Meter

This meter displays the input signal level in dBu. The dot to the left of the meter lights green to indicate signal presence or red to indicate clipping.

## Input

Use this field to assign an analog input channel.

## Channel Assignment Array

This assignment array displays an overview of the analog input channels assigned in the current configuration.

## Preamp Gain

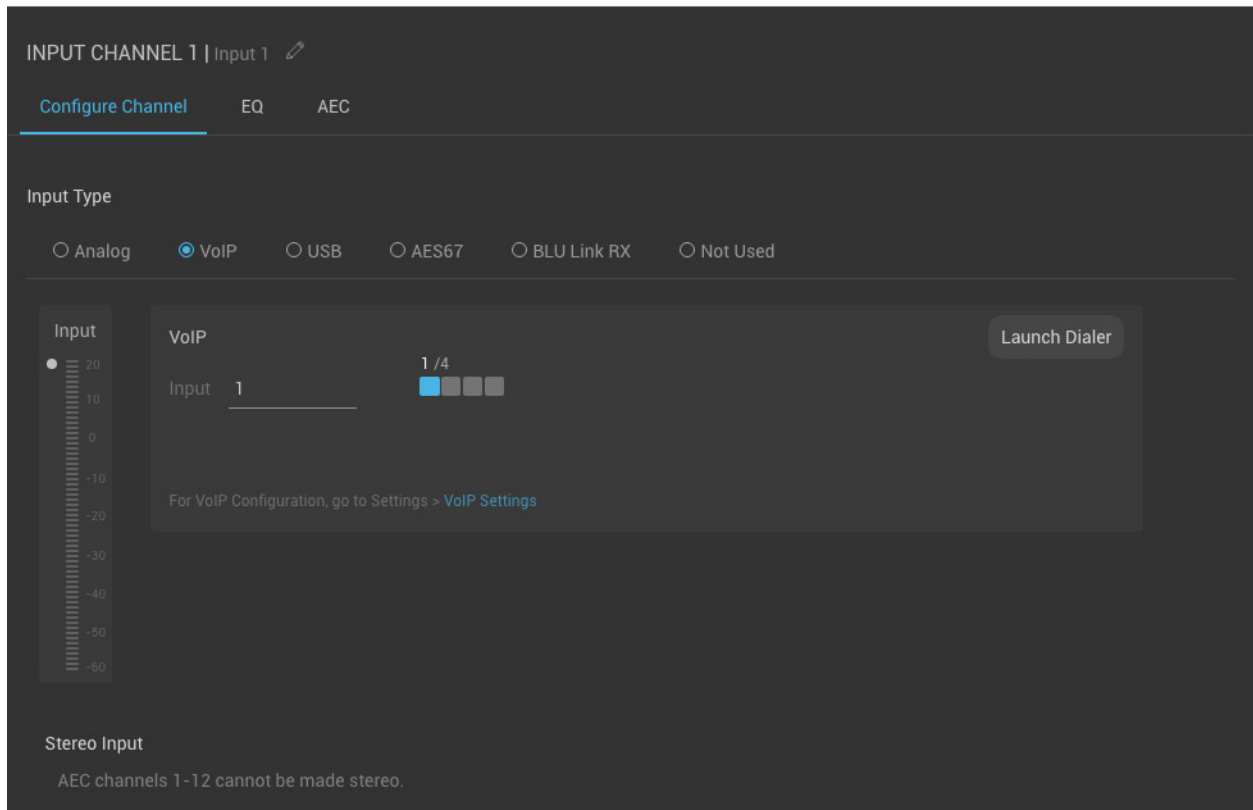
Use the **Preamp Gain** field or slider to adjust the channel's analog input gain. Adjustments can be made in 2dB increments.

## Phantom Power

Pressing this button enables +48VDC at the microphone input. Enable this button only if you are connecting condenser-type microphones which require phantom power to operate. To avoid any potential plosives through the system, it is recommended to ensure the input gain is turned down, connect the condenser microphone, and then enable this button.

**Warning!** Do not enable this button if connecting equipment or microphones that do not require phantom power.

# VoIP Input Type



## Input Meter

This meter displays the input signal level in dBu. The dot to the left of the meter lights green to indicate signal presence or red to indicate clipping.

## Input

Use this field to assign a VoIP input channel.

## Channel Assignment Array

This assignment array displays an overview of the VoIP input channels assigned in the current configuration.

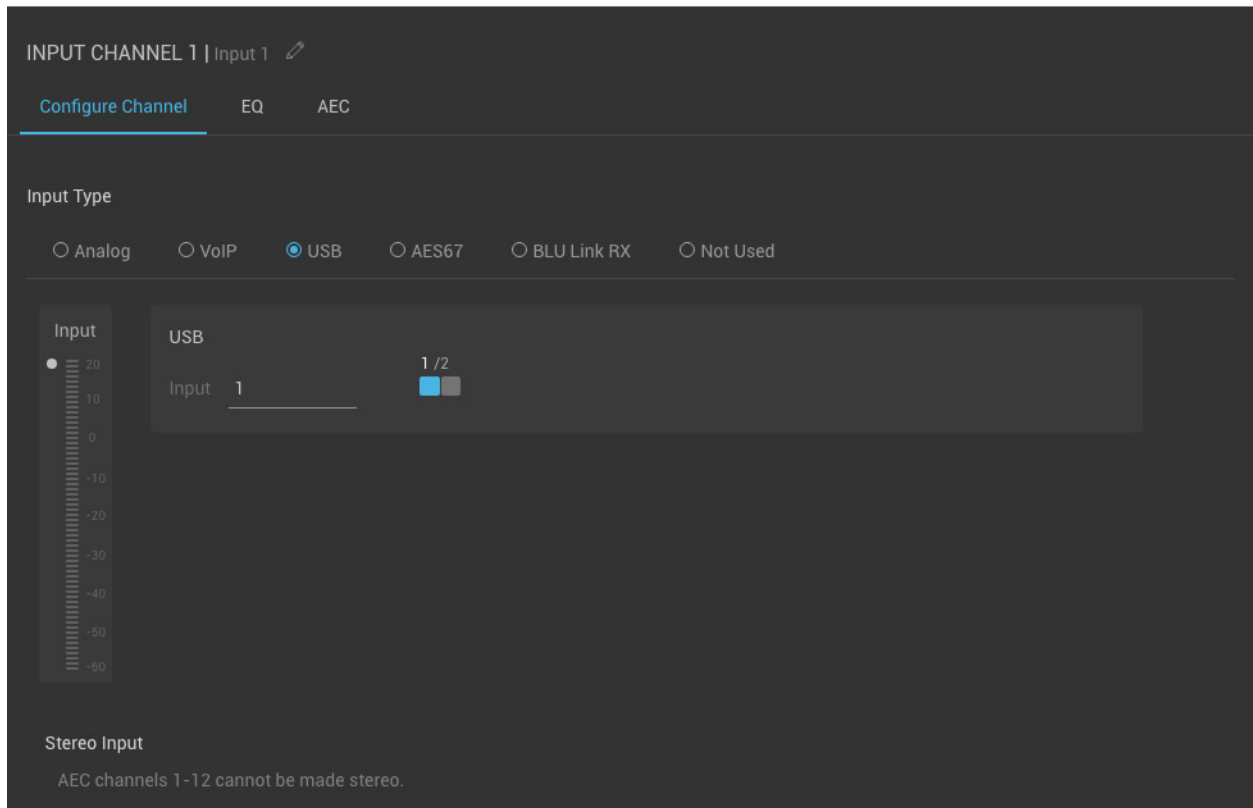
## Launch Dialer Button

## DCP-555 Help File

Press this button to open the [Dialer](#) for the selected VoIP input line.



# USB Input Type



## Input

Use this field to assign a USB input channel.

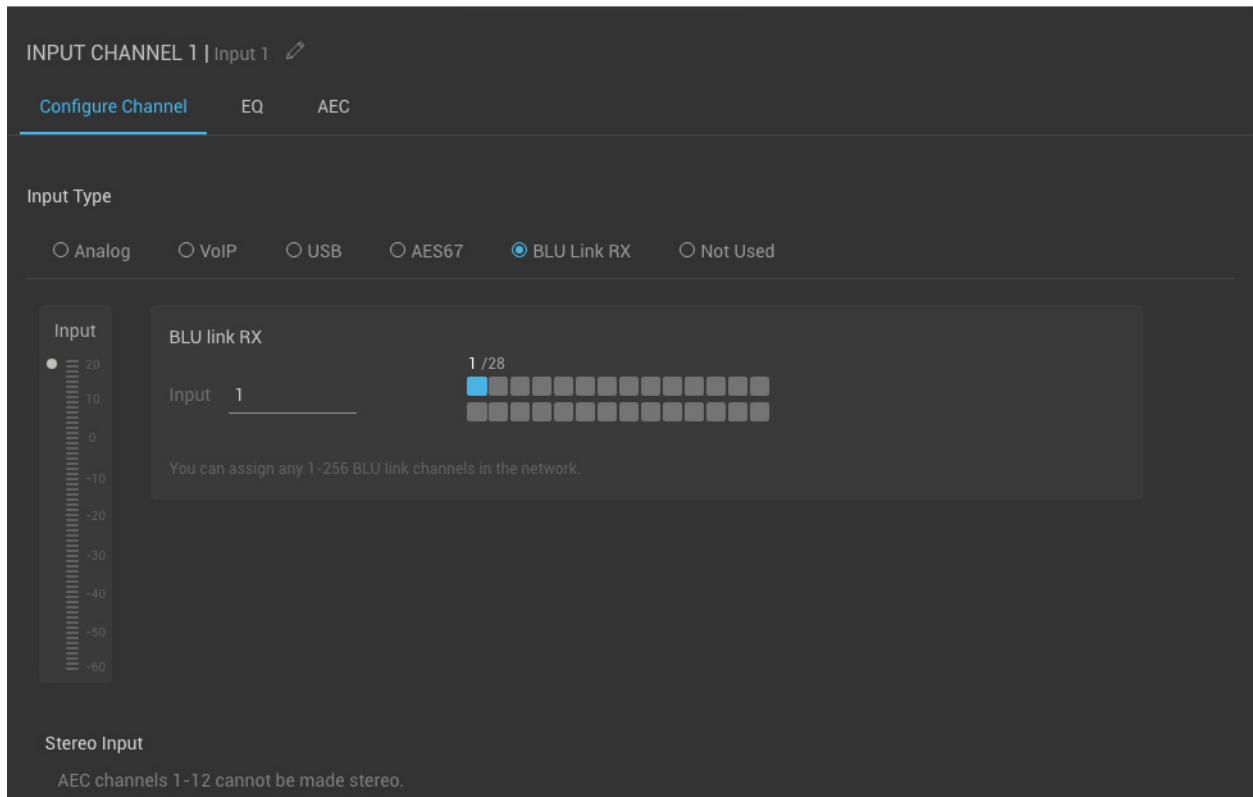
## Channel Assignment Array

This assignment array displays an overview of the USB input channels assigned in the current configuration.

# **AES67 Input Type**

This feature to be implemented in a future firmware release.

# BLU Link Input Type



## Input Meter

This meter displays the input signal level in dBu. The dot to the left of the meter lights green to indicate signal presence or red to indicate clipping.

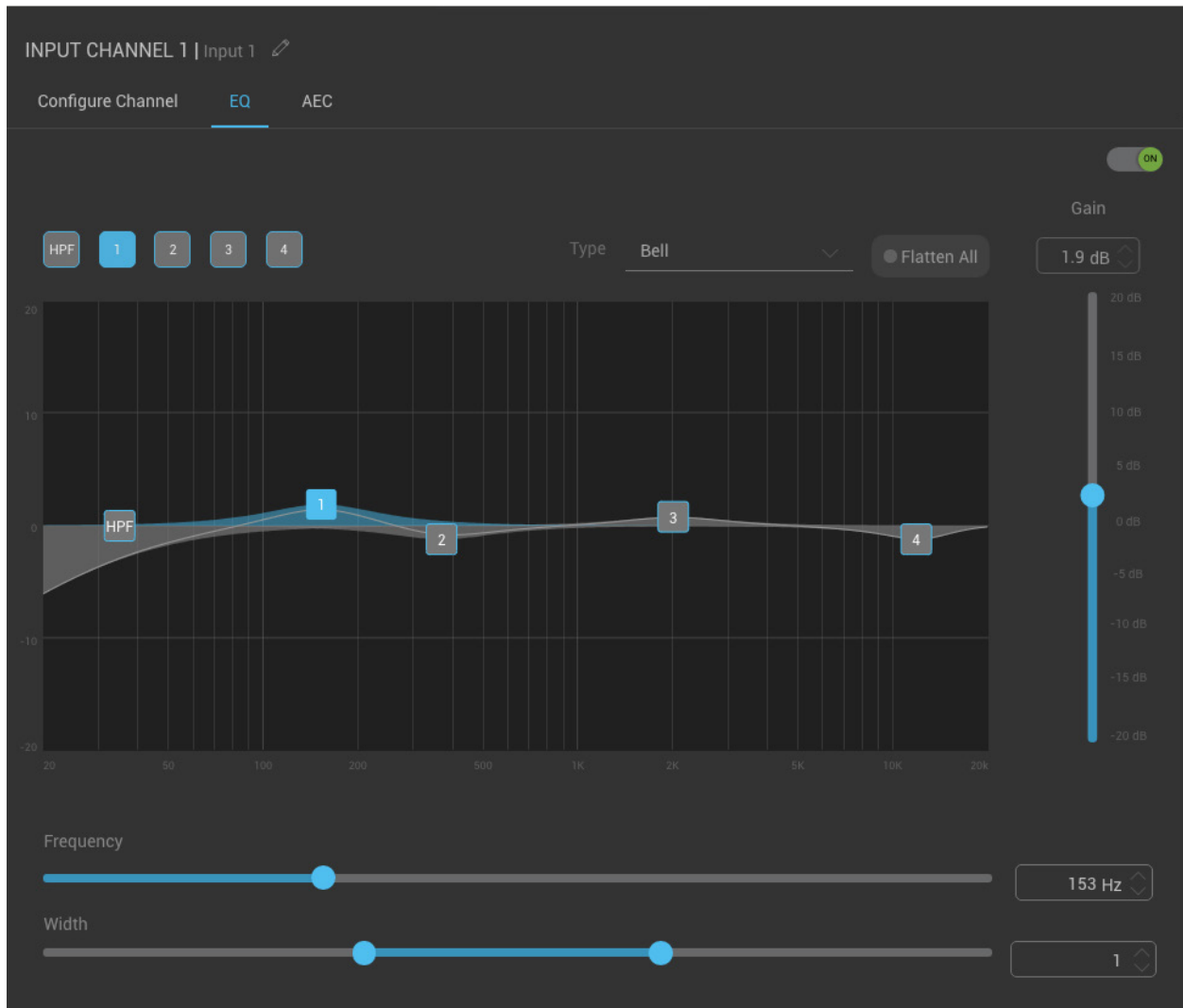
## Input

Use this field to assign a BLU link input channel.

## Channel Assignment Array

This assignment array displays an overview of the BLU link input channels assigned in the current configuration.

# Input PEQ



## On/Off Switch

This switch turns the PEQ on or off.

## Filter Band Selection

Use these buttons to select one of the five available bands for editing.

## Type

When "Filter 1" is selected, use this button to toggle between a bell or low shelf filter type. When "Filter 4" is selected, use this button to toggle between a bell or high shelf filter type.

### **Flatten All Button**

Press this button to flatten all bands. Press the button again to restore flattened bands.

### **Graphical Display**

Grab the numbered filter bands in this display to adjust gain and frequency settings graphically.

### **Gain**

This slider adjusts the gain of the selected filter band, with a range of +/-20dB.

### **Frequency**

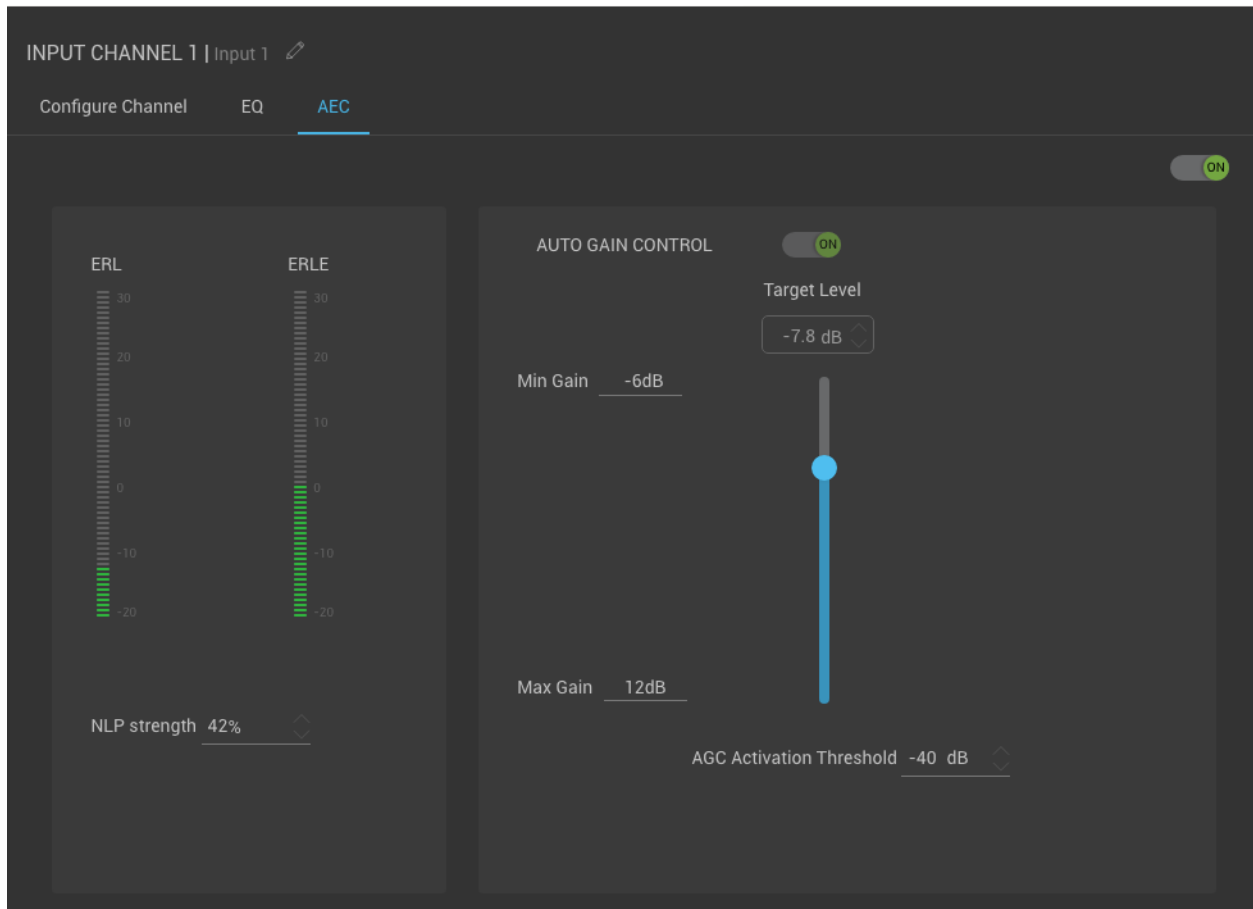
Use this slider or field to adjust the frequency of the selected filter band.

### **Width**

Use this slider or field to adjust the width of the selected filter band. Settings range from 0.1 to 128.

# AEC

**Important:** Once AEC has been enabled and configured for the desired channels, [AEC Reference Groups](#) must then be assigned for AEC to function.



## On/Off Switch

This switch turns AEC on or off.

## ERL Meter

The Echo Return Loss (ERL) meter is a measure of the room's natural attenuation of the far-side audio as it is emitted through the conference room speaker(s) and re-enters the microphone(s). A proper gain structure is critical for distortion free sound and optimal performance when using AEC.

**Note:** This meter will not update during double-talk. It is updated based on far-side

speech only.

## ERLE Meter

The Echo Return Loss Enhancement (ERLE) meter displays how much acoustic echo is being removed from the signal. This measurement includes both the natural room attenuation as indicated by the ERL meter and the amount of echo removed by the AEC algorithm. A lower level indicates more echo being removed. The lower the meter, the better.

**Note:** As dictated by industry standards, NLP (Non-Linear Processing) contributions are not included in the ERLE reading. NLP contributions are made in addition to this meter's reading.

## NLP Strength

The Non-Linear Processing (NLP) Strength setting determines the amount of non-linear suppression that will be applied in conjunction with the AEC algorithm. NLP will remove the residual echo not removed by the linear part of the AEC algorithm.

This parameter represents a trade-off between achieving good double-talk performance, with no suppression of the local speech signal, and very robust echo suppression, with no echo audible on the far side.

At its most aggressive setting (NLP at 80%), the non-linear processing will remove any of the residual far-side echo picked up by the microphone. However, this is done with an increased risk that some of the near-side speech will be degraded as well, especially during double-talk. At its least aggressive setting (NLP at 0%), the non-linear processing is effectively disabled, which may let some echo through, but will allow for a more natural double-talk performance.

The best setting for this parameter may depend on several factors, including the acoustic properties of the room and user preference. The default value of 42% may give a good balance between these two competing goals.

## Auto Gain Control On/Off Switch

This switch turns AGC on or off.

**Note:** Do not use AGC in conjunction with an AutoMixer. Doing so will cause unpredictable and uncontrollable level fluctuations.

### **Min Gain**

This parameter sets the maximum amount of attenuation that will be applied to the signal when the signal level is above the target. Values range from -20dB to 0dB.

### **Max Gain**

This parameter sets the maximum amount of gain that will be applied to the signal when the signal level is below the target. Values range from 0dB to +20dB.

### **Target Level**

Adjust this field or slider to set the desired target level to be achieved and maintained.

### **AGC Activation Threshold**

This parameter sets the level over which AGC recognizes a signal as a legitimate level to be processed. When the signal level is below this threshold, no gain will be applied to the signal. This prevents low-level signals and noise from being boosted excessively. Values range from -60dB to -10dB.



# Compressor



## On/Off Switch

This switch turns the Compressor on or off.

## Graphic Display

Displays the compression curve, with input level on the bottom and gain reduction on the right. Click and drag the cyan node to graphically adjust **Threshold** and the yellow node to adjust **Ratio**. The orange node indicates signal level.

## **In Meter**

This meter displays incoming signal level in dBu.

## **Threshold**

Use this slider or field to adjust the level at which compression will occur.

## **Attack**

Use this slider or field to adjust how quickly the Compressor will clamp down on the signal. Lower values will clamp down on the signal quickly, whereas higher values will let some of the signal pass before clamping down on the signal.

## **Release**

Use this slider or field to adjust how quickly the Compressor will release the compressed signal once it falls back below threshold.

## **Gain Reduction Meter**

This meter displays the amount of gain reduction being applied to the signal.

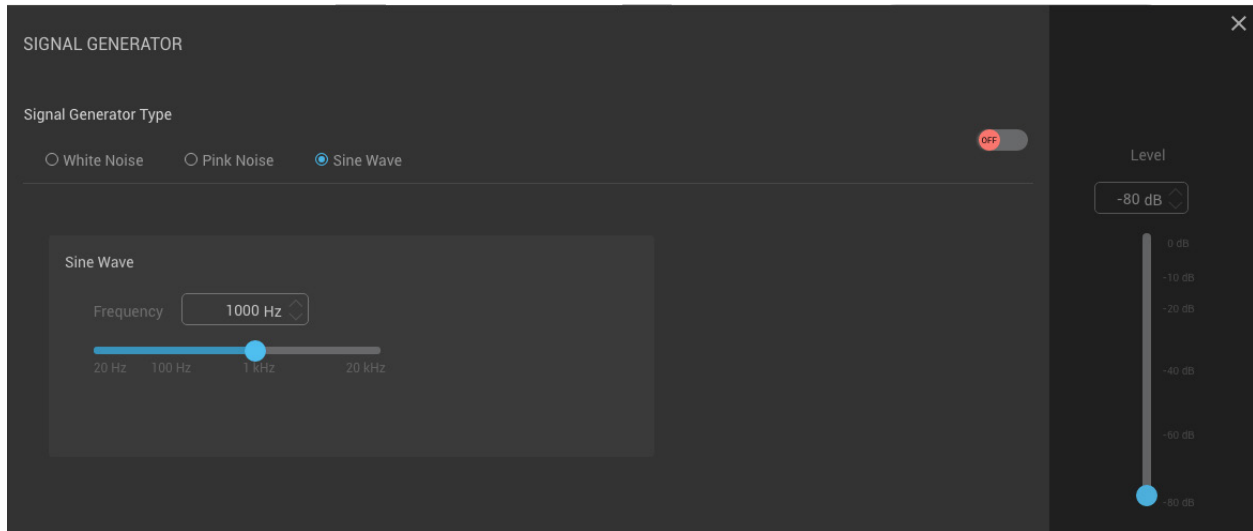
## **Ratio**

Use this slider or field to adjust how much compression will be applied once the threshold has been exceeded. Use higher ratios to apply more compression and lower ratios for subtle compression.

## **Gain**

Use this slider or field to adjust the output gain to makeup for gain lost due to compression.

# Signal Generator



## On/Off Switch

This switch turns the Signal Generator on or off.

## Signal Generator Type

Use these radio buttons to select the desired type of signal to generate: White Noise, Pink Noise, or Sine Wave.

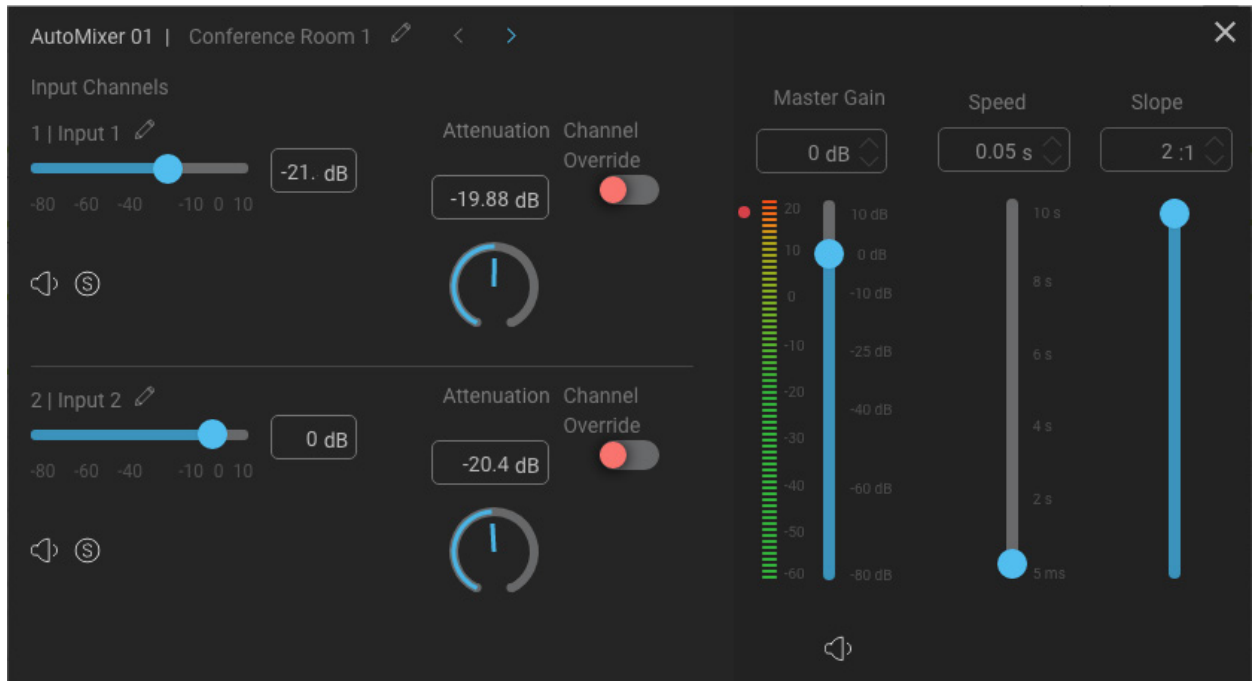
## Frequency

When the "Sine Wave" **signal generator type** is selected, use this slider or field to select the desired sine wave frequency.

## Level

Use this field or slider to adjust the Signal Generator's output level.

# AutoMixer



## AutoMixer Name

Click the **Pencil** icon to edit the AutoMixer name.

## Arrow Buttons

Use these buttons to navigate between the four available AutoMixers.

## Input Channel Name

Click the **Pencil** icon to rename an input channel.

## Input Gain

Use this slider or field to adjust the input gain for an input channel.

## Mute Button

Click this button to mute an input channel.

## **Solo Button**

Click this button to solo an input channel for auditioning.

## **Attenuation**

Use this control or field to attenuate each input channel in order to match the various channel levels.

## **Channel Override Switch**

Engage this switch to force a channel to be heard above all others.

## **Master Gain, Meter, and Mute**

Use this slider or field to adjust the signal level feeding the output(s). This meter displays outgoing signal level in dBu. The dot to the left of the meter lights green to indicate signal presence or red to indicate clipping. Use the **Mute** button to mute the AutoMixer output.

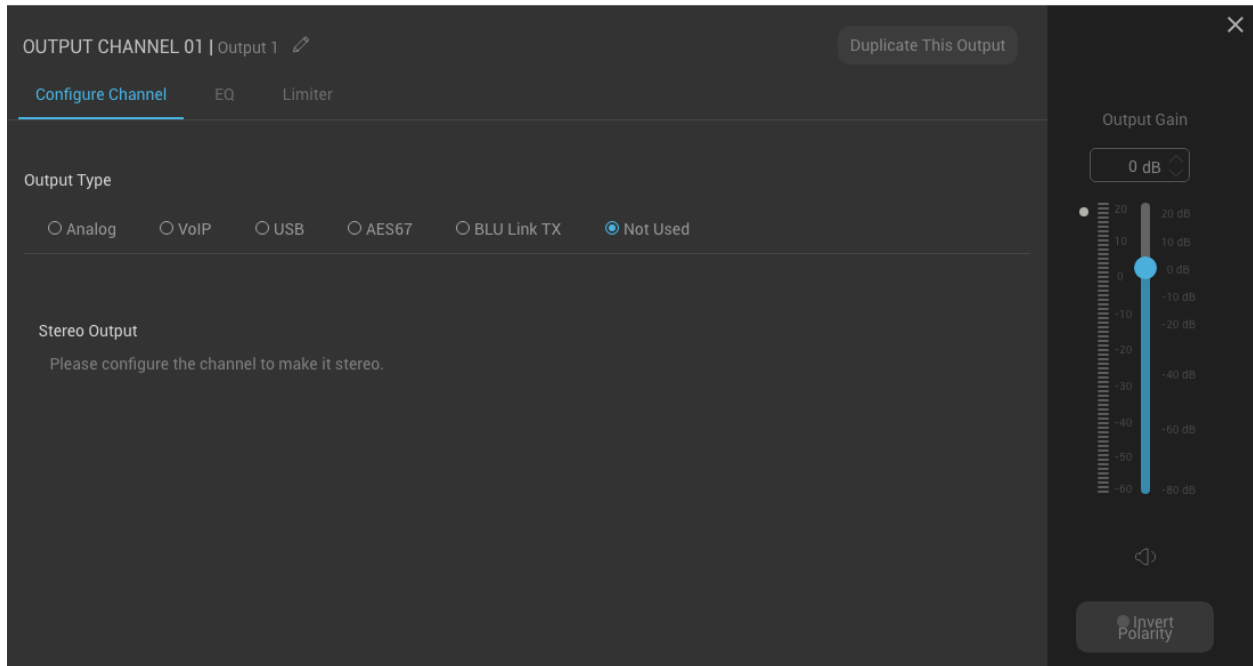
## **Speed**

This control determines how quickly a channel recovers from being open. Fast settings will be needed if different speakers are speaking in quick succession since slow settings would prevent a newly active channel from fading up quickly enough. If the setting is too fast however, the gain sharing could 'pump' the gain up and down between spoken words. A setting somewhere near the center of the travel will be found suitable in most cases.

## **Slope**

This control determines how aggressive the gain sharing will be. Low slope settings (1:1) cause the processor to behave just like a conventional mixer. High slope settings cause the inputs to attenuate more deeply for 'closed' microphones. The full 2:1 setting will be appropriate for most applications. If you do find the action to be too aggressive and do not want idle microphones to be 'dimmed' as much, then back this control off slightly.

# Output Channel Configuration



## Output Channel Name

Click the **Pencil** icon to rename the output channel.

## EQ, and Limiter Tabs

Use these tabs to navigate to the output channel's [PEQ](#) or [Limiter](#) page.

## Duplicate This Output Button

Press this button to create a 'mirrored' output which can then be assigned to a different output channel. This provides the ability to route a signal to multiple outputs. Any changes made to an output's processing parameters will also affect its duplicated outputs. Up to 7 duplicated outputs can be created for each output channel.

## Output Type

Select the desired output type using these radio buttons. Options include [Analog](#), [VoIP](#), [USB](#), [AES67](#), [BLU Link](#), and >Not Used.

**Note:** To configure a channel, you must select an **Output Type** and then an **Output** channel number before closing the window.

## Output Gain, Meter, and Mute Button

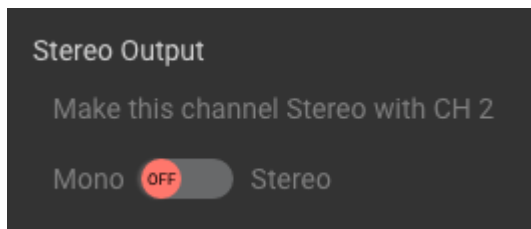
This meter displays output signal level (in dBu) post output processing. The dot to the left of the meter lights green to indicate signal presence or red to indicate clipping. Use the **Output Gain** field or slider to adjust the final output level pre Limiter. Use the **Mute** button to mute the output channel. Note that these settings will also affect any duplicated outputs for the channel.

## Invert Polarity Button

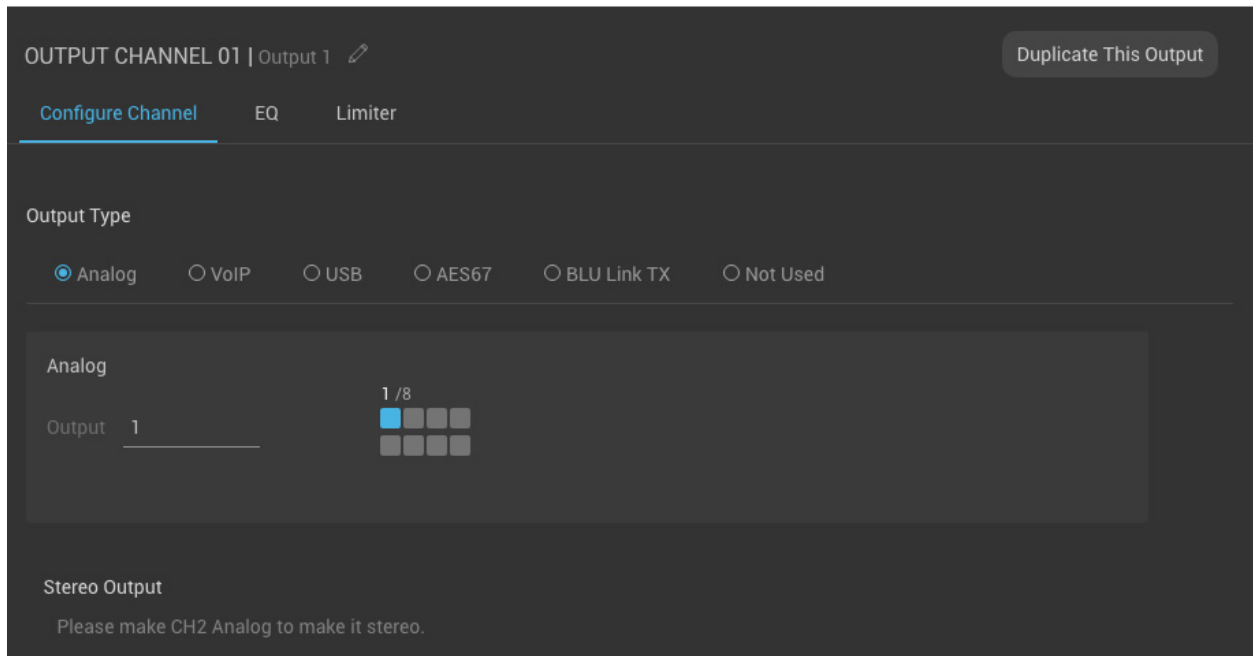
When required for the application, use this button to invert the output's polarity.

## Stereo Output Switch

Use this switch to stereo gang output channels. To stereo gang two channels, you must first configure both with matching **Output Type** selections (e.g., analog, VoIP, USB, etc.).



# Analog Output Type



## Output

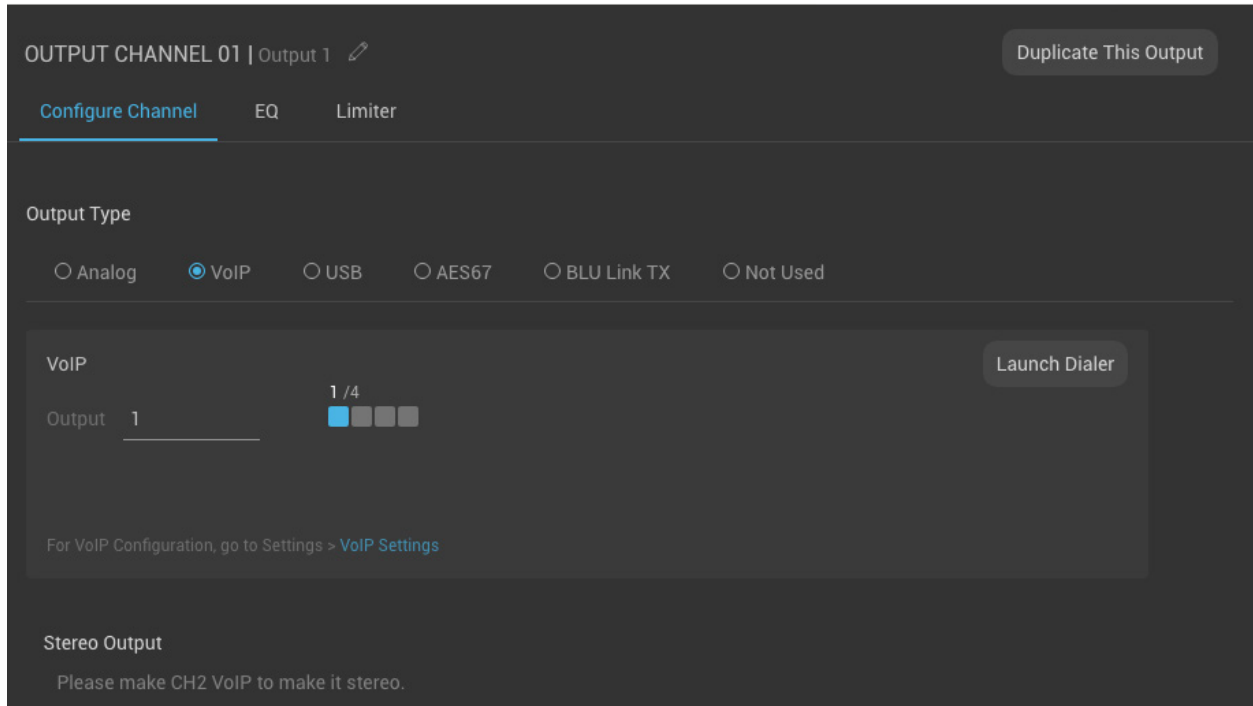
Use this field to assign an analog output channel.

## Channel Assignment Array

This assignment array displays an overview of the analog output channels assigned in the current configuration.



# VoIP Output Type



## Output

Use this field to assign a VoIP output line.

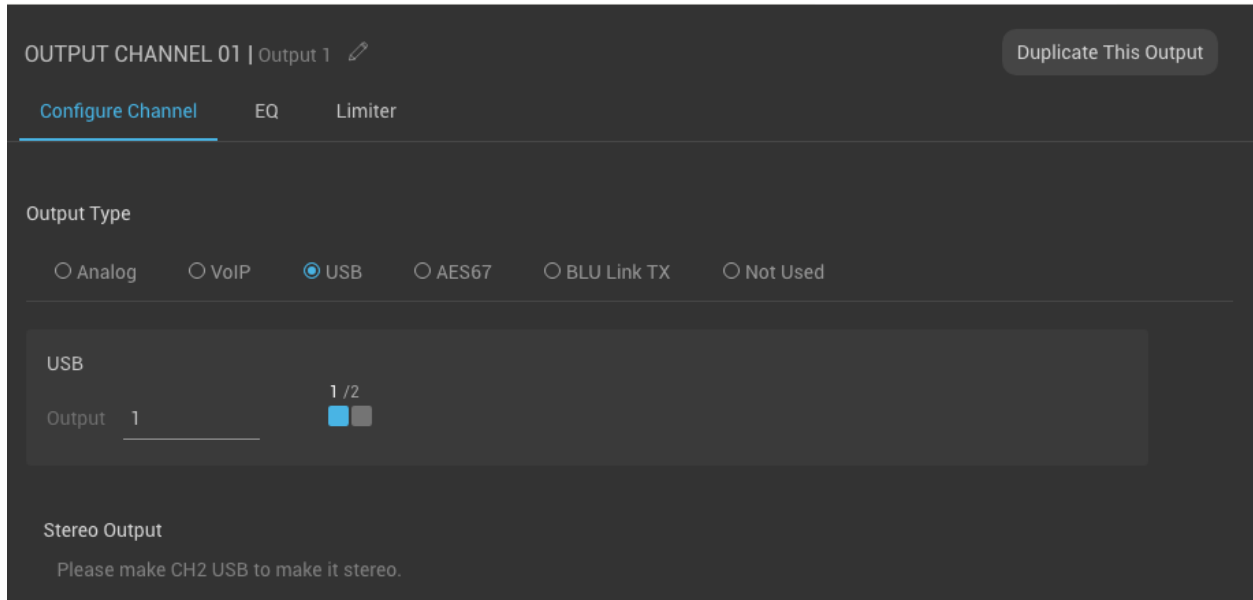
## Channel Assignment Array

This assignment array displays an overview of the VoIP output lines assigned in the current configuration.

## Launch Dialer Button

Press this button to open the [Dialer](#) for the selected VoIP output line.

# USB Output Type



## Output

Use this field to assign a USB output channel.

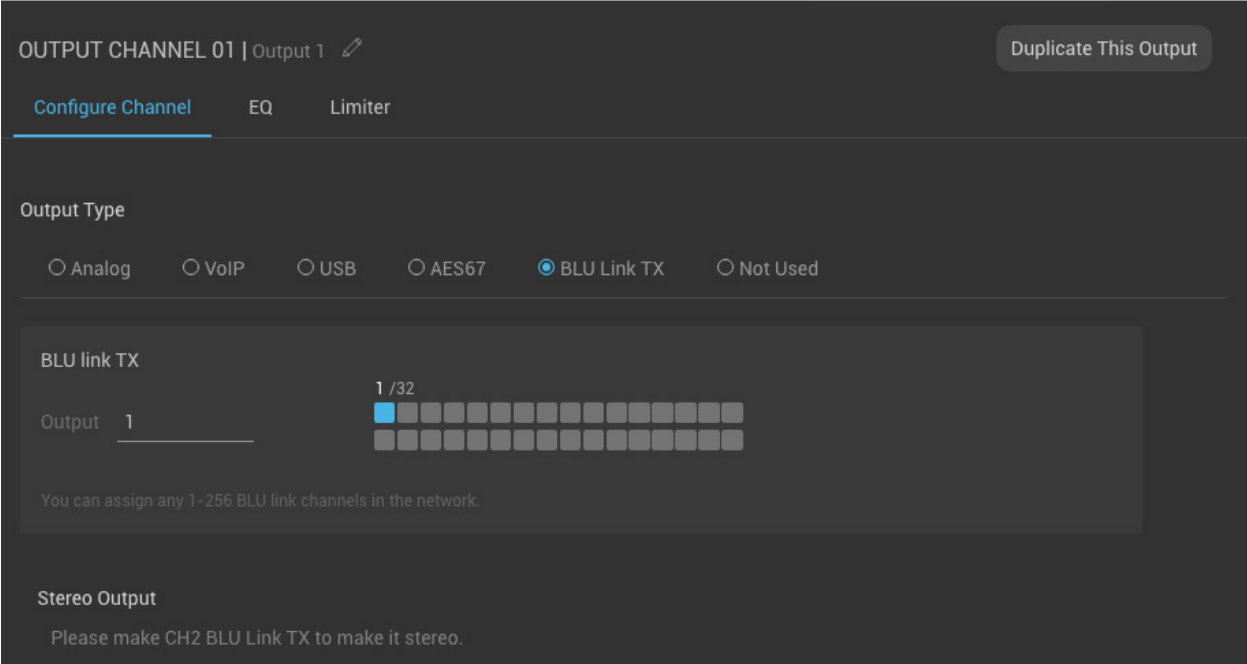
## Channel Assignment Array

This assignment array displays an overview of the USB output channels assigned in the current configuration.

# AES67 Output Type

This feature to be implemented in a future firmware release.

# BLU Link Output Type



## Output

Use this field to assign a BLU link output channel.

## Channel Assignment Array

This assignment array displays an overview of the BLU link output channels assigned in the current configuration.

# Output PEQ

The screenshot displays the 'Output PEQ' interface for 'OUTPUT CHANNEL 01 | Output 1'. It features a 'Duplicate This Output' button and navigation tabs for 'Configure Channel', 'EQ', and 'Limiter'. The EQ section is active, showing a frequency response graph with a gain of 2.29 dB. The graph has a logarithmic x-axis from 20 Hz to 20 kHz and a linear y-axis from -20 dB to 20 dB. Five filter bands are visible: HPF, 1, 2, 3, and 4. The 'Type' is set to 'Bell' and 'Flatten All' is a button. Below the graph are sliders for 'Frequency' (set to 189 Hz) and 'Width' (set to 1).

## On/Off Switch

This switch turns the PEQ on or off.

## Filter Band Selection

Use these buttons to select one of the five available bands for editing.

## **Type**

When "Filter 1" is selected, use this button to toggle between a bell or low shelf filter type. When "Filter 4" is selected, use this button to toggle between a bell or high shelf filter type.

## **Flatten All**

Press this button to flatten all bands. Press the button again to restore flattened bands.

## **Graphical Display**

Grab the numbered filter bands in this display to adjust gain and frequency settings graphically.

## **Gain**

Use this slider or field to adjust the gain of the selected filter band, with a range of +/- 20dB.

## **Frequency**

Use this slider or field to adjust the frequency of the selected filter band.

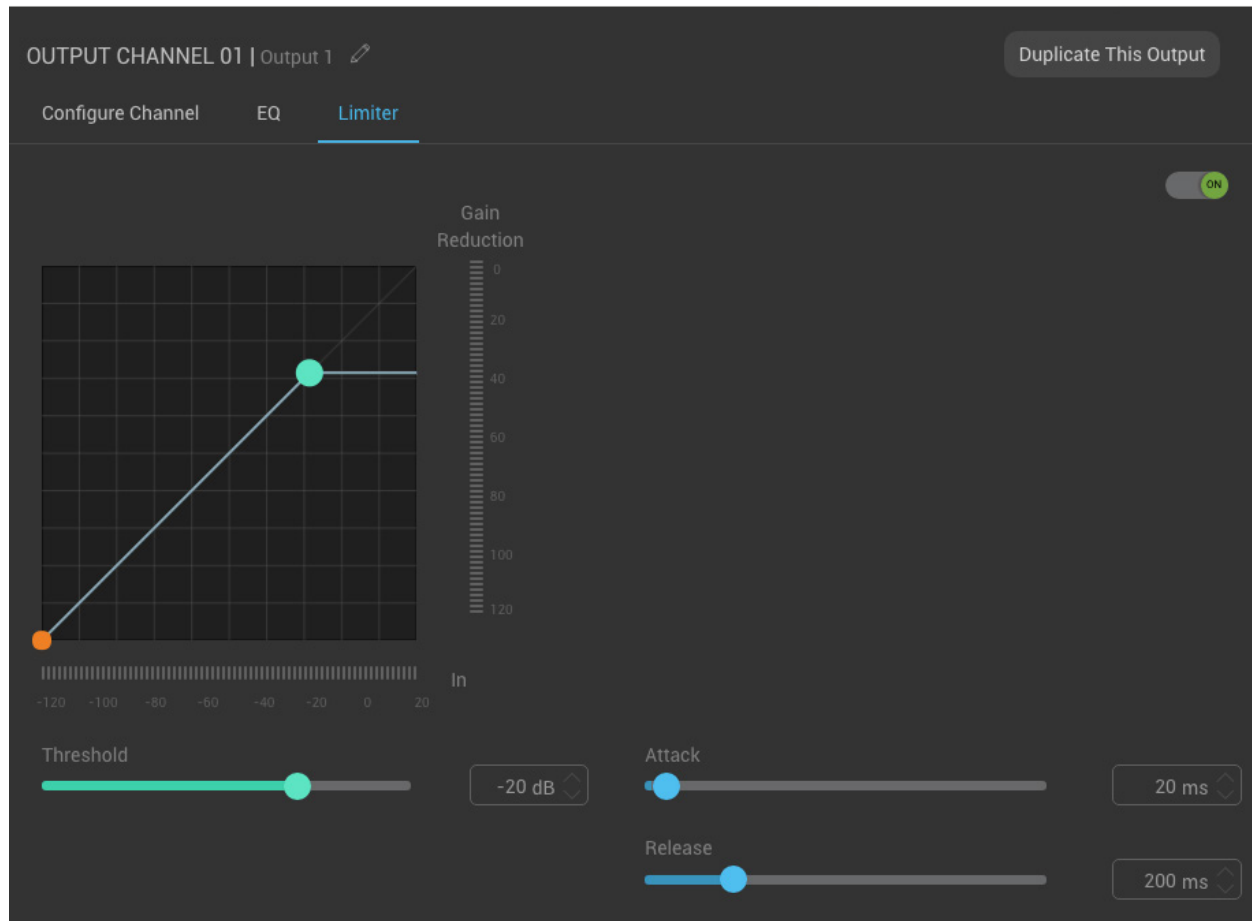
## **Width**

Use this slider or field to adjust the width of the selected filter band. Settings range from 0.1 to 128.

## **Invert Polarity**

When required for the application, use this button to invert the output's polarity.

# Limiter



## On/Off Switch

This switch turns the Limiter on or off.

## Graphic Display

Displays the limiting curve, with input level on the bottom and gain reduction on the right. Click and drag the cyan node to adjust the **Threshold** parameter of the Limiter. The orange node indicates signal level.

## Gain Reduction Meter

This meter displays the amount of gain reduction being applied to the signal.

## **In Meter**

This meter displays incoming signal level in dBu.

## **Threshold**

Use this slider or field to adjust the level at which limiting will occur.

## **Attack**

Use this slider or field to adjust how quickly the Limiter will clamp down on the signal. Lower values will clamp down on the signal quickly, whereas higher values will let some of the signal pass before clamping down on the signal.

## **Release**

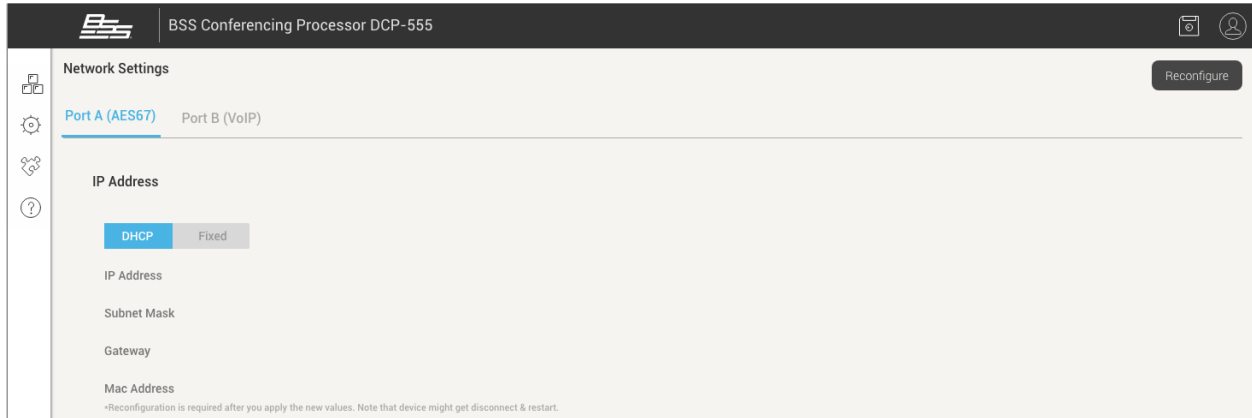
Use this slider or field to adjust how quickly the Limiter will release the limited signal once it falls back below threshold.



# Network Settings Port A (AES67)

This port carries the AES67 traffic and should be connected to the AES67 network.

**Note:** Both the A and B Ethernet ports carry the control data traffic for communication with the DCP-555 application and 3rd party control devices, and either can be used for DCP-555 control.



## Reconfigure Button

Once the desired edits have been made on the page, press this button to save the changes.

## DHCP/Fixed Buttons

"DHCP" is the default setting. This allows the DCP-555 to initially receive an IP address from a DHCP server. If the application requires a static IP address, click the **Fixed** button and manually enter the desired IP address settings.

## IP Address

If "DHCP" is selected, this field displays the IP address assigned to the DCP-555. If "Fixed" is selected, this field can be edited and the desired IP address can be entered.

## Subnet Mask

If "DHCP" is selected, this field displays the subnet mask assigned to the DCP-555. If "Fixed" is selected, this field can be edited and the desired subnet mask can be entered.

## **Gateway**

If "DHCP" is selected, this field displays the gateway assigned to the DCP-555. If "Fixed" is selected, this field can be edited and the desired gateway can be entered.

## **MAC Address**

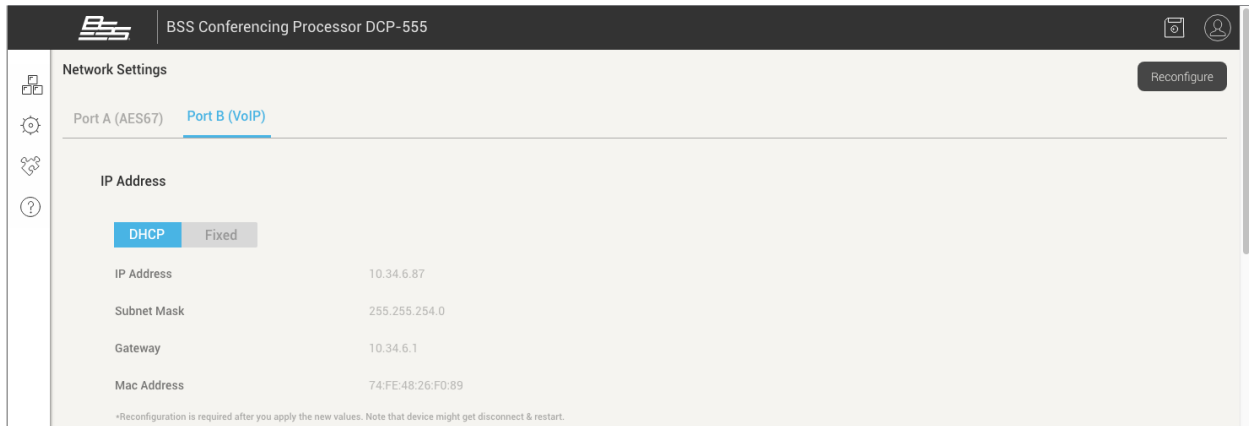
This field displays the MAC address of Ethernet port A.

**Note:** AES67 to be implemented in a future firmware release.

# Network Settings Port B (VoIP)

This port carries the VoIP traffic and should be connected to the VoIP network.

**Note:** Both the A and B Ethernet ports carry the control data traffic for communication with the DCP-555 application and 3rd party control devices, and either can be used for DCP-555 control.



## Reconfigure Button

Once the desired edits have been made on the page, press this button to save the changes.

## DHCP/Fixed Buttons

"DHCP" is the default setting. This allows the DCP-555 to initially receive an IP address from a DHCP server. If the application requires a static IP address, click the **Fixed** button and manually enter the desired IP address settings.

## IP Address

If "DHCP" is selected, this field displays the IP address assigned to the DCP-555. If "Fixed" is selected, this field can be edited and the desired IP address can be entered.

## Subnet Mask

## DCP-555 Help File

If "DHCP" is selected, this field displays the subnet mask assigned to the DCP-555. If "Fixed" is selected, this field can be edited and the desired subnet mask can be entered.

### **Gateway**

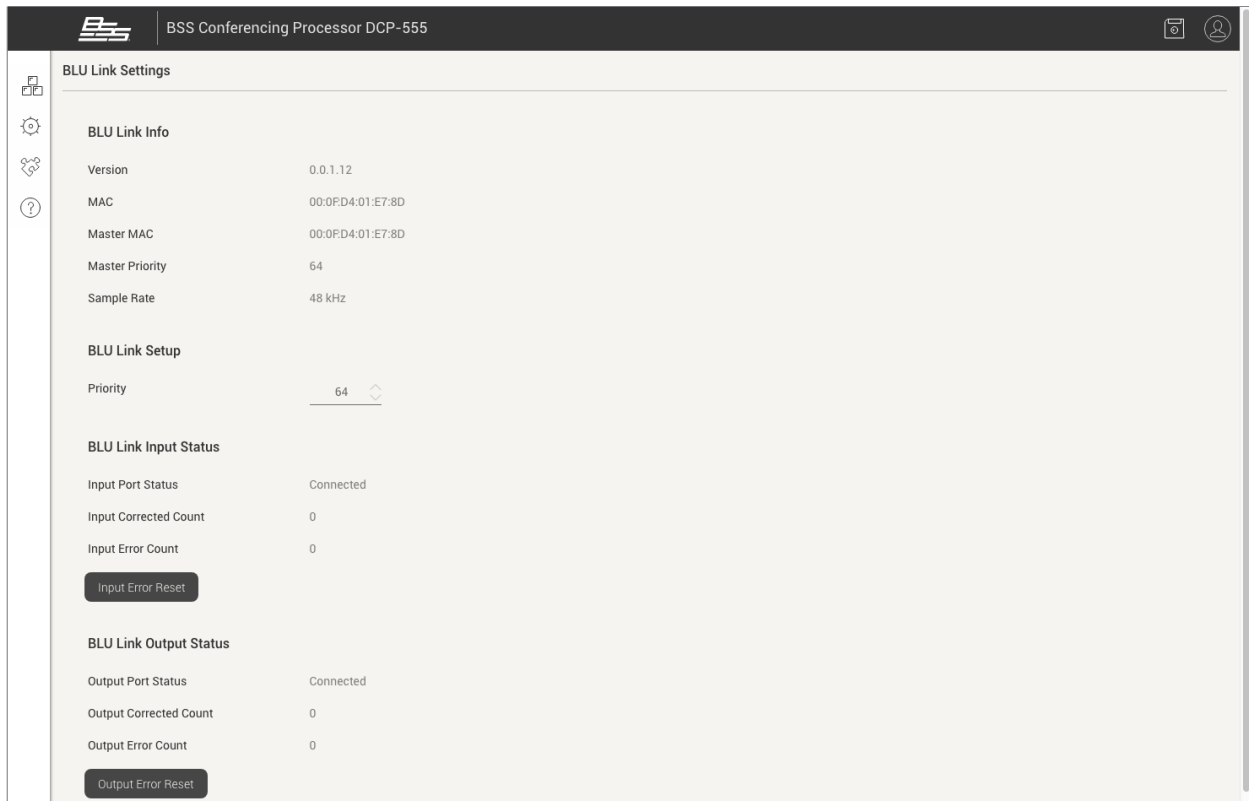
If "DHCP" is selected, this field displays the gateway assigned to the DCP-555. If "Fixed" is selected, this field can be edited and the desired gateway can be entered.

### **MAC Address**

This field displays the MAC address of Ethernet port B.

# BLU Link Settings

BLU link is a proprietary point-to-point digital audio bus that supports up to 256 audio channels at 48kHz or 128 audio channels at 96kHz. The DCP-555 supports 28x32 channels of BLU link audio at 48kHz.



## Version

This field displays the current firmware version of the DCP-555's BLU link hardware.

## MAC

This field displays the MAC address of the DCP-555's BLU link hardware.

## Master MAC

This field displays the MAC address of the device providing master clock for the BLU link bus.

## Master Priority

This field displays the Master Priority setting of the device providing master clock for the BLU link bus.

## Sample Rate

This field displays the DCP-555's BLU link sample rate setting.

**Note:** The DCP-555 has a fixed sample rate of 48kHz. All devices connected to the DCP-555 via BLU link must also be configured with a 48kHz BLU link sample rate.

## Priority

This field set's the DCP-555's Master Priority setting for the BLU link bus.

Each BLU link device contains a Master Priority setting, with a possible value between 0 and 254. Some BLU link devices have a fixed Master Priority setting, while others are configurable.

The device with the highest Master Priority setting on the bus will be configured as the BLU link master. If two BLU link devices have the same Master Priority setting, then each device's MAC address are compared to determine which will be the master, with the device with the higher MAC address winning.

If connecting to a BLU link bus that requires a certain device to be master, ensure that the master device has a higher Master Priority setting than all other BLU link devices on the bus.

## Input Port Status

This field indicates whether or not a valid BLU link connection has been made to the BLU link input port. Under normal operating conditions, "Connected" will be displayed. If no valid connection is made to the BLU link bus, "Not Connected" will be displayed. If "Error" is displayed, this indicates the BLU link input and output cables may be swapped, or the BLU link sample rate of the DCP-555 does not match the sample rate of other BLU link devices connected to the BLU link bus.

## Input Corrected Count

This field displays the number of BLU link input data errors corrected.

### **Input Error Count**

This field displays the number of BLU link input data errors that have occurred.

### **Input Error Reset Button**

Press this button to reset the Input Error Count and Input Corrected Count back to 0.

### **Output Port Status**

This field indicates whether or not a valid BLU link connection has been made to the BLU link output port. Under normal operating conditions, "Connected" will be displayed. If no valid connection is made to the BLU link bus, "Not Connected" will be displayed. If "Error" is displayed, this indicates the BLU link input and output cables may be swapped, or the BLU link sample rate of the DCP-555 does not match the sample rate of other BLU link devices connected to the BLU link bus.

### **Output Corrected Count**

This field displays the number of BLU link output data errors corrected.

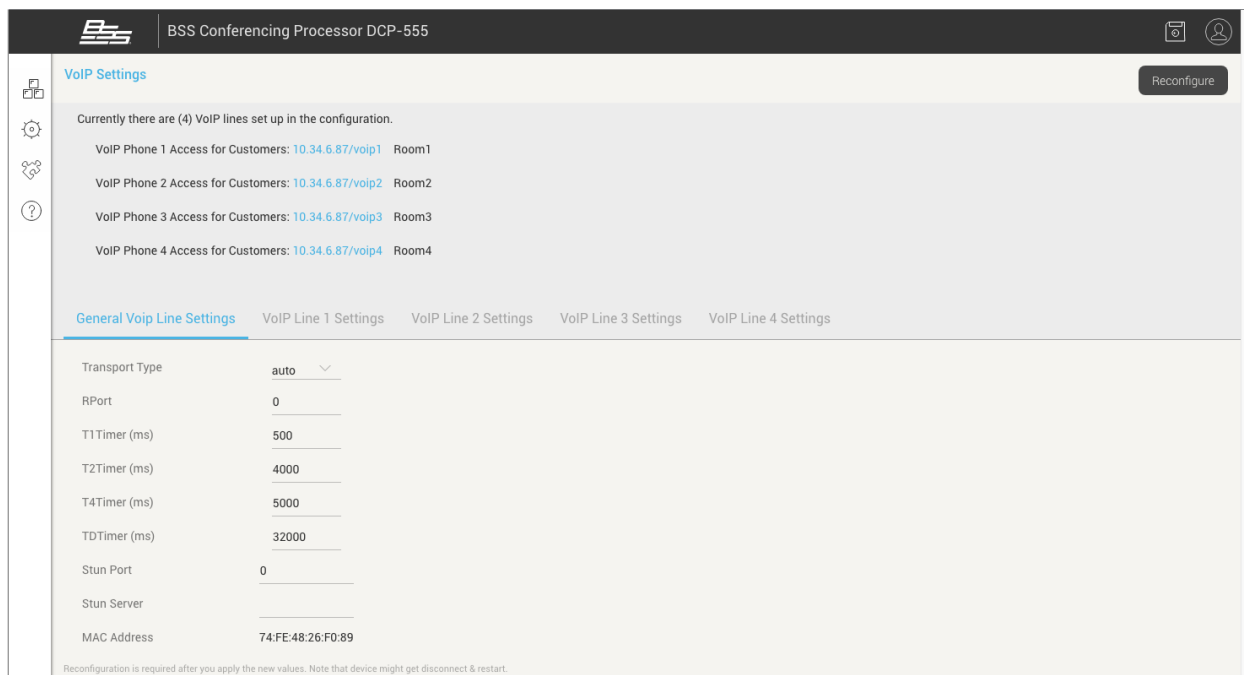
### **Output Error Count**

This field displays the number of BLU link output data errors that have occurred.

### **Output Error Reset Button**

Press this button to reset the Output Error Count and Output Corrected Count back to 0.

# General VoIP Line Settings



**VoIP Settings** Reconfigure

Currently there are (4) VoIP lines set up in the configuration.

- VoIP Phone 1 Access for Customers: [10.34.6.87/voip1](http://10.34.6.87/voip1) Room1
- VoIP Phone 2 Access for Customers: [10.34.6.87/voip2](http://10.34.6.87/voip2) Room2
- VoIP Phone 3 Access for Customers: [10.34.6.87/voip3](http://10.34.6.87/voip3) Room3
- VoIP Phone 4 Access for Customers: [10.34.6.87/voip4](http://10.34.6.87/voip4) Room4

**General Voip Line Settings** | VoIP Line 1 Settings | VoIP Line 2 Settings | VoIP Line 3 Settings | VoIP Line 4 Settings

Transport Type	auto
RPort	0
T1Timer (ms)	500
T2Timer (ms)	4000
T4Timer (ms)	5000
TDTimer (ms)	32000
Stun Port	0
Stun Server	
MAC Address	74:FE:48:26:F0:89

Reconfiguration is required after you apply the new values. Note that device might get disconnect & restart.

## VoIP Line Dialer URLs

The URLs for all configured VoIP lines are displayed at the top of the VoIP Settings page. Provide these URLs to the client for access to the [Dialer](#) for each configured meeting room.

## Reconfigure Button

After changing settings on the page, press this button to save the changes.

## Transport Type

This field sets the SIP protocol transport mode and provides the following options:

- Auto
- TCP
- TLS
- UDP Only

**Note:** By default this field is set to "Auto".



## RPort

This field sets the default port that RTP is using. The port number ranges from 0 (default) to 65535. Setting a value other than "0" for the RPort changes the SIP routing behavior so that responses can be received through a NAT, even if private addresses are used in the SIP headers. The Rport should only be changed if the transport is unreliable.

## T1, T2, T4, and TD Timers (ms)

Timer	Default	Description
T1Timer	500 ms	For unreliable transports, such as UDP, the client transaction retransmits requests at an interval that starts at T1 seconds and doubles after every retransmission. T1 specifies an estimate of the round-trip time (RTT). Nearly all of the SIP transaction timers scale with T1, and changing T1 adjusts their values.
T2Timer	4000 ms	Specifies the maximum retransmit interval for non-INVITE requests and INVITE responses. For unreliable transports, requests are retransmitted at an interval which starts at T1 and doubles until it reaches T2. If a provisional response is received, retransmissions continue for unreliable transports, but at an interval of T2. T2 represents the amount of time a non-INVITE server transaction takes to respond to a request if it does not respond immediately.
T4Timer	5000 ms	Specifies the amount of time the network takes to clear messages between client and server transactions.
TDTimer	32000 ms	Specifies the maximum duration a message will remain in the network.




**Note:** Do not change these timer settings unless you are well aware of their consequences. Contact the network administrator before changing these settings.

## MAC Address

## DCP-555 Help File

This field displays the MAC address of the DCP-555's VoIP port.

# VoIP Line (1-4) Settings

 BSS Conferencing Processor DCP-555  

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**VoIP Settings** Reconfigure

Currently there are (4) VoIP lines set up in the configuration.

- VoIP Phone 1 Access for Customers: [10.34.6.87/voip1](#) Room1
- VoIP Phone 2 Access for Customers: [10.34.6.87/voip2](#) Room2
- VoIP Phone 3 Access for Customers: [10.34.6.87/voip3](#) Room3
- VoIP Phone 4 Access for Customers: [10.34.6.87/voip4](#) Room4

General Voip Line Settings **VoIP Line 1 Settings** VoIP Line 2 Settings VoIP Line 3 Settings VoIP Line 4 Settings

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Launch Dialer and Volume Settings

### VoIP Line 1 - Protocol

Account Status	NotRegistered (0)
Username	<input type="text"/>
Authentication Name	<input type="text"/>
Password	<input type="text"/>
Display Name (Caller ID)	<input type="text"/>
Domain Name	<input type="text"/>
Stun Enabled	<input type="checkbox"/>
Register with Domain	<input checked="" type="checkbox"/>
Registration Life	<input type="text" value="3600"/>
Registration Retry	<input type="text" value="30"/>
Proxy Address	<input type="text"/>
Proxy Port	<input type="text" value="0"/>
Signaling Port	<input type="text" value="0"/>
SRTCP Preference	<input type="text" value="disabled"/>
RTP Port	<input type="text" value="23556"/>
RTP Port End	<input type="text" value="23556"/>

---

### VoIP Line 1 - Tones

Ring Type	<input type="text" value="classic"/>
Auto Answer Ring Count	<input type="text" value="Immediately"/>
DTMF Mode	<input type="text" value="SipInfo"/>
DTMF Tone Mute	<input type="checkbox"/>
DTMF Fixed Length	<input type="checkbox"/>
DTMF On Time (ms)	<input type="text" value="250"/>
DTMF Off Time (ms)	<input type="text" value="500"/>

---

### VoIP Line 1 - Timers

Session Timer	<input type="checkbox"/>
Default Session Timer (s)	<input type="text" value="300"/>
Minimum Session Timer (s)	<input type="text" value="90"/>

---

### VoIP Line 1 - NAT

Keep Alive	<input type="checkbox"/>
Keepalive Interval(s)	<input type="text" value="30"/>

Reconfiguration is required after you apply the new values. Note that device might get disconnect & restart.

**Version - 0.0.30 (15th May 2018)**

## **VoIP Line Dialer URLs**

The URLs for all configured VoIP lines are displayed at the top of the VoIP Settings page. Provide these URLs to the client for access to the [Dialer](#) for each configured meeting room.

## **Reconfigure Button**

After changing settings on the page, press this button to save the changes.

## **Launch Dialer and Volume Settings Button**

Press this button to access the [Dialer and Volume Settings](#) window, where you can test the selected VoIP line and configure volume levels.

## **Account Status**

This field displays the registration status of the VoIP account.

## **Username**

Enter the VoIP extension assigned to the user's VoIP line when signed up for VoIP service.

## **Authentication Name**

Usually, the username is sufficient to authenticate a VoIP account with the server. If the username and authentication name are different, specify the authentication name in this field.

## **Password**

Enter the password in this field to authenticate the VoIP account with the server.

## **Display Name (Caller ID)**

Enter the desired name that will be presented as the user's caller ID to the remote party.

## **Domain Name**

Enter the IP address or domain name of the VoIP server.

## **Stun Enabled Switch**

Enable this switch if the application requires connection to a STUN server. This switch is disabled by default

## **Register With Domain Switch**

When this switch is enabled, VoIP registration will include the name entered in the **Domain Name** field. This switch is enabled by default and should be left on if the VoIP service provider requires the domain name to register.

## **Registration Life**

This parameter determines the interval in seconds during which the VoIP line will attempt to re-register with the server. Note that the proxy may override this setting with a value of its own. If an acknowledgement has not been received from the proxy within the agreed time, the VoIP port registration information kept in the proxy's database will be cleared. The default registration expiration period is 3600 seconds and should be left at this value unless specified by the network administrator.

## **Registration Retry**

This parameter determines the interval in seconds after which the VoIP port tries to re-register when there is a registration failure.

## **Proxy Address**

Enter the proxy address for accessing VoIP service.

## **Proxy Port**

In this field, enter the network port of the VoIP phone that is used to communicate with the proxy server. The default proxy port is 5060 in VoIP systems, but this number can be modified if needed. Entering a value of "0" indicates any port can be used.

## Signaling Port

This port is used to direct incoming SIP traffic to the correct line for communications between the DCP-555 VoIP port and the proxy. This setting should be left at its default value unless specified by the network administrator.

## SRTP Preference

This field sets the preference for the Secure Real-time Transport Protocol. SRTP provides secure communication using encryption and message authentication. Options include "Disabled", "Optional", and "Required".

## RTP Port

This field changes the default port that RTP is using.

## RTP Port End

This field sets the last RTP port that will be used by the VoIP line. This parameter's range is from 0–65535.

## Ring Type

Ring type can be selected as "Classic" or "Silent".

## Auto Answer Ring Count

This parameter is used to automatically answer an incoming call either immediately or after a set number of rings. The Auto Answer button on the VoIP [Dialer](#) must be set to "ON" for this setting to have any effect. Options include "Immediately", "1 Ring", "2 Rings", and "3 Rings".

## DTMF Mode

Within a VoIP conversation, DTMF tones are delivered either in-band (as a beep), or out-of-band via SIP or RTP signaling messages.

The available options include:

- "None" – Disables DTMF support for both Tx and Rx.

- "In Band" – Incoming stream delivers DTMF signals in-audio independently of codecs; VoIP processor listens to the audio stream and will detect/extract the DTMF signals.
- "SipInfo" – This mode uses out-of-band signaling using SIP packets.

**Note:** The "SipInfo" option listed above is not recommended for DTMF delivery, since it cannot deliver strokes synchronously with the audio stream and will introduce timing artifacts.

### DTMF Tone Mute Switch

When this switch is "ON", the DTMF tone is not generated whenever a key on the [Dialer](#) keypad is pressed.

### DTMF Fixed Length Switch

When this switch is set to "ON", the DTMF tone's on time is fixed at 160ms and the **DTMF On Time** and **DTMF Off Time** settings are not applicable. When this switch is set to "OFF", the **DTMF On Time** and **DTMF Off Time** values can be set manually. Minimum on/off time is 50ms and maximum is 1000ms.

### DTMF On Time (ms)

This field sets the length of time at which DTMF tones will be generated. This is more applicable when the "In Band" **DTMF Mode** is selected. All other times, this is the length of time that the DTMF feedback tone is heard back on the speaker.

### DTMF Off Time (ms)

This field sets the pause length between DTMF tones.

### Session Timer Switch

When turned "ON", this switch enables periodic refresh of SIP sessions through a Re-INVITE or UPDATE request (referred to as session refresh requests). The Re-INVITES ensure that active sessions stay active and completed sessions are terminated.

When this switch is set to "OFF", the **Default Session Timer** and **Minimum Session Timer** fields will be disregarded.

If a call unexpectedly disconnects, disabling this option may help.

### **Default Session Timer (s)**

This field sets the default interval rate in seconds for SIP session refresh requests.

### **Minimum Session Timer (s)**

This field sets the lowest possible interval rate in seconds for SIP session refresh requests. This value resides in the 'Min-SE' header sent in the initial INVITE request. When a call is made, the presence of the Min-SE header informs the user agent server (UAS) and any proxies of the minimum value that the user agent client (UAC) accepts for the session timer duration.

### **Keepalive Switch**

When this switch is turned "ON", TCP keepalive functionality will cause TCP messages to be sent to the SIP server periodically for the sole purpose of keeping a NAT pinhole open. This is preferred over the standard SIP keepalive when the application may sleep as the OS sends the TCP keepalive packet.

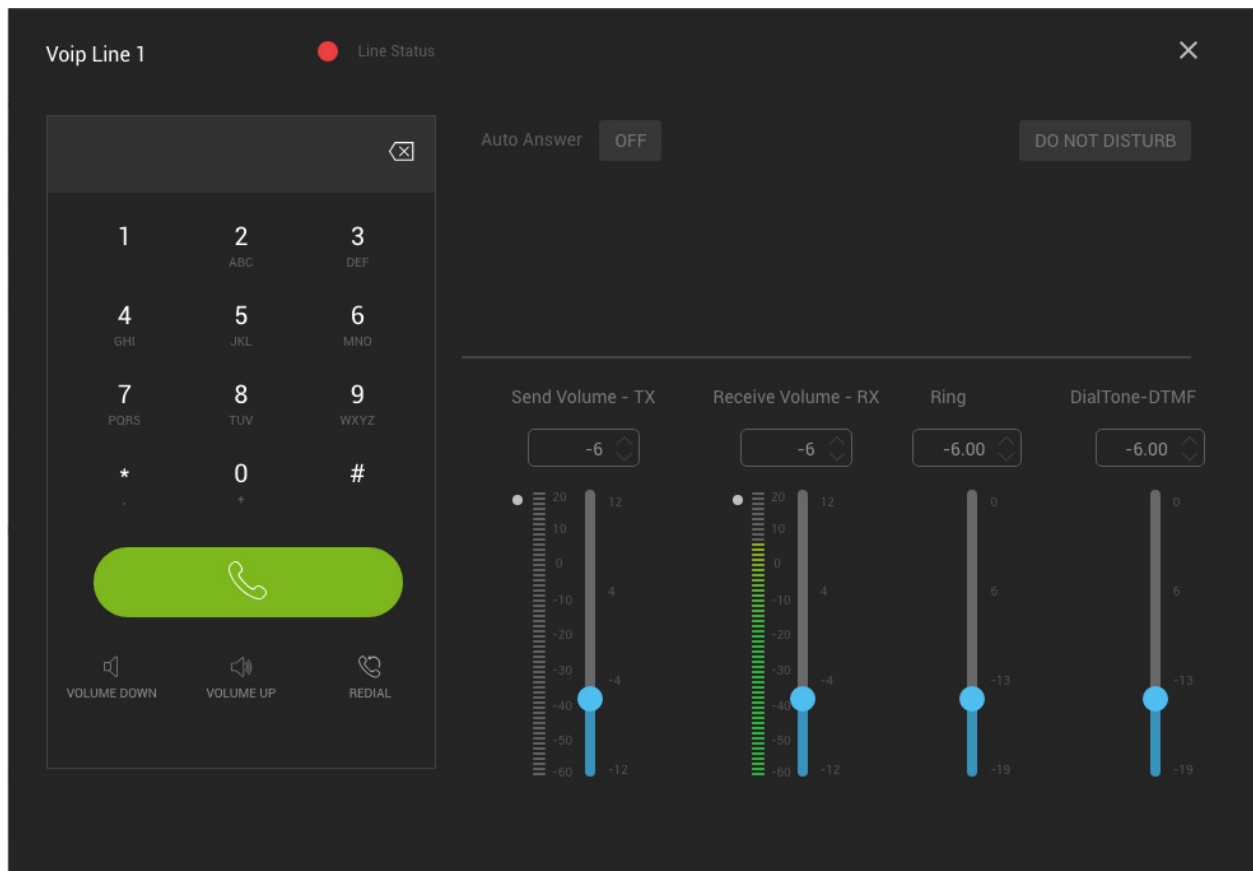
### **Keepalive Interval(s)**

When the **Keepalive** switch is "ON", this value determines the interval, in seconds, between two successive TCP keepalive messages.



# Dialer and Volume Settings

The screen shot below shows an example of the Dialer and Volume Settings window, which allows the installer to test the VoIP line and adjust levels. There are up to four Dialer and Volume Setting windows, one for each meeting room.



## Line Status Indicator

This indicator lights red to indicate the VoIP line is not connected and green to indicate it is connected.

## Telephone Keypad

This keypad provides the standard dialing keys for telephone operation:

- Telephone Display – This field displays the current telephone number being entered via the keypad buttons.
- Backspace – Use this button to remove previously entered characters.
- Keypad Buttons 0-9, \*, # – Use these buttons to enter the number to dial. If these buttons are pressed during a call, then a DTMF tone is played.

- Pick Up/Hang Up – The large green/red button with a telephone icon is the **Pick Up/Hang Up** button. It is used to take the VoIP line off hook and then place it back on hook again.
- Volume Up/Down – Use these buttons to adjust the level of the incoming audio of the VoIP interface.
- Redial – Press this button to redial the last dialed number.

### **Auto Answer On/Off Button**

Turn this button "ON" to allow incoming calls to be automatically answered by the system. The number of rings before the system auto-answers the line can be configured from the VoIP Line Settings window using the [Auto Answer Ring Count](#) field.

### **Do Not Disturb On/Off Button**

Enable this button to disallow incoming calls.

### **Send Volume - TX**

Use this field or slider to adjust the level of the outgoing audio. The meter displays the outgoing signal level in dBu. The dot to the left of the meter lights green to indicate signal presence or red to indicate clipping.

### **Received Volume - RX**

Use this field or slider to adjust the level of the incoming audio. The meter displays the input signal level in dBu. The dot to the left of the meter lights green to indicate signal presence or red to indicate clipping.

### **Ring**

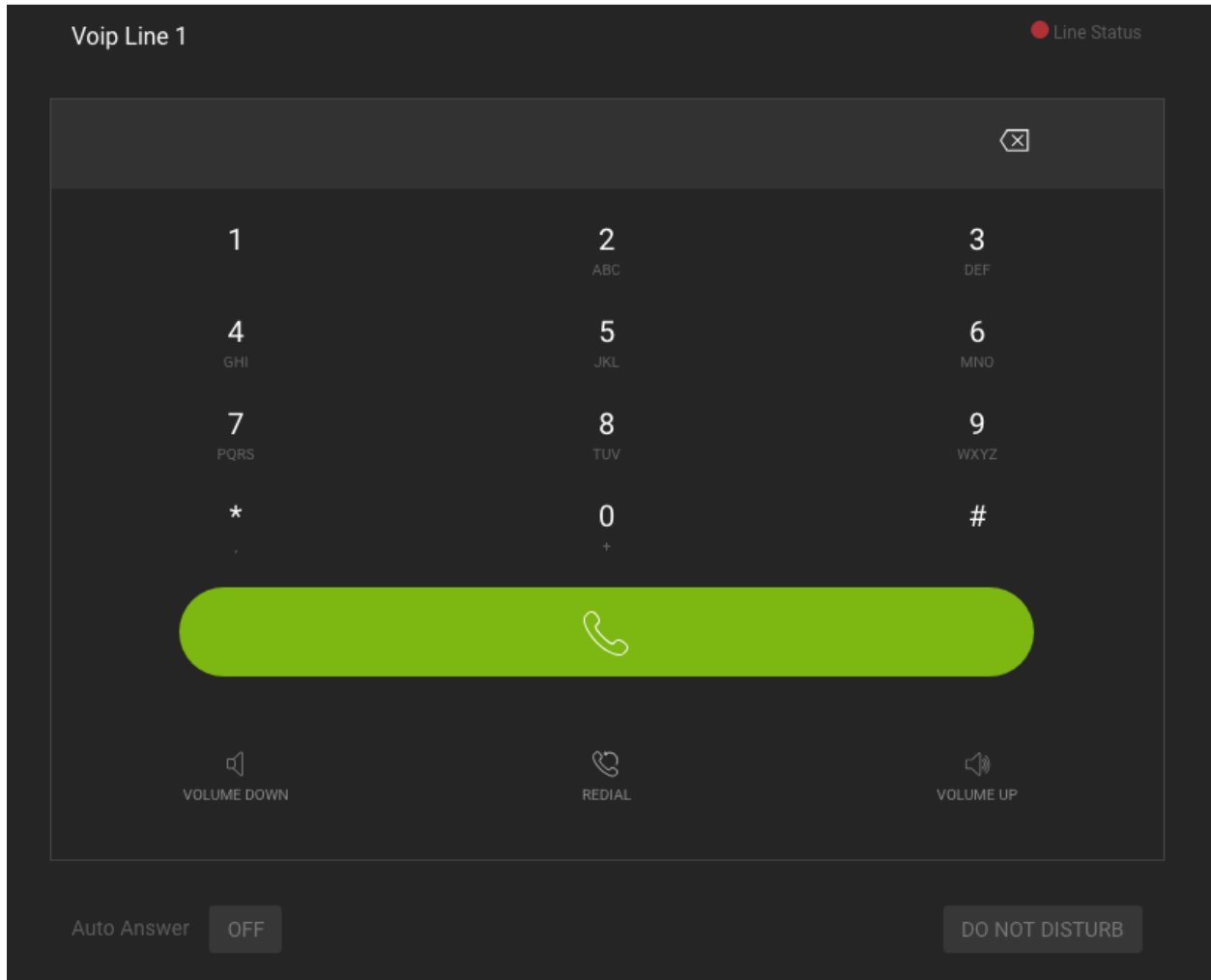
Use this field or slider to adjust the incoming call ring level.

### **DialTone-DTMF**

Use this field or slider to adjust the dialtone level.

# Dialer

The screen shot below shows an example of the Dialer window that the users will see. There are up to four Dialer windows, one for each meeting room.



## Line Status Indicator

This indicator lights red to indicate the VoIP line is not connected and green to indicate it is connected.

## Telephone Keypad

This keypad provides the standard dialing keys for telephone operation:

- Telephone Display – This field displays the current telephone number being entered via the keypad buttons.

- Backspace – Use this button to remove previously entered characters.
- Keypad Buttons 0-9, \*, # – Use these buttons to enter the number to dial. If these buttons are pressed during a call, then a DTMF tone is played.
- Pick Up/Hang Up – The large green/red button with a telephone icon is the **Pick Up/Hang Up** button. It is used to take the VoIP line off hook and then place it back on hook again.
- Volume Up/Down – Use these buttons to adjust the level of the incoming audio of the VoIP interface.
- Redial – Press this button to redial the last dialed number.

### **Auto Answer On/Off Button**

Turn this button "ON" to allow incoming calls to be automatically answered by the system. The number of rings before the system auto-answers the line can be configured from the VoIP Line Settings window using the [Auto Answer Ring Count](#) field.

### **Do Not Disturb On/Off Button**

Turn Do Not Disturb on to disallow incoming calls.

# GPI Ports

**BSS Conferencing Processor DCP-555**

**GPIO settings**

GPI Ports    GPO Ports    Relay

**GPI Port 1**

**Input Connected to :**  
 Contact Closure (Latching) ▾

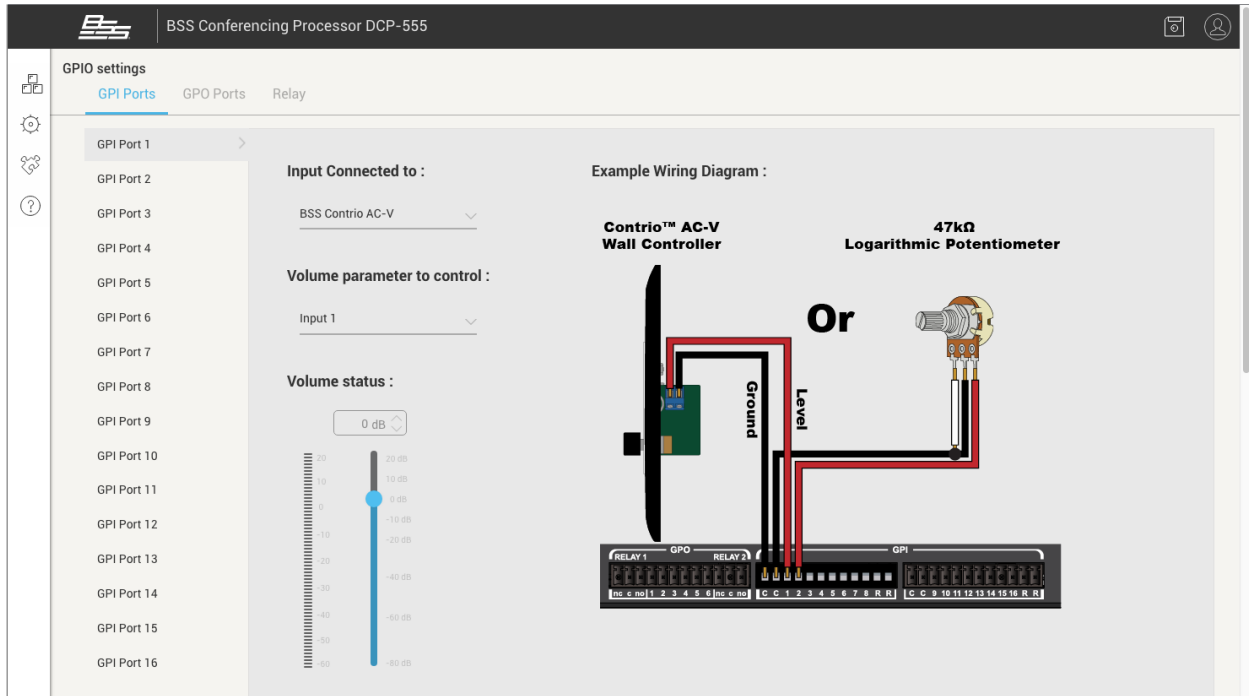
**Control Setting :**  
 Closed = Muted ▾

**Select parameter to control :**

- Mute - Input 1
- Mute - Input 2
- Mute - Input 3
- Mute - Input 4
- Mute - Conference Room 1
- Mute - Conference Room 2
- Mute - Conference Room 3
- Mute - Conference Room 4
- Mute - Output 1
- Mute - Output 2
- Tx Mute - VoIP Line 1
- Tx Mute - VoIP Line 2
- Tx Mute - VoIP Line 3
- Tx Mute - VoIP Line 4
- Off Hook - VoIP Line 1
- Off Hook - VoIP Line 2
- Off Hook - VoIP Line 3
- Off Hook - VoIP Line 4
- Mute All

**Example Wiring Diagram :**

The diagram shows a terminal block with three main sections: RELAY 1 (GPO), RELAY 2 (GPI), and GPI. A Latching Switch is connected to terminals 1 and 2 of the RELAY 2 section. A red line indicates the connection path from the switch to the terminal block.



## GPI Port 1-16

Use this list to select a GPI port to configure.

## Input Connected To

Use this field to select one of the following control options:

- BSS Contrio AC-V – Select this option to control volume via a BSS® Contrio™ AC-V Wall Controller or a 47kΩ potentiometer.
- Contact Closure (Latching) – Select this option to control muting for a channel using a latching contact closure switch.
- Contact Closure (Momentary) – Select this option for push-to-talk control of a channel using a momentary contact closure switch.
- None – Select this option to disable the selected port.

## Control Setting

Available only when the "Contact Closure (Latching)" or "Contact Closure (Momentary)" option is selected for the **Input Connected To** field above, use this field to select the state that matches the desired switch operation. The available options are "Closed = Muted" or "Closed = Unmuted".

### Select Parameter to Control

Available only when the "Contact Closure (Latching)" or "Contact Closure (Momentary)" option is selected for the **Input Connected To** field above, use this field to select the parameter that will be controlled via the switch connected to the associated GPI port. Options include all configured input and output channel mutes, conference room AutoMixer mutes, VoIP line Tx mutes, and mute all.

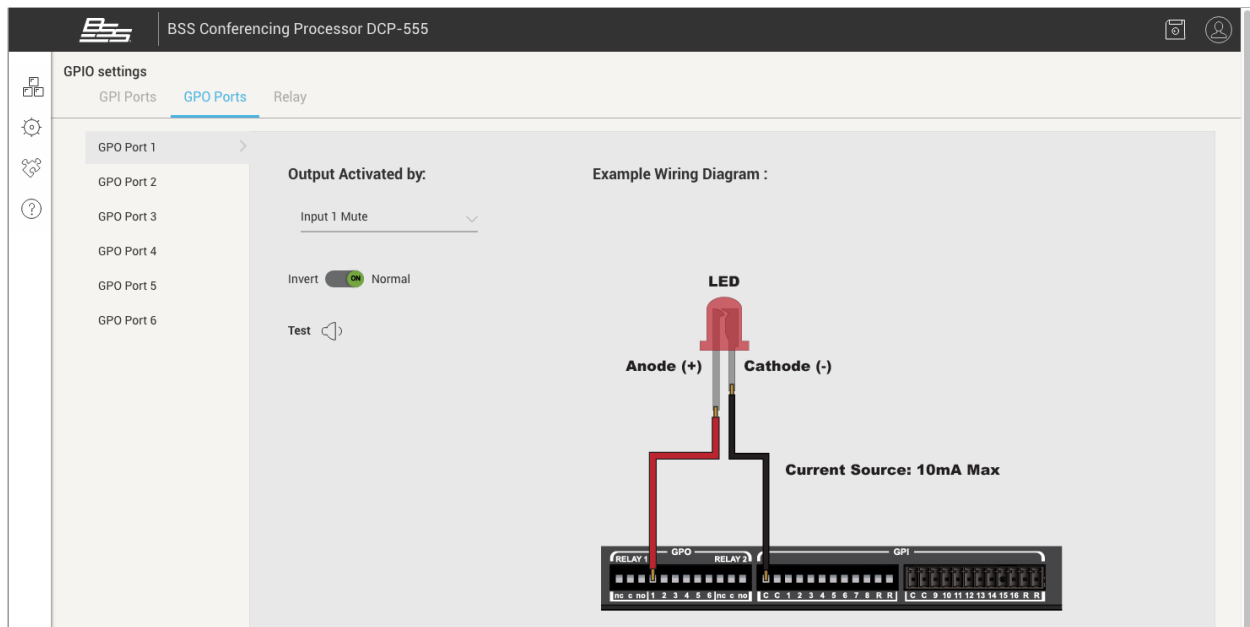
### Volume Parameter to Control

Available only when the "BSS Contrio AC-V" option is selected for the **Input Connected To** field above, use this field to select the volume parameter that will be controlled. Options include all configured input and output channels, as well as the conference room AutoMixers.

### Volume Status

Available only when the BSS Contrio AC-V option is selected for the **Input Connected To** field above, this field and slider will move when adjusting the volume control connected to the associated GPI port, allowing it to be tested for proper operation.

# GPO Ports



## GPO Port 1-6

Use this list to select a GPO port to configure.

## Output Activated By

Use this field to select the parameter that will control the associated GPO port. Options include all configured input and output channel mutes, conference room AutoMixer mutes, VoIP line Tx mutes, and VoIP line off hook status.

## Inversion Switch

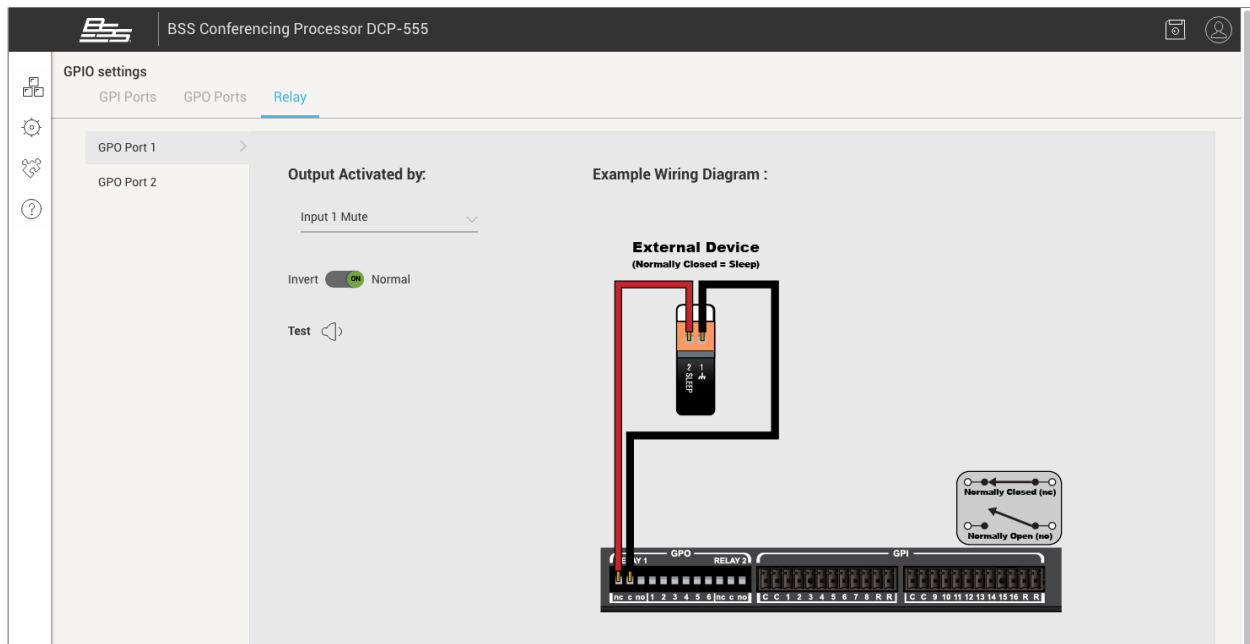
Use this switch to invert the output, thereby inverting the state of the connected LED.

## Test Button

Press this button to test the connected GPO port and LED for proper operation.



# Relay Ports



## Relay Port 1-2

Use this list to select a relay port to configure.

## Output Activated By

Use this field to select the parameter that will control the associated relay port. Options include all configured input and output channel mutes, conference room AutoMixer mutes, VoIP line Tx mutes, and VoIP line off-hook status.

## Inversion Switch

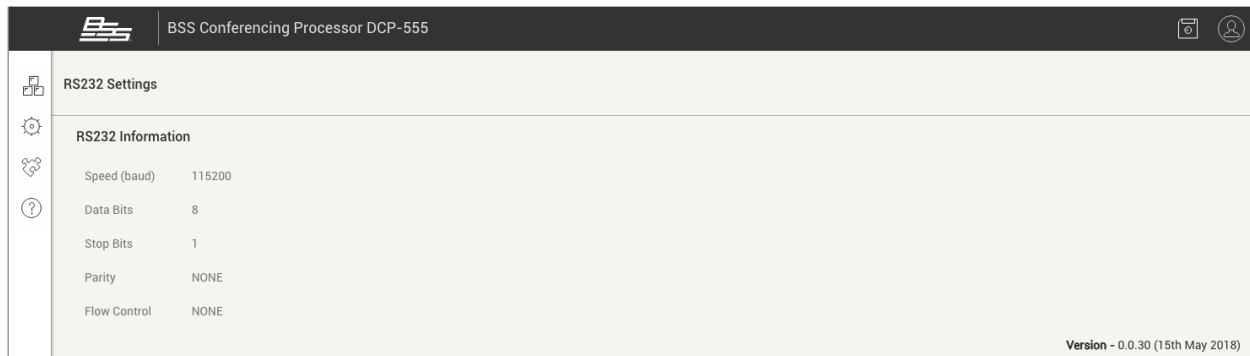
Use this switch to select the output polarity that matches the desired operation.

## Test Button

Press this button to test the relay port and connected device for proper operation.

# RS232 Settings

The RS232 port can be used to control the DCP-555 via an external control device (see [3rd Party Control](#) for more information). The DCP-555's RS232 settings are fixed as shown below.



## Speed (baud)

This field specifies the rate at which bits are transmitted for the serial interface.

## Data Bits

This field specifies the number of data bits to transmit over the serial interface.

## Stop Bits

This field specifies the number of bits used to indicate the end of a byte.

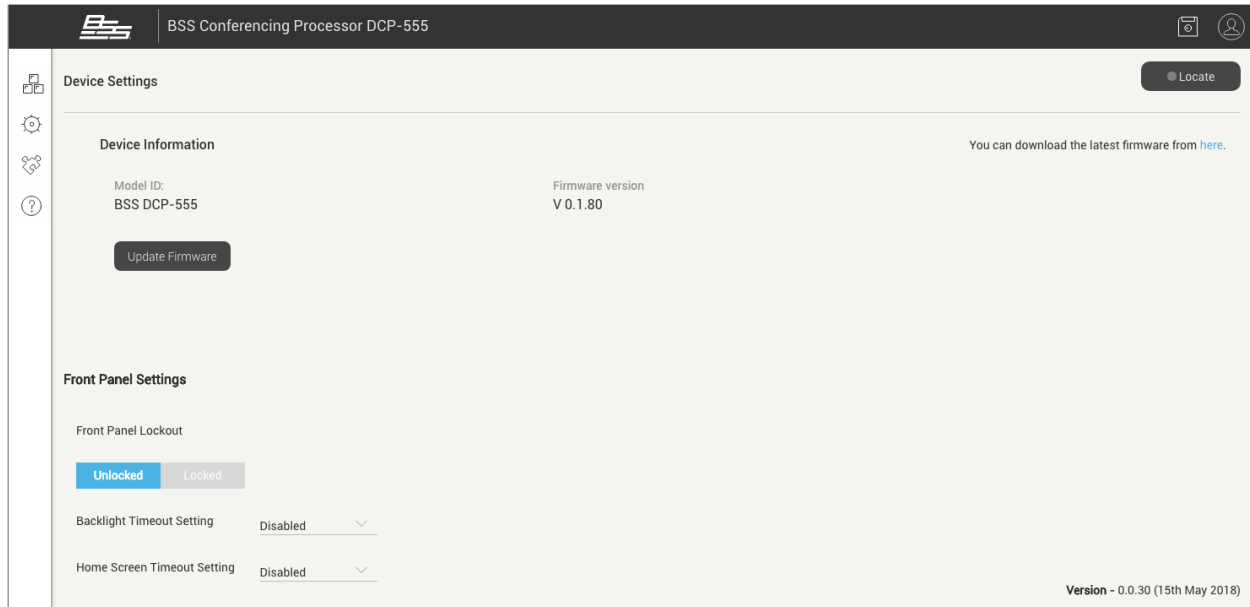
## Parity

This field specifies how to check parity bits in the data bits transmitted via the serial port.

## Flow Control

This field specifies the process of managing the data transmission rate of the serial port.

# Device Settings



## Locate Button

Pressing this button will cause the display to flash on the connected DCP-555. This allows confirmation of which device is being controlled when multiple DCP-555s exist on the network.

## Model ID

This field displays the model of the connected device.

## Firmware Version

This field displays the firmware version installed in the DCP-555.

## Update Firmware Button

Click this button to search online for any available DCP-555 firmware updates.

## Front Panel Lockout

Use these buttons to lock or unlock the front-panel controls.

### **Backlight Timeout Setting**

This field sets the amount of time it takes for the front-panel LCD screen backlight to time out after a period of inactivity. The available options are Disabled, 10 minutes, 5 minutes, 4 minutes, 3 minutes, 2 minutes, 1 minute, 30 seconds and 10 seconds.

### **Home Screen Timeout Setting**

This field sets the amount of time it takes to return to the front-panel home screen after a period of inactivity. The available options are Disabled, 10 minutes, 5 minutes, 4 minutes, 3 minutes, 2 minutes, 1 minute, 30 seconds and 10 seconds.

# 3rd Party Control

The DCP-555 supports 3rd party control and monitoring via RS232 or Ethernet. For information on the DCP-555 control protocol, see the DCP-555 Control API Guide located at <https://bssaudio.com/en/products/dcp-555#documentation>.



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