



DSP Control Center 3

Reference Manual



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Chapter 1 Overview

1-1 Minimum Hardware Requirements

Processor:	1 GHz
Memory:	512MB
Disk Space:	20MB
Operating System:	Windows Vista, Windows 7, Windows 8.1 or Windows 10
Software:	Microsoft .NET Framework 4.0 or higher

1-2 Installation Procedure

To get the latest copy of the DSP Control Center Software, go to

<http://www.stewartaudio.com/DSPCC>

After downloading the latest copy of the installer, launch the installer and follow the on-screen installation instructions. Refer to **Figure 1-1**.



Figure 1-1 – Installation screenshots

During the installation process, the Driver Installation Wizard will install the necessary device drivers. Refer to **Figure 1-2**.



Figure 1-2 – Driver Installation Wizard

1-3 Software Updates

The software will automatically check for updates upon program launch. If an update is found it will be downloaded and installed upon the next program launch. If no update is found, a message stating that you have the most recent version is displayed. Refer to **Figure 1-3**.



Figure 1-3 – Startup form displaying software update check

Chapter 2

User Interface

2-1 Startup Window

The startup window (**Figure 2-1**) is displayed when the DSP Control Center is launched. If the computer is connected to the Internet, the software automatically checks for updates at this time. (See **Section 1-3** for more information on update procedures.) Select the device type you wish to configure from the pull-down menu. Click "Launch" to open to configuration interface for the selected device type.



Figure 2-1 – Startup window

2-2 Device Configuration Window

When a device type has been selected and launched from the startup window, its Device Configuration window appears. Each Device Configuration window is unique to that particular device’s DSP design. Refer to **Figure 2-2** for an example Device Configuration Window.

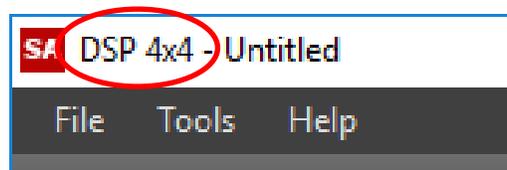


Figure 2-2 – Example device configuration window for DSP 4x4

The Device Configuration Window presents the Processing Blocks (see **Chapter 3**) in the same configuration and order as they exist in the processing signal path.

2-2-1 Title Bar

The title bar displays the device type of the active Device Configuration window.



2-2-2 Menu Bar

2-2-2-1 File Menu

Save Configuration <Ctrl + S>

Saves the current configuration to its current name. If the current configuration has not yet been given a name, the “Save As...” dialog will appear. Files are saved with the extension .scfg, (“Stewart Audio Configuration”).

Open Configuration <Ctrl + O>

Opens the Windows file browser to select a previously saved configuration file to open. If the current configuration has been modified, a Windows “Save” dialog will open to provide the option of saving the current configuration to a file before loading the selected configuration file.

Exit

Closes the Device Configuration Window. If changes have been made to the configuration, a prompt with an option to save the configuration before exiting is displayed.

2-2-2-2 Tool Menu

Open Device Manager

Opens the Device Manager for connecting or disconnecting to/from a physical device (see **Connecting to a Device in section 2-3**).

2-3 Connecting to a Device

Connecting to a device assists the user in the configuration process by providing real-time configuration with signal meter feedback. The Device Manager is opened by clicking the Connect button on the tool bar or from selecting “Open Device Manager” from the top Device menu.

When the Device Manager (**Figure 2-3**) is opened, all connected devices (of the selected type or model) are automatically discovered and populated in the Devices dropdown menu, with their serial number and firmware version.

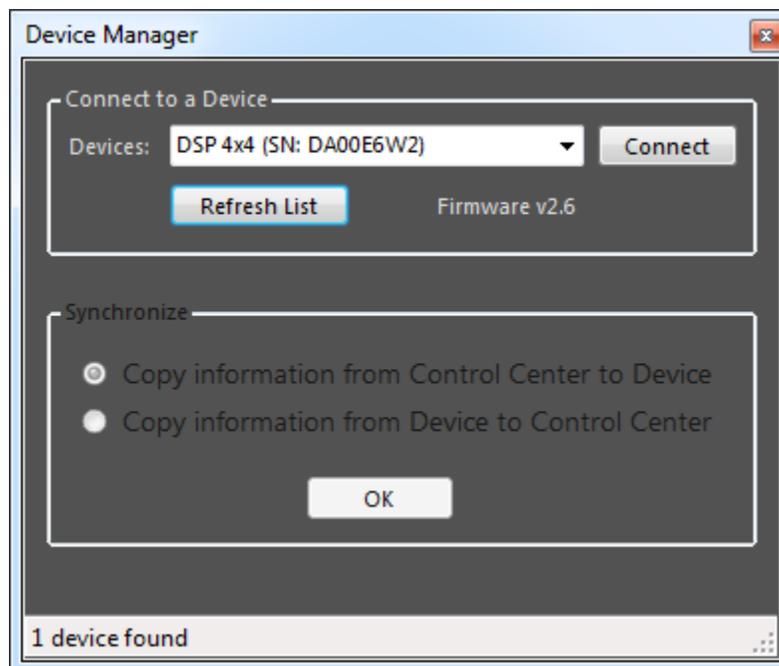


Figure 2-3 – Device Manager window

When the Device Manager is initially connected to a physical device, the configuration in the DSP Control Center and the configuration of the physical device must be synchronized. The configuration must be either copied from the DSP Control Center to the device or copied from the device to DSP Control Center (see **Figure 2-4**). Note that this will overwrite either the configuration on the device or in the DSP Control Center.



Figure 2-4 - Device Manager window after connecting to a device

After a connection is established and the device has been synchronized (the synchronization process can take up to 1 minute), various features in the Device Configuration Window are enabled. These include:

- Realtime configuration updates
- Signal metering (**Section 2-4**)
- Live preset changes (**Section 2-5**)

To disconnect from the device, press the Disconnect button in the Device Manager or in the Device Configuration Window toolbar. **Do not disconnect from the device while data is being transferred.**

2-4 Signal Metering

Signal metering (**Figure 2-5**) is utilized throughout the audio signal path to monitor live-audio RMS signal levels in dBV at the selected point in the processing signal path. The meter is capable of displaying signal levels between +35dB and -35dB. It should be noted that since all processing blocks have +24dB of headroom, any signal higher than +24dB on the signal meter will be clipped.

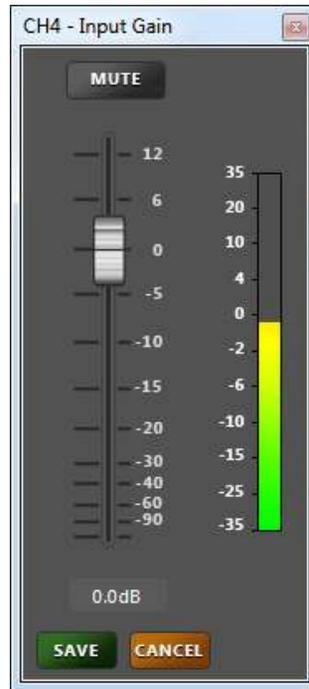


Figure 2-5 – Example signal meter in gain block

2-5 Presets

Presets are stored device configurations which can be recalled during device operation using RS-232 commands. Presets are configured via the Preset Selection dropdown menu on the device's main configuration screen (**Figure 2-6**). Selecting a preset from the dropdown updates the DSP Control Center user interface to the currently selected program.

In Connected mode, the physical device is configured with the selected program preset. A window is displayed briefly while the device loads the parameters.

There are 10 presets for the DSP4X4 and FLX models (numbered 0 through 9); there is only one preset for the DSP100 models.



Figure 2-6 – Program Selection dropdown

Chapter 3

Processing Blocks

Stewart Audio's DSP products include signal processing chains comprised of various processing blocks. This chapter documents each processing block including its purpose, user interface, and configuration properties.

Section 3-1 – Input Configuration

Section 3-2 - Gain

Section 3-3 - Compressor/Limiter

Section 3-4 - Filters

Section 3-5 - Ducker

Section 3-6 - Delay

Section 3-7 - Output Configuration

Section 3-8 - Mixer

3-1 Input Configuration

3-1-1 Purpose

The Input configuration block configures an input’s display property, pre-gain level, and phantom power (if available). This processing block should be configured first as adjusting pre-gain levels will have a dramatic effect on subsequent processing blocks further down the signal path.

3-1-2 User Interface

The user interface for emulation and live input configuration is shown in **Figure 3-1**.

Display Name is an editable textbox that accepts all alphanumeric characters as well as a limited number of special characters. The display name will appear on the input label as well as in mixer matrices (if included in the device).

Input Type is a dropdown menu populated with hardware-specific pre-gain levels. These pre-gain levels include an input device type as well as gain level in decibels (dB). All processing blocks have 24dB of headroom so selecting a pre-gain higher than 0dB will adjust the upper limit of the first gain block to compensate for this loss of headroom.

Phantom Power is a checkbox that is visible only if the device currently being configured supports phantom power on that channel. Refer to the device’s hardware manual for phantom power characteristics including voltage and current. **Note** – this checkbox will be automatically un-checked if a Line Level pre-gain is selected to prevent harm to devices that do not support phantom power. It will display a warning if phantom power is selected after it has been unchecked.

Signal Meter displays the current RMS level in dBV of the audio signal including both pre-gain and input pre-amplification (applied in the Input Configuration block). For configuration of the Input, refer to **Section 3-1**.

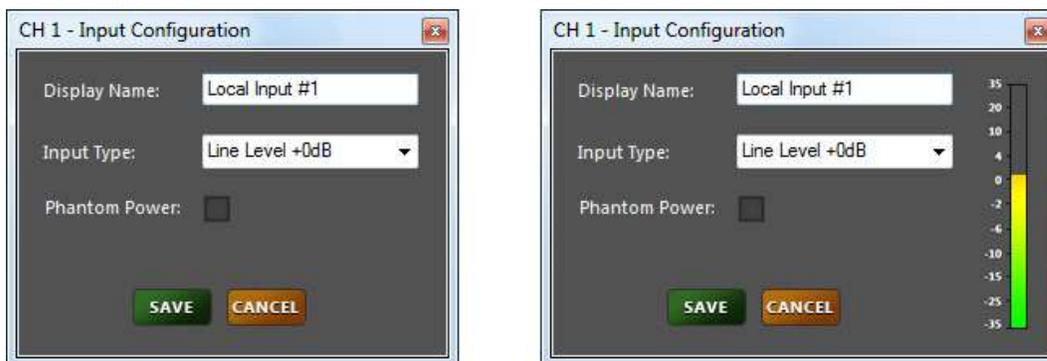


Figure 3-1 – Input configuration in emulation and connected modes

3-2 Gain

3-2-1 Purpose

The Gain block configures and monitors the gain level at various places along the processing signal path. The two types of gain blocks available are Gain and Trim blocks. Gain blocks will allow signal gain of +12dB per block (with 24dB of system headroom) while Trim blocks will only allow attenuation of the signal.

3-2-2 User Interface

The user interface for gain configuration is shown in **Figure 3-2**.

Gain is adjusted using the fader control similar to the interface on an analog mixing console. When moved up and down the textbox below the fader control will update to reflect the current gain setting. The gain slider is disabled when the block is muted.

Mute is an on/off button that will turn red when active. The gain fader will be disabled and >100dB of attenuation will be applied to the signal. A red indicator will appear on the gain block button in the main Device Configuration Window when that block is muted.

Signal Meter displays the current RMS signal level in dBV of the audio signal at the output of the gain block. Refer to **Section 2-4**.

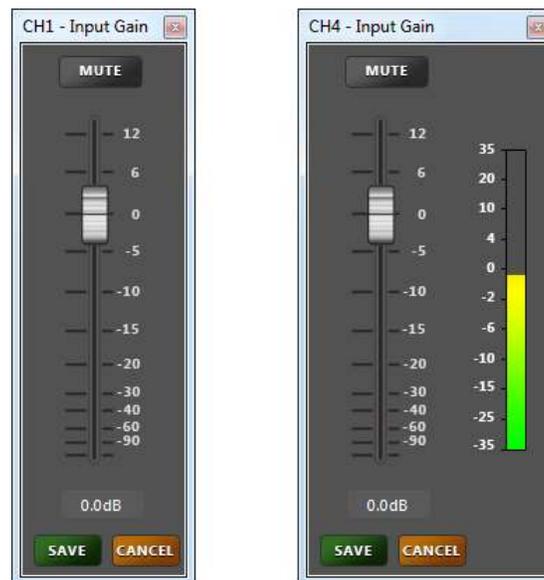


Figure 3-2 – Gain block in emulation and connected modes

3-3 Compressor/Limiter

3-3-1 Purpose

The Compression block allows configuration of the dynamic compressor/limiter available on most DSP processing channels. Audio compression is a specialized function used to reduce dynamic range - the span between the softest and loudest sounds. The use of compressors can make recordings or live-sound mixes sound better musically by controlling maximum levels and maintaining higher average loudness. Limiters are derived from a compressor and use a ratio of 100:1 to provide a fixed upper signal level.

3-3-2 User Interface

The user interface for emulation and live compressor configuration is shown in **Figure 3-3**.

Threshold is a decibel property that can be adjusted by dragging the yellow marker tracking the equality line on the graph. Alternatively, it can be adjusted using the numeric up-down control below the graph. The threshold is the level of the incoming signal at which the compressor amplifier changes from a unity gain amplifier into a gain reducing compressor. The compressor has no effect on the signal below the threshold level setting. Once threshold is reached, the compressor starts reducing gain based on the amount by which the signal exceeds the threshold and according to the ratio control setting. Threshold level could be thought of as the "sensitivity" of the compressor and is expressed as a specific level in dB.

Ratio is property available only in a standard compressor (not limiter) that can be adjusted by dragging the right-most yellow marker on the graph. Additionally, it can be adjusted used a similar up-down control as the threshold. Ratio is a way to express the degree to which the compressor is reducing dynamic range. Ratio is expressed as the change in output level for a change in input level above the threshold. A ratio of 10:1 specifies an increase of 10 dB above that threshold at the input results in a 1dB increase in the output level. Ratio is a constant value which is independent of the degree of compression taking place; the ratio of the input change to output change is constant.

Soft Knee is an option selected using the checkbox to the left of the bypass button. The level at which the compressor starts gain reduction is called the knee. A Hard Knee is a sudden change from uncompressed to compressed signal once the threshold has been reached. A Soft Knee applies a gradual level of compression and is a less obtrusive change from unity gain amplifier to compressor. Note that a soft knee will result in a small amount of compression starting around 6dB below the threshold.

Attack is a time property that is adjusted using the blue dial of the compressor configuration window. Attack time refers to the time it takes the compressor to start compressing after the threshold level has been reached. Attack time affects the sound quality in terms of overall perceived brightness or high-frequency content. With

very fast attack time settings, the compressor will activate very quickly, reducing gain instantly at the waveform level.

Release is a time property that is adjusted using the orange dial of the Compressor Configuration Window. Release time is the time the compressor uses to return to unity gain after the input signal has fallen below threshold. It should be noted that very fast attack or release times will distort low-frequency sounds, as the compressor is capable of gain change within one period (the 360-degree cycle of the lowest fundamental frequency) of the sound's waveform.

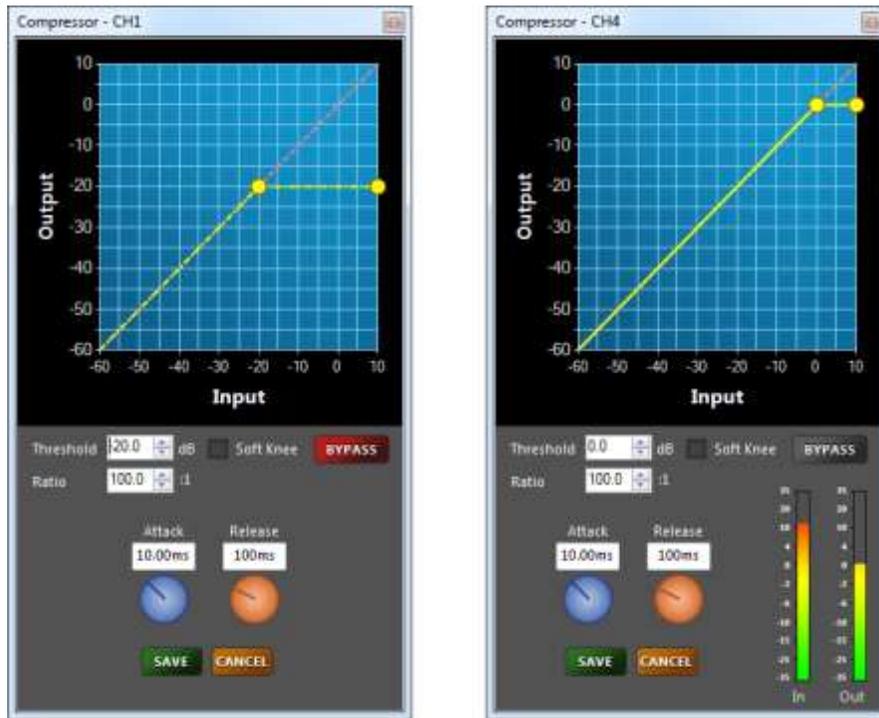


Figure 3-3 – Compressor block in emulation and connected modes

3-4 Filters

3-4-1 Purpose

The Filter block is used to configure the parametric filters available on each channel. Parametric filters, which come in a variety of types, alter the frequency response of an audio signal using configurable parameters. These filters can be used to provide cleaner sounding audio, reduce feedback, and protect speakers.

3-4-2 User Interface

The Filter Designer interface is shown in **Figure 3-4**.

Filters are configured using either the pulldown or textbox controls in the lower portion of the filter interface or by manipulating the graphical display in the upper portion.



Figure 3-4 – 3 channel filter designer interface

3-4-2-1 Configuration Using Standard Controls

The standard controls of the Filter Designer (see **Figure 3-5**) use pulldown menus and textbox controls. This interface is convenient when accurate configuration of gain or center frequency is required for matching speaker specifications. The available controls will vary depending on the currently selected filter type.

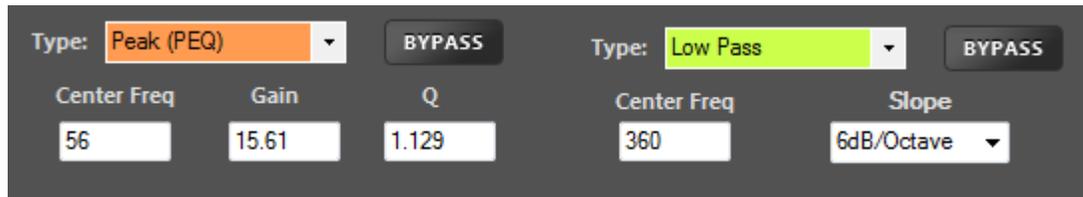


Figure 3-5 - Standard filter designer controls

Filter Type is a pulldown menu that selects the filter type or disables the filter (selecting Not Used).

Bypass is a button that enables or disables the filter bypass. When bypassed, the button will turn red and the filter response curve on the graphical interface will become dotted and no longer be included in the cascaded filter.

Center Frequency is the frequency in Hz at which the filter is centered.

Gain is the gain or attenuation level of the filter in dB. This is not available on all filter types.

Slope is applicable to high/low pass filters and is the rate at which the filter rolls off above or below the cutoff frequency.

Q controls the bandwidth (affected frequency range) of various filter types. A higher Q represents a narrower bandwidth.

3-4-2-2 Configuration Using Graphical Interface

The graphical interface of the Filter Designer provides a visual representation of the response curves for both the individual filters as well as the cascaded filter output. Individual filter response is represented by a thin line color-matched to the filter selector in the lower standard filter interface. The cascaded output is represented by a thicker yellow line. Bypassed filters are displayed with a dotted line and are not included in the cascaded output.

A filter is selected for editing on the graph by using the Filter Select (see **Figure 3-6**) buttons in the lower left-hand portion of the graphical interface. Only filters that are active are displayed in the filter select. Clicking on a filter number will display the drag markers for the active filter.



Figure 3-6 - Filter Selection

When a filter is selected, drag markers (see **Figure 3-7**) appear for that filter to allow configuration of center frequency, gain, and bandwidth (Q).

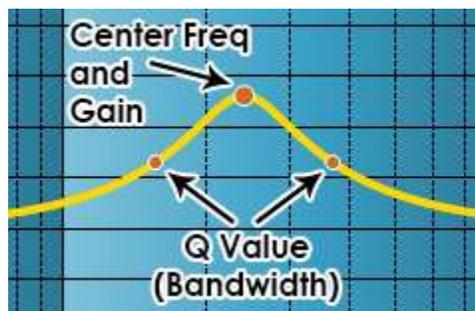


Figure 3-7 - Drag markers on single filter

To check values on the cascaded filter frequency response, click anywhere on the graph (other than on drag markers) and red crosshairs will appear on the cascaded filter (see **Figure 3-8**). Moving the mouse horizontally will drag the crosshairs along frequency axis displaying the gain level at the corresponding frequency.

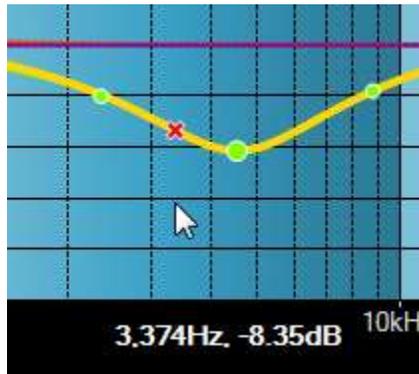


Figure 3-8- Cascaded filter crosshairs

3-5 Ducker

3-5-1 Purpose

The Ducker allows prioritization of one audio channel over other channels. Applications include prioritizing microphone audio or pages over other channels of program audio.

3-5-2 User Interface

The user interface for ducker configuration is shown in *Figure 3-9*.

Priority Channel is a dropdown to select the channel which is given priority and will activate the ducking (attenuation) of all other channels when the signal level of the priority channel reaches the set threshold. The dropdown menu will display the name set in the channel's input configuration.

Channel Enables select the channels that will be ducked when the priority channel signal level reaches the threshold. Only the channels where the checkbox is checked (green) will be attenuated by the ducker.

Threshold is a decibel property that can be adjusted using the numeric up-down control. The threshold is the level of the priority channel signal at which the ducker will attenuate non-priority channel signals. Threshold level could be thought of as the "sensitivity" of the ducker and is expressed as a specific level in dB.

Depth is a decibel property that can be adjusted using the numeric up-down control. The depth is the amount of attenuation to be applied to non-priority channels once the priority channel signal has reached the threshold.

Attack is a time property that is adjusted using the blue dial of the ducker configuration window. Attack time refers to the time it takes the ducker to start attenuating non-priority channels after threshold has been reached on the priority channel.

Release is a time property that is adjusted using the orange dial of the ducker configuration window. Release time is the time the ducker uses to return non-priority channels to unity gain after the priority channel signal has fallen below threshold *and* the hold time has elapsed.

Hold Time is a time property that is adjusted using the green dial of the ducker configuration window. Hold time is the time that the ducker will hold attenuation of non-priority channels after the priority channel signal has fallen below the threshold. This time allows for pauses between spoken words or audio tracks on the priority channel. Increase this time if non-priority signals are momentarily cutting through the priority channel during brief pauses in audio.

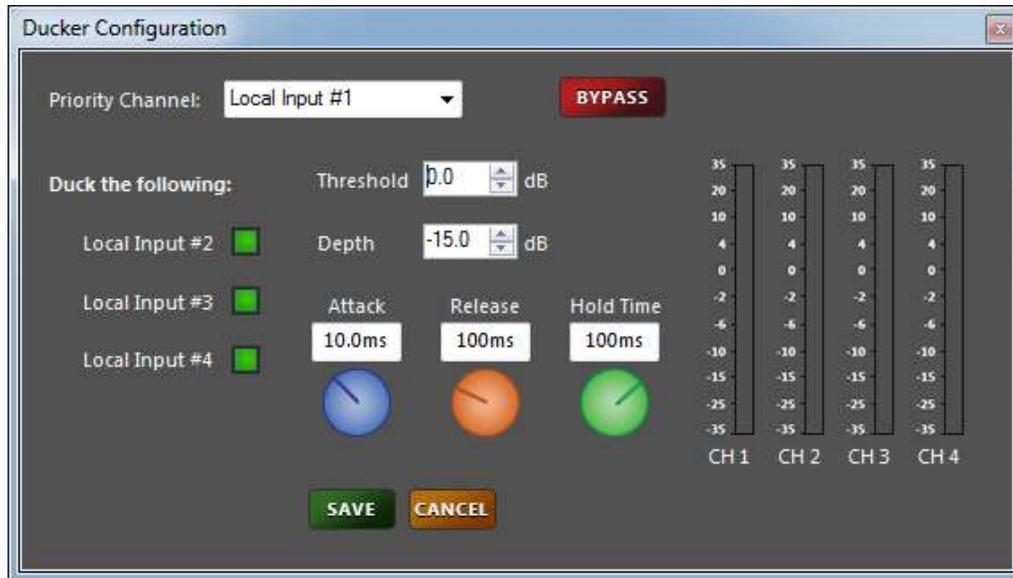


Figure 3-9 – Ducker configuration

3-6 Delay

3-6-1 Purpose

The Delay block delays (in time) a channel's output signal. Typically audio is selectively delayed to ensure that the sound emanating from speakers at differing physical distances from the listener all arrive at the listener at the same time. **Note that there is no delay block on Dante-enabled models.**

3-6-2 User Interface

The user interface for delay configuration is shown in **Figure 3-10**.

Time is a dial that controls the amount of time in milliseconds the source is delayed from the input to the output. The upper limit is 90msec for the DSP4x4. All other devices have an upper limit of 80msec. Dante networked models do not have delay. Changing the time dial will cause all distance dials to update to reflect an equal delay value.

Distance is a dial that controls the amount of time the source is delayed from the input to the output by utilizing the speed of sound at 70 °F (343 m/s or 1,126 ft/s). Changing a single distance dial will cause all other dials to update to reflect an equal delay value.

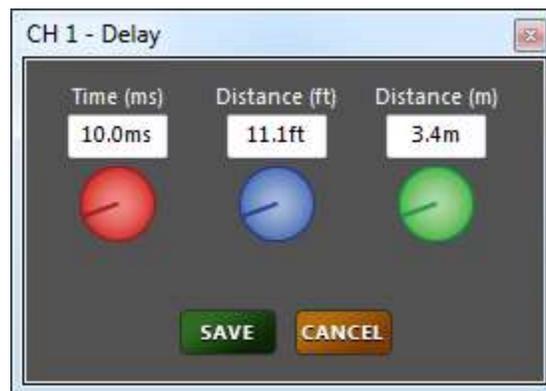


Figure 3-10 – Delay block

3-7 Output Configuration

3-7-1 Purpose

The Output Configuration block allows configuration of an output’s display properties and, if available on the device, RS-232 volume control.

3-7-2 User Interface

The user interface for output configuration is shown in **Figure 3-11**.

Display Name is an editable textbox that accepts all alphanumeric characters as well as a limited number of special characters. When edited, the display name will update on the output label.

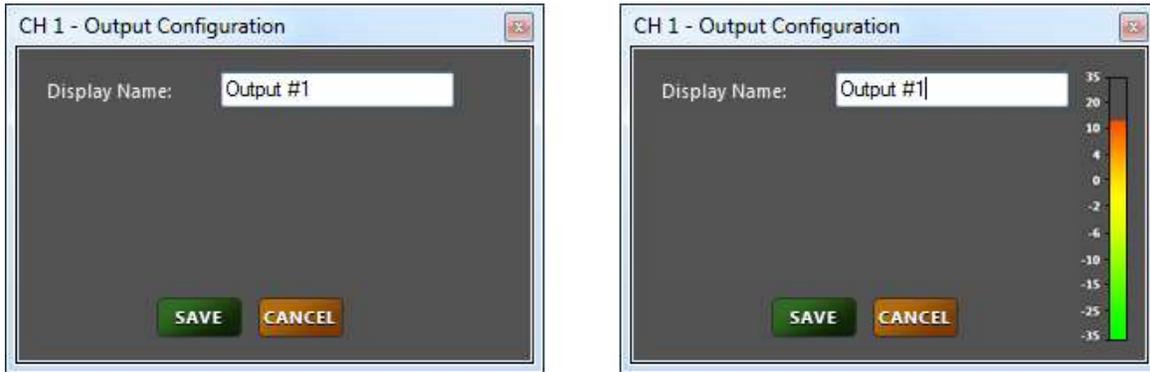


Figure 3-11 – Output configuration in emulation and connected modes

3-8 Mixer

3-8-1 Purpose

The mixer block allows the configuration and monitoring of the matrix mixer which mixes or routes input signals and signal generators to the output busses.

3-8-2 User Interface

The user interface for mixer configuration is shown in **Figure 3-11**.

Mixer Matrix consists of small round buttons that will open the gain block control (see **Gain**) corresponding to the appropriate input-to-output connection. The inputs are on the left-hand side of the matrix while the outputs are listed on the top. Muted crosspoints will display as red while unmuted crosspoints display as blue.

Signal Generators are included on the input list in order to provide signals for testing purposes. The generators include a 1kHz sine wave and pink noise generated at 0dB.

Signal Meter displays the current RMS signal level in dBV of the mixed audio output. Refer to **Section 2-4**.

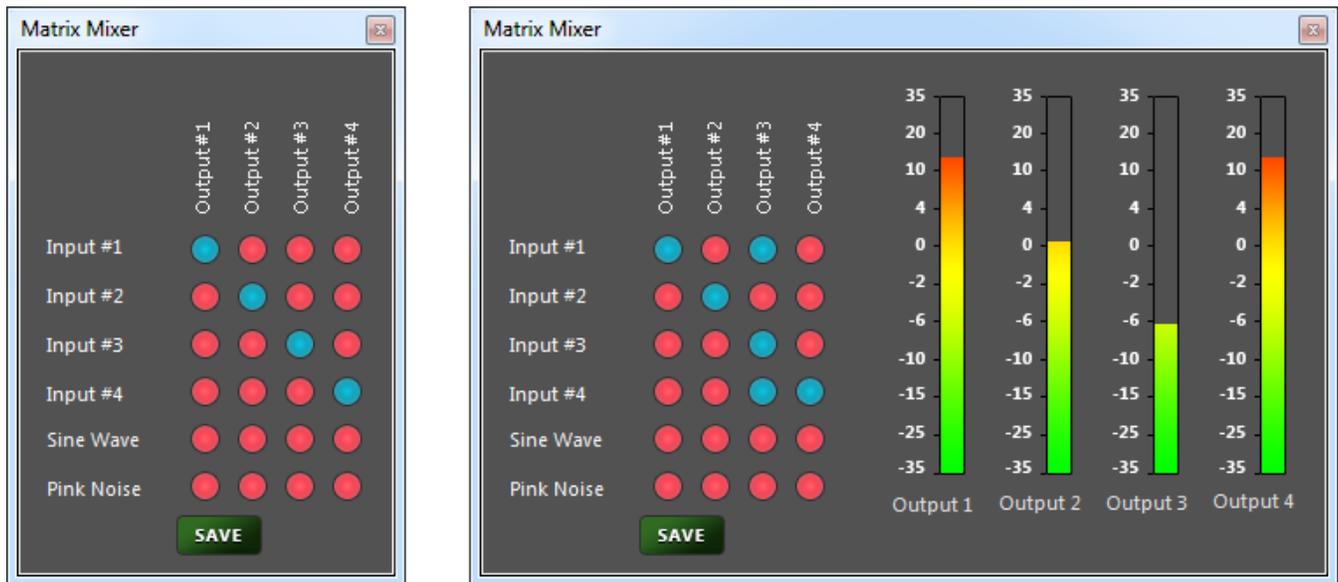


Figure 3-12 – Mixer in emulation and connected modes

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