

PE Series DSP Input Option (1 & 2)

Overview:

PE Series DSP Input Cards are add on card that replaces the standard Input Card shipped with PE Amps or can be installed by the factory. The Protea^{ne} DSP Option Cards have a SHARC 200MHz 32-bit floating-point processor and will operate at 48 or 96kHz with a 24-bit AD/DA. Both the XLR and Euroblock versions offer two additional processed output channels to drive additional amplifier channels. Standard Ethernet protocol (auto or manual IP configuring) is used to control everything. Both analog and digital (AES3) inputs are provided. Protea^{ne} DSP cards have an extensive DSP library including: a full set of filter types including graphic, parametric, notch, shelving and high/low/allpass, Bessel, Butterworth and Linkwitz/Riley crossover filters up to 48dB/octave as well as time delays, limiters, gates, signal generators (white, pink noise, sinewave) and a matrix mixer. All DSP functions use a “Click and Hot-Plug” user interface to place processor blocks in the signal chain. Contact closure terminals achieve provisions for 3 preset recall, DC remote attenuators and remote standby/power-down.

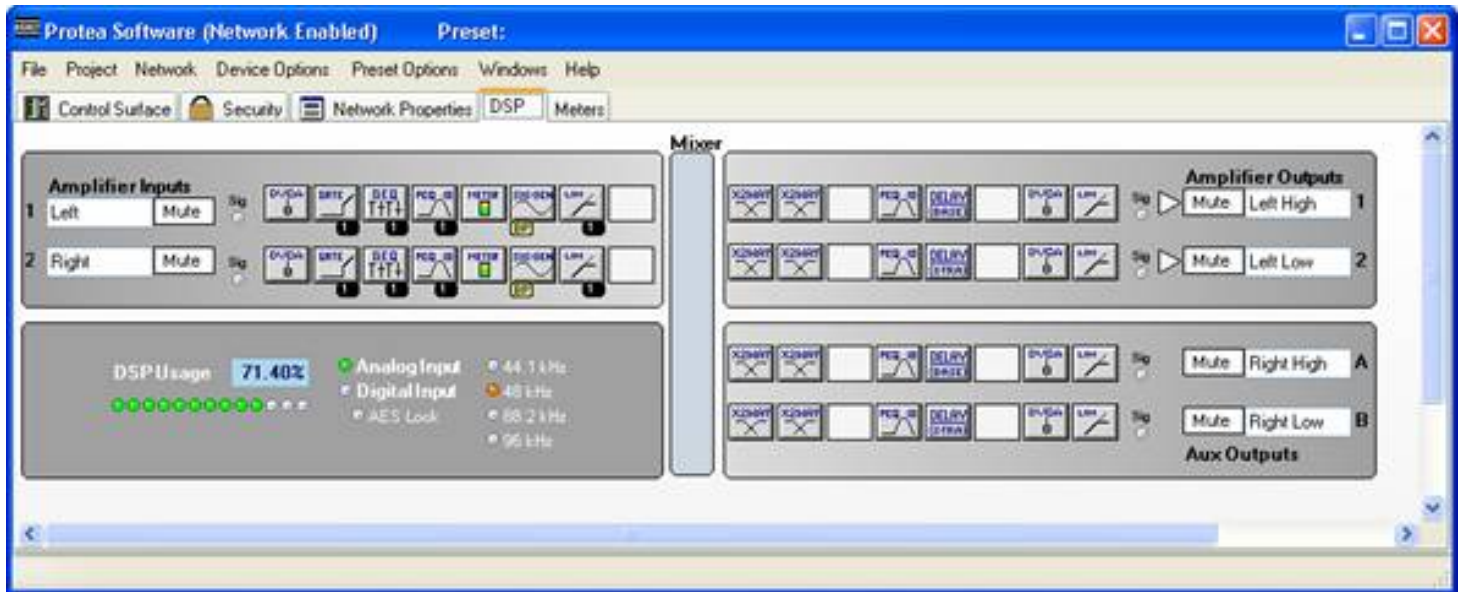
[DSP Control Screen](#)

[DSP Meter Screen](#)

PE Series DSP Control Screen

Overview:

The DSP Control Screen (pictured Below) is Protea^{ne} Software's window into the DSP. From this screen the DSP can be configured to do any type of DSP required. This includes setting the [Sampling Rate](#) (48khz, 96khz) selecting either the digital ([AES3](#)) or analog input as well as choosing which processing blocks are required where.



Channel Overview

Each DSP Channel in the PE Series DSP may be configured with up to 8 processing blocks. However only one of each block may be placed on a single channel. Each Channel may also be disabled through the Device Options menu. The input signal level led indicates input signal of a sufficient level is preset on a particular channel. In addition to the user configurable processing, there is a "Mixer" available in the center between the inputs and outputs. This allows each output to be given an combination of the two inputs. To access the mixer "Left Click" on the blue mixer section. To access any of the individual processing blocks simply left click on them.

Crossovers

Unlike most processing blocks the crossover blocks can not be placed anywhere. They must be placed directly following the mixer in available blocks. This means to place a 2 way crossover on outputs 1 & 2 the first two blocks after the mixer on both outputs 1 & 2 must be empty. To place a crossover somewhere else simply use the HPF and LPF blocks instead of the crossovers.

DSP Processing and Usage

Although you may configure your DSP As required there are limits. Namely in the amount of DSP "Horsepower" available. Below the input channels is a DSP Usage Gauge. This allows you to see how much of

the DSP Is consumed by the current signal path. Note that you must be below 44% to switch from 48Khz to 96Khz, this is because 96Khz consumes more than twice the usage than 48Khz. Also the sampling rates of 44.1 and 88.1 are do not provide any extra usage over 48Khz or 96Khz respectively.

Source Selection

The signal source may be configured to come from either digital or analog inputs. See [Source Selection](#)

AES3/EBU Digital Input

The DSP Supports Digital inputs using AES3/EBU signals. For more details see the [AES3](#) Section

Bypass & Link Group Indication

Below each processing block there are two indicators. “ByP” indicates the block is bypassed. A number indicates the block belongs to a specific link group.

Links to processing blocks

Dynamics

[Compressor/ Limiter](#)

[Gate](#)

Gain

[Standard Gain](#)

[VCA Gain](#)

EQ

[28 Band Graphic](#)

[31 Band Graphic](#)

[10 Band Parametric](#)

[6 Band Parametric](#)

[4 Band Parametric](#)

[2 Band Parametric](#)

Crossover

[4 way Crossover](#)

[3 way Crossover](#)

[2 way Crossover](#)

[High Pass Filter](#)

[Low Pass Filter](#)

Delay

[Base Delay](#)

[Extra Delay](#)

Tools

[Meter](#)

[Signal Generator](#)

PE Amplifier Source Selection

Overview:

The PE Amplifier DSP Option card supports several source combinations. These may be selected through the software. However since these require the DSP to change sampling rates a momentary drop out in audio is to be expected. The factory default is AES3 with Analog Backup at 48Khz.



Available Sources:

Analog Input at 48Khz

Analog Input at 96Khz

AES3 Input at 44.1Khz

AES3 Input at 48Khz

AES3 Input at 88.2Khz

AES3 Input at 96Khz

AES3 with Analog Backup at 48Khz

AES3 with Analog Backup at 96Khz

Details

Analog Inputs - will use the standard XLR/ Euroblock inputs available on the DSP Option card

AES3 Inputs - will use the AES3 XLR and receive a digital AES3 signal

AES3 with Analog Backup - will attempt to use the AES3 Signal if it is available. If there is an error (such as if the signal isn't present) the DSP will switch back to the Analog inputs.

48 KHz vs. 96 KHz

The DSP supports multiple sampling rates however it should be noted that increasing the sampling rate will effectively decrease the available DSP horsepower available. Therefore if a large amount of processing is required, a 48Khz sampling rate may also be required.

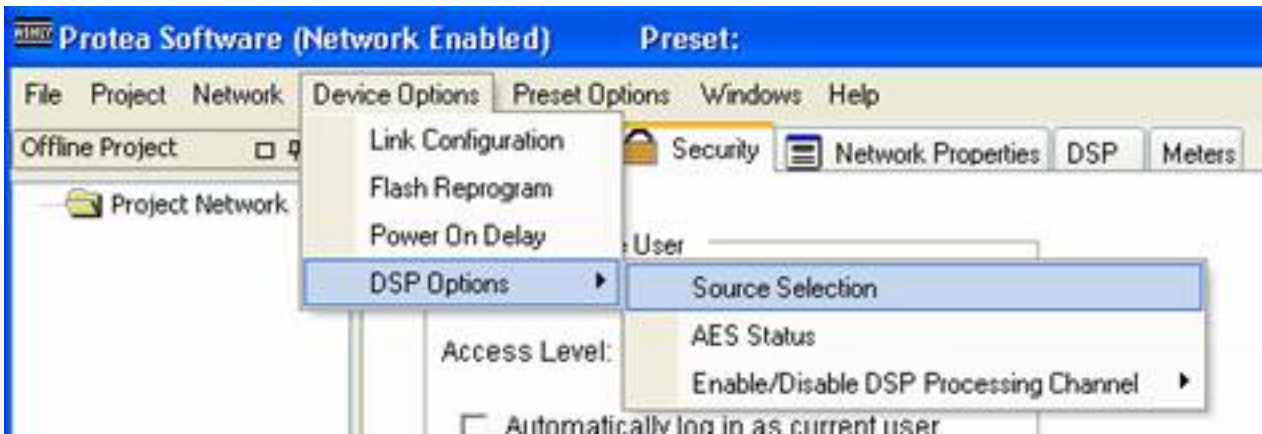
Note: AES3 signals must have the same sampling rate as the selected source or a frequency mismatch error will result.

AES3/ EBU

Overview

Some Protea^{NE} products now support digital inputs using the AES3 format. This allows 2 channels of digital audio to be transmitted over a single cable, thus maintaining a completely digital signal path until just before the amplifier. Currently 44.1, 48, 88.2 & 96kHz are supported as transfer rates of the AES signal. There is a bi-color led beside the AES connector which will indicate “Lock” or “Error” if the AES receiver is active. In software there is a AES Control Panel which shows the status of the AES receiver in greater detail.

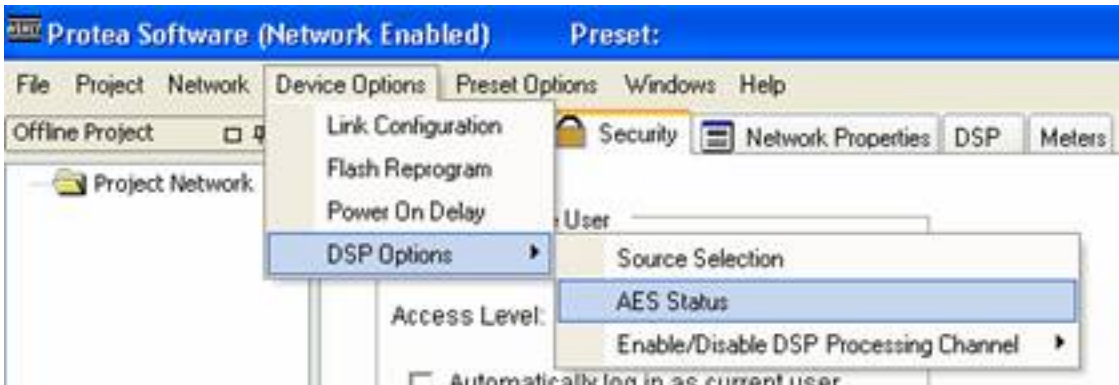
Source Selection



The DSP Must be configured to accept the AES source before it can be used. This is done through “Source Selection”. See the section on source selection for more details

Cable Specification - Standard 110ohm Microphone Cable (3 conductor) with Standard XLR connectors up to 100 Meters.

Software





Status Indicators

AES Input Status (green) – AES receiver is “Active” and has locked onto an incoming AES Signal and data is ready for DSP Processing.

(red) – AES receiver is “Active “ and has an error. See “Errors Since Reset” for specific Error.

Detected Sample Rate – Sample rate that the AES receiver is locked at.

Errors Since Reset – Displays the errors that have occurred since either the Unit has been reset or “Reset Errors” has been used.

AES Parity Error – parity error with incoming signal

AES Non-Audio Error – incoming signal is not Audio

AES Sample Rate – Unable to lock onto incoming frequency

AES Invalid Error – incoming signal is invalid

AES Lock Loss Error – unable to lock onto incoming signal, or lost lock of incoming signal

AES Sample Rate Mismatch – AES receiver frequency does not match DSP frequency.

AES Non-PCM Error – invalid AES format

Note: Use “Reset Errors” to reset the errors shown and see any current errors. This is required since error status will latch in software to allow a user to see intermittent errors.

Hardware Indicators

AES Status (green) – AES receiver is “Active” and has locked onto an incoming AES Signal.

(red) – AES receiver is “Active” and has an error. AES Signal will be muted.

Compressor/ Limiter Block

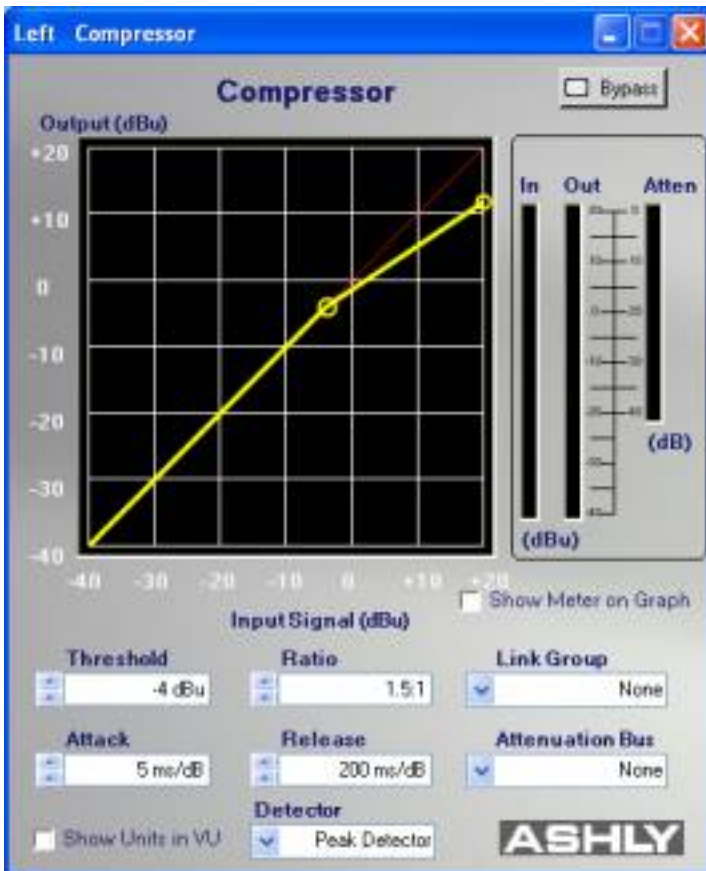


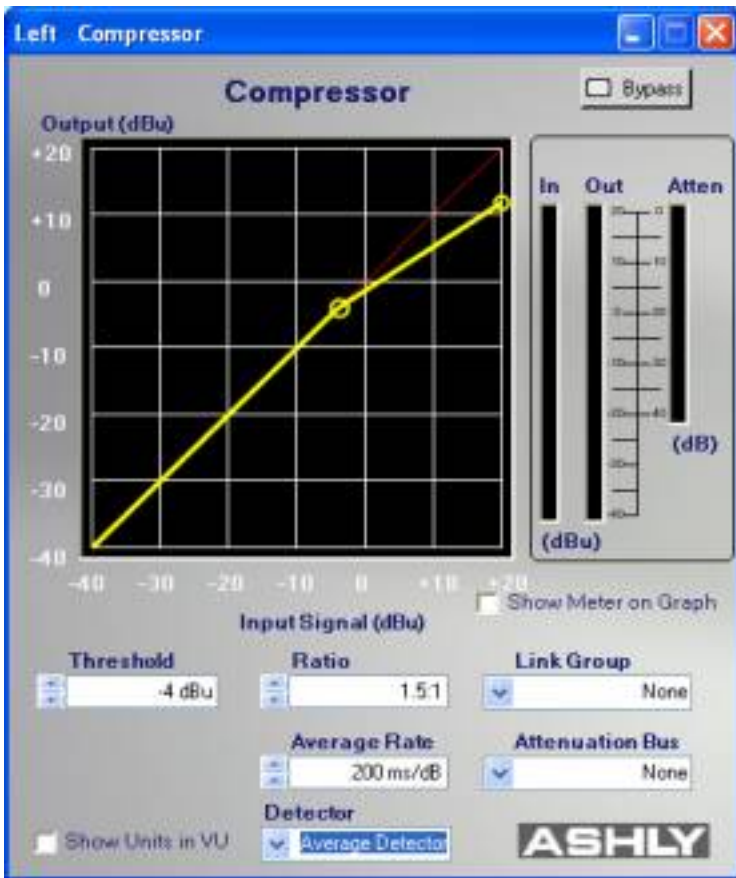
Description:

The “Comp/ Limiter DSP Block” is a full featured Compressor/ Limiter. It contains a Threshold, Ratio, Attack Time and Release Time to modify the response of the compressor limiter. The compressor limiter also contains a [Link Group](#) to link the parameters of this block with limiters on other channels as well as an “Attenuation Bus” to link the amount of attenuation applied to all channels on an attenuation bus within a single DSP. The Comp/Limiter also has a selectable Detector. For simple peak limiting see the [Brick Wall Limiter](#).

Peak Detector

Average Detector





Parameters:

Threshold – signal level at which the limiter begins to attenuate the signal. (+20 to -20 dBu)

Ratio – degree of attenuation to the signal above the limiter threshold point.

Attack Time - rate at which the limiter attenuates the input signal in ms / dB.

Release Time – rate at which the limiter attenuation will return to normal in ms/dB.

Link Group – Link group to which all parameters are tied.

Attenuation Bus – bus to which attenuation is tied.

Show Input Trace – will display the input trace inside the Comp Limiter Graph.

Detector – will selected between Peak and Average

Average Rate – Attack/Release Time for Average Detector

Details:

Attenuation Bus – all limiters within a given DSP assigned to the same Attenuation Bus will apply the largest amount of attenuation on any of the assigned Limiters.

Detector – The Comp Limiter typically functions with a peak detector. This means that the peak level of a signal is measured and is adjusted based on this level. However the detector

may be set to average which will use the average level of a signal to determine the amount of attenuation required. The Average Rate controls the speed at which the Average Detector functions (similar to the Attack and Release Times of the Peak Detector.)

Linking in Protea^{NE}

Overview

Linking has been incorporated into all Protea^{NE} Products. This allows specific function blocks to be linked with other function blocks. (ie. you can link two Graphic Equalizers or Amp Channels). This is handled in two ways. First the blocks may be linked only inside of a Protea device. Second they may also be linked across devices (with Protea^{NE} Software). This is all managed via "Link Groups" any function can be assigned to just about any group however only like functions in a single group will be linked with each other.

Linking Groups:

Each Protea device will maintain several Link Groups. DSP Functions (or offsets) may be assigned to these groups through Protea^{NE} Software. Once Assigned to a group all LIKE functions within the group will track parameter changes. However more than one function type may be assigned to a group. Each Link group may be given a Name and also be assigned a color for easy identification.

Linking Inside a Device

Linking within a single Protea device is handled by the protea device. Whenever a change is made to a function that is assigned to a "Link Group" all other link functions in that link group are also modified. This can easily allow for stereo channels or 5.1 setups. This also allows 3rd Party Controllers to send a single message to update multiple Channels.

Linking Across Devices

In Protea^{NE} Software it is possible to link function blocks across devices. For example you want the offset gain for both channels on 4 amplifiers to all be linked. Simply assign them all to the same link group and ensure that the link group is configured to "Link Across devices" for each device. All link Functions will be linked and their parameters will track for each Link Group configured to "Link Across devices".

Linking Configuration Screen:

The link Configuration Screen Allows the name, color of a link group to be modified as well as allowing the group to be setup to either link across devices or only within a single device

Link Group Configuration



Link Group Configuration

Group	Title	Color	Link Across Devices?
1	<input type="text" value="Group 1"/>	<input type="text" value="black"/>	<input checked="" type="checkbox"/>
2	<input type="text" value="Group 2"/>	<input type="text" value="red"/>	<input checked="" type="checkbox"/>
3	<input type="text" value="Group 3"/>	<input type="text" value="blue"/>	<input checked="" type="checkbox"/>
4	<input type="text" value="Group 4"/>	<input type="text" value="green"/>	<input checked="" type="checkbox"/>
5	<input type="text" value="Group 5"/>	<input type="text" value="yellow"/>	<input checked="" type="checkbox"/>
6	<input type="text" value="Group 6"/>	<input type="text" value="orange"/>	<input checked="" type="checkbox"/>
7	<input type="text" value="Group 7"/>	<input type="text" value="purple"/>	<input checked="" type="checkbox"/>
8	<input type="text" value="Group 8"/>	<input type="text" value="Grey"/>	<input checked="" type="checkbox"/>

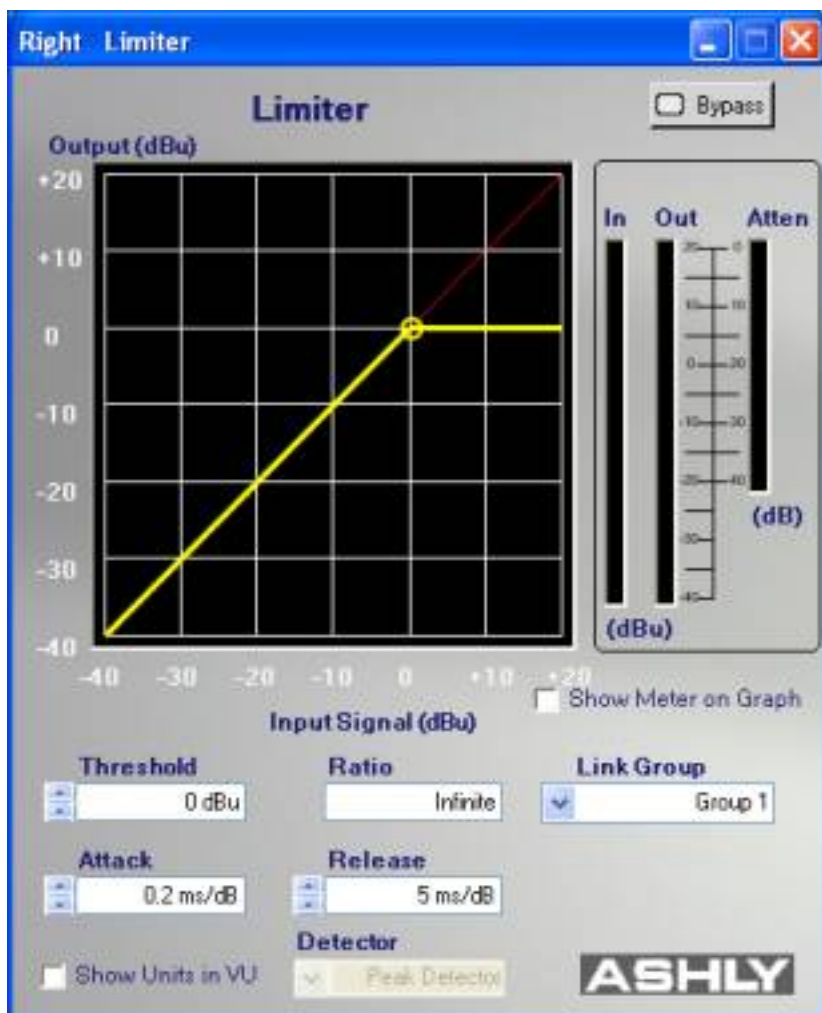
Brick Wall Limiter Block



Description:

The “Brick Wall Limiter DSP Block” is a stripped down Comp/Limiter, which has a fixed Ratio (Infinite). This block allows for hard limiting to protect speakers or avoid Clipping. It is worth noting that with a Brick wall limiter Clipping internal to the DSP may all but be eliminated as a floating point DSP is not easily clipped internally and there for the limiter can force any signal above +20dBu down below +20dBu.

The Brick Wall Limiter contains a Threshold, Attack Time and Release Time to modify the response of the compressor limiter. The compressor limiter also contains a [Link Group](#) to link the parameters of this block with limiters on other channels. For a more configurable compressor/Limiter see the [Comp/Limiter Block](#).



Parameters:

Threshold – signal level at which the limiter begins to attenuate the signal. (+20 to -20 dBu)

Ratio – Fixed at Infinite.

Attack Time - rate at which the limiter attenuates the input signal in mS / dB.

Release Time – rate at which the limiter attenuation will return to normal in ms/dB.

Link Group – Link group to which all parameters are tied.

Detector – Fixed at Peak Detector

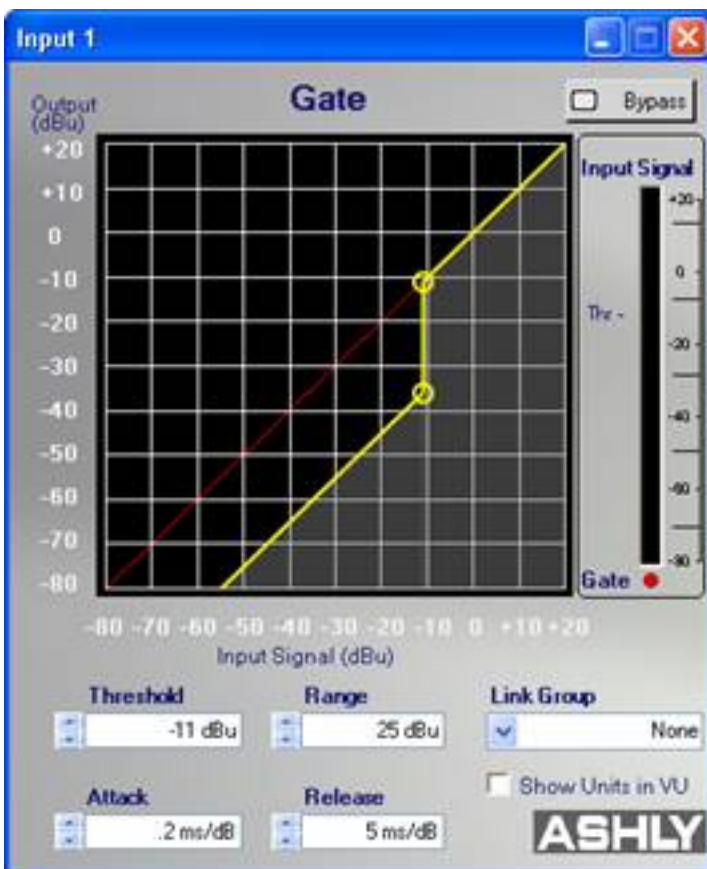
Show Input Trace – will display the input trace inside the Comp Limiter Graph.

Noise Gate Block



Description:

Noise gates are used to minimize unwanted or ambient low level signal from an individual microphone input. THRESHOLD is the level above which an input signal will pass through, and below which its signal is attenuated by the RANGE value. ATTACK and RELEASE control the time characteristics of the gating action. Attack will set the amount of time used to return from the floor to unity. While Release will set the time required to attenuate the signal. The Gate may be assigned to [a link group](#) to link its parameters with those of Gates on other channels.



Parameter

- Threshold – The level at which the gate allows signal to pass unaffected.
- Range – The amount of attenuation applied to the signal.

Attack – The amount of time used to return to unity Gain after the signal has reached the Threshold

Release – The amount of time to attenuate the signal

Link Group – The link group to which this block belongs

Standard DSP Gain Block

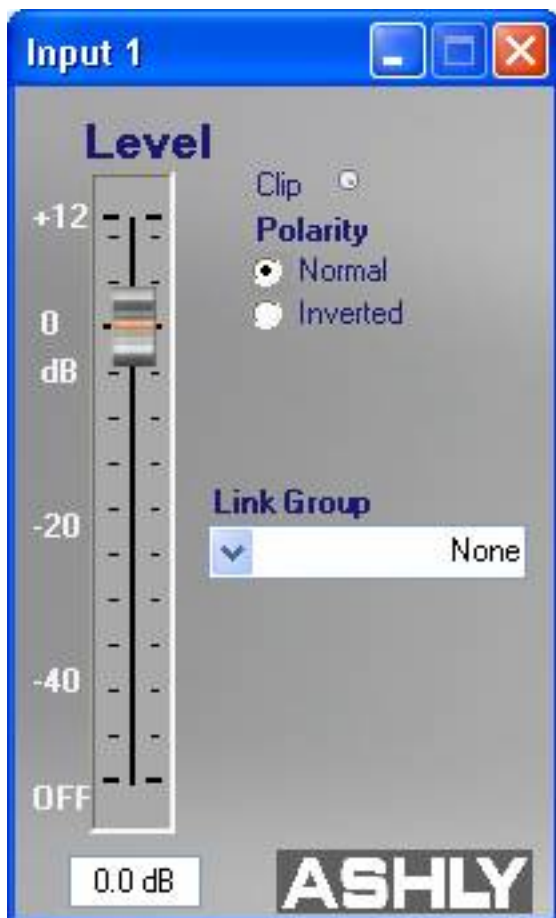


Description:

The Standard DSP Gain block allows the user to modify a signals level by either applying gain or attenuation to the signal. Also the polarity of the signal may be inverted. The gain control allows for level adjustment between +12db and -50dB as well as “OFF” which effectively mutes the signal. Adjustments are allowed in this range in .1dB increments while the fader moves in .5dB increments finer adjustment may be made by typing in the value into the provided text box.

The gain block may be linked with any other “Standard DSP Gain Block” though PnsS. This will force all parameters (level & polarity) to be the same. When one parameter is changed, the same change is made to all linked Gain Blocks. Note that changes to a Linked Standard DSP Gain block do not affect a DSP [VCA Gain Block](#). Also the values are set identically and can not be offset from each other.

For a more powerful linking of gains consider using the [“DSP VCA Gain Block”](#).



Parameters:

Level - Off, -50dB to +12dB in .1dB increments

Polarity - Normal or Inverted

Link Group - None, or any available Link Group in the DSP < [LINK TO LINK GROUP INFO](#)
HERE >.

VCA DSP Gain Block



Description:

The VCA DSP Gain block allows the user to modify a signals level by either applying gain or attenuation to the signal. Also the polarity of the signal may be inverted. The gain control allows for level adjustment between +12dB and -50dB as well as “OFF” which effectively mutes the signal. Adjustments are allowed in this range in .1dB increments while the fader moves in .5dB increments finer adjustment may be made by typing in the value into the provided text box.

In addition to using the Level adjustment features of the VCA DSP Gain block. The Gain can be assigned to a “Digital VCA.” This allows a “Master” fader to adjust multiple gains each with its own offset (set on the normal gain fader). In addition to assigning to a single VCA the Gain be assigned to multiple VCA’s. This allows for more complex configurations.



Parameters:

Level - Off, -50dB to +12dB in .1dB increments

Polarity - Normal or Inverted

VCA Enabled – VCAs which adjust (offset) the level applied in this gain stage. See “Total” for total amount of Gain.

Total – Total Attenuation/ Gain Applied after Fader + all Engaged VCAs are summed.

Digital VCA Group



VCA Enabled (for this gain stage)

VCA's Level (read only)

VCA's Title

Details:

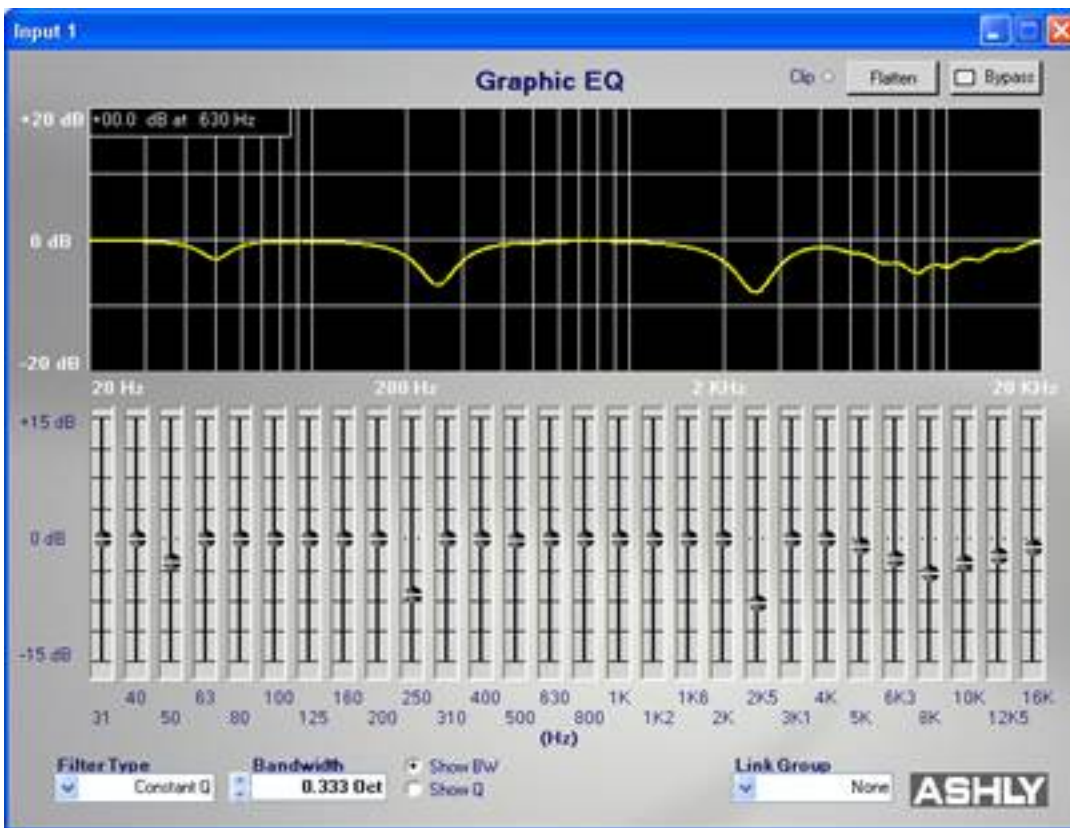
Digital VCA – A Digital VCA is very similar to an analog VCA except no voltage is actually involved. The idea however is the same. From a master fade the level of a channel or group of channels may be adjusted however unlike linking in the standard DSP Gain block the Gains do not have to be equal.

Graphic Equalizer Block (GEQ & GEQ31)



Description:

The Graphic Equalizer (GEQ) Block allows for a 28 Band Graphic EQ and a 31 Band Graphic EQ to be inserted into the signal path. Although a typical graphic equalizer has a fixed Q (Bandwidth) this graphic equalizer allows the Bandwidth to be adjusted between $\frac{1}{4}$ octave to $\frac{1}{2}$ octave. Allowing for more versatile applications. Also the GEQ may be used as a Constant Q or Proportional Q Equalizer. A flatten button is available to return all Bands to 0. The GEQ may be assigned to a [Link Group](#) so that GEQ's on multiple channels will track each other.



Parameter

Filter Type – Proportional Q or Constant Q

Bandwidth – Bandwidth of the filter (before Proportional Adjustment).

Link Group – Link Group to which this block is assigned.
Faders – Graphic representation of 28 filters.

Details:

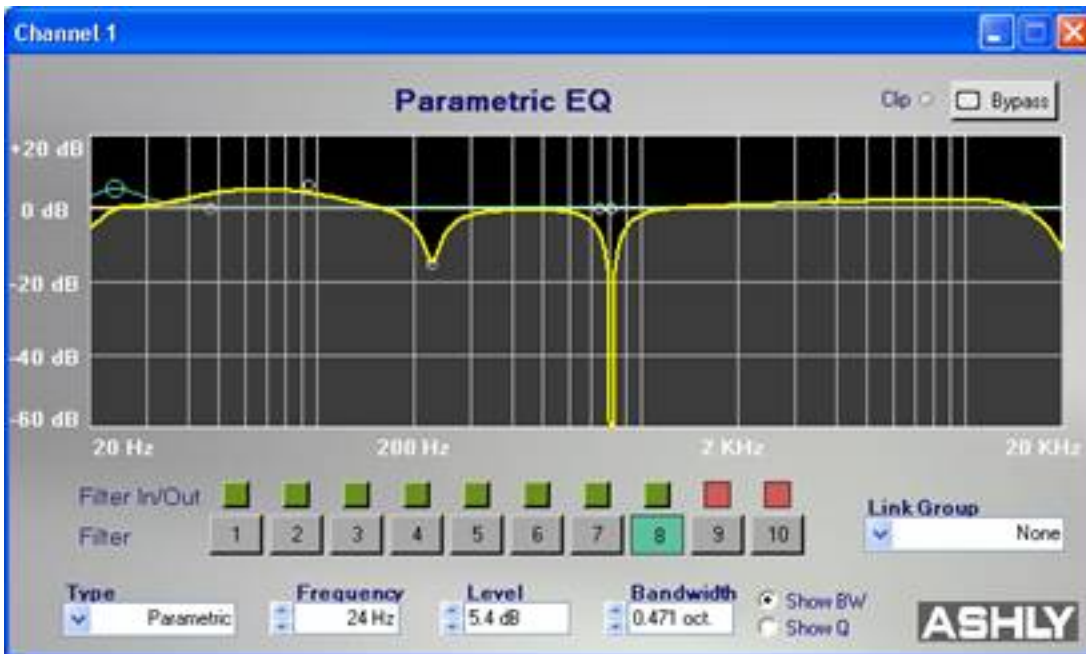
Proportional vs Constant Q – Constant Q uses the Q or Bandwidth set in for the GEQ for all levels of Gain / Attenuation. The Proportional Q will adjust the Q based on the amount of Gain Attenuation. This can help reduce interactions into other bands and limit the signal impact for large Adjustments or broaden the impact for minor adjustments. The default is constant Q at 0.333 octaves.

Parametric EQ (10,6,4 or 2) Block



Description:

The “Parametric EQ” Block allows the user to use several types of filters to tailor the signal response to just about any situation. The Parametric EQ comes in several sizes, with 2,4,6 or 10 filters available. All of these sizes may be used in conjunction to achieve 22 filters total on any channel + any other desired processing. The Parametric EQ has Parametric filters, High Shelf, Low Shelf, High Pass, Low Pass, All Pass, and Notch Filters available. The PEQ may be [linked](#) with any other PEQ of the same Size!



Parameters:

- Filter – Current Active Filter (Highlighted)
- Filter In/Out – Green = Filter In (engaged), Red = Filter Out (disengaged)
- Type – Current Active Filter Type.
- Frequency – Current Active Filter Center Frequency
- Level – Current Active Filter Level
- Bandwidth – Current Active Filter Bandwidth (or Q)
- Link Group – Link Group that this PEQ is linked with.

Details:

Parametric Filter – filter where the center frequency, level and bandwidth

may be adjusted

High Shelf – shelving filter where the corner frequency and level may be adjusted, this is available with slopes of 6 or 12dB/oct.

Low Shelf - shelving filter where the corner frequency and level may be adjusted, this is available with slopes of 6 or 12dB/oct.

High Pass – 24dB/oct Linkwitz-Reighly High Pass Filter with adjustable corner frequency

Low Pass – 24dB/oct Linkwitz-Reighly Low Pass Filter with adjustable corner frequency

All Pass – Filter with 0dB level adjustment however with 180degree Phase adjustment

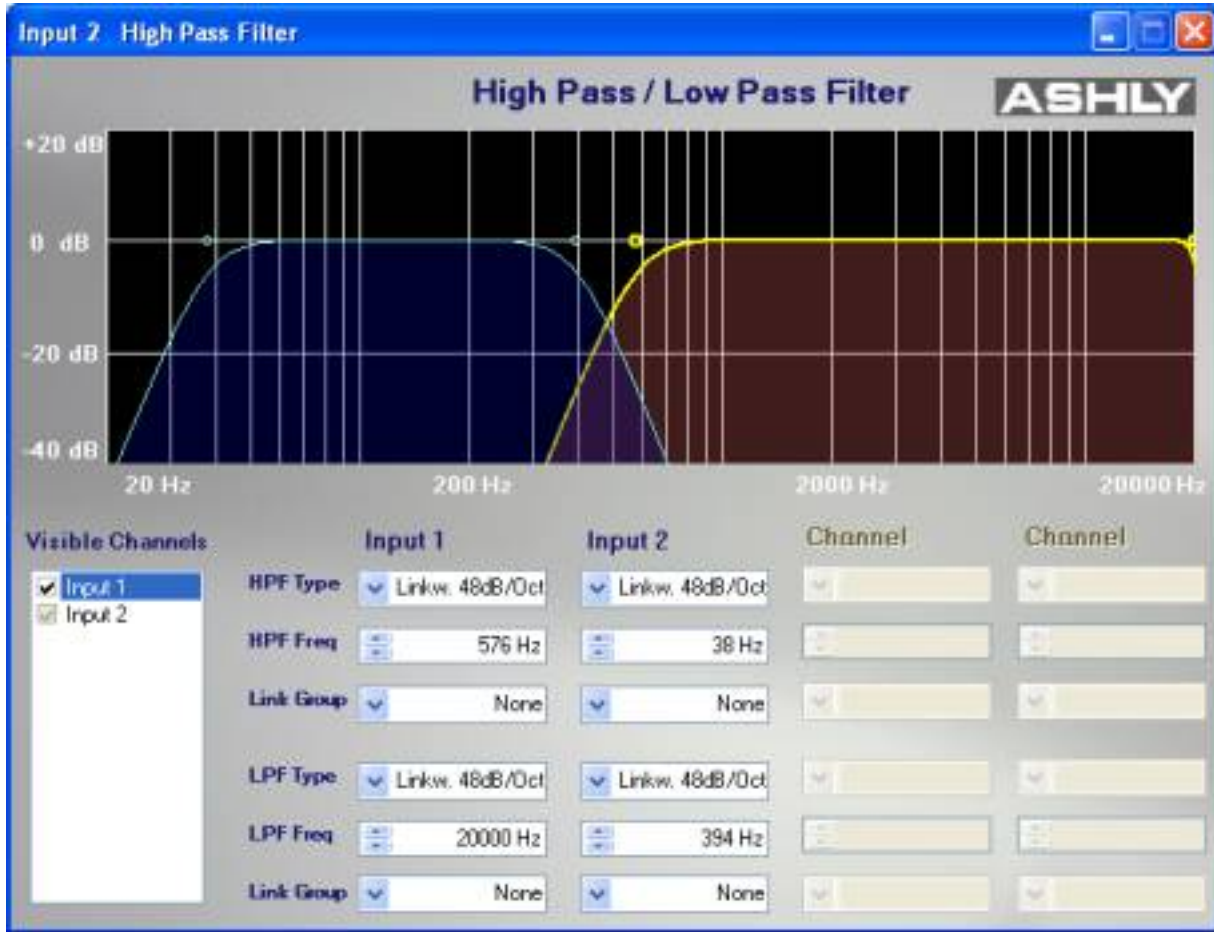
Notch – Infinite Notch at specific center frequency with varying width.

Variable Q HPF – Second Order High Pass filter with a Variable Q. The Q adjustment is similar to the Response Adjustment in the Ashly Analog Crossovers.

Variable Q LPF – Second Order Low Pass filter with a Variable Q. The Q adjustment is similar to the Response Adjustment in the Ashly Analog Crossovers.

Band Pass – Filter to band limiter a signal with a specific Q and Center Frequency. At it's Center Frequency it has Unity Gain. This is useful for Triggers/ Special Effects or if you wish to have your own manual Wah-Wah effect.

High Pass Filter / Low Pass Filter (HPF/LPF) Block



High Pass Filter determines the lower frequency limit of the signal, while the Low Pass Filter determines the upper frequency limit. Be careful not to accidentally send low frequency signals to high frequency drivers. Check the loudspeaker specifications to determine a safe operating frequency range.

Description:

The HPF or LPF block(s) allow for users to create and edit a high pass and/or low pass filter(s) on a specific channel. By left clicking on a HPF/LPF block, users can open the GUI for manipulating block settings. Users can

choose to edit filter parameters such as *dB/Octave*, *Filter Type*, and associated [Linkgroup](#) for each HPF or LPF that is present in the system, with up to four different channels of information being displayed simultaneously.

dB/Octave choices:

- 12 dB/Oct.
- 18 dB/Oct.
- 24 dB/Oct.
- 48 dB/Oct.

Available Filter Types:

- Butterworth
- Bessel
- Linkwitz-Riley

Note: The Crossover blocks are special case versions of multiple HPF/LPF blocks. When clicking on a crossover block to enter the GUI for parameter manipulation, only those HPF/LPF's that correspond with the particular crossover will be available for viewing and editing.

Crossover Details:

In addition to the frequency and slope, crossover filters can be selected as having Butterworth, Bessel, or Linkwitz-

Riley response. These refer to the shape of a filter's slope at the cut-off frequency, affecting the way two adjacent pass

bands interact at the crossover point. 24dB/octave Linkwitz-Riley filters produce a flat transition through the crossover

region, assuming both overlapping filters are set to the same frequency, slope, and response type. ***24dB/oct Linkwitz-***

Riley filters are the industry standard, the easiest to use, and the filter type recommended by Ashly.

Other filter types

are available, but may require polarity switching or other adjustments for proper results. The following paragraphs offer

summary of the three filter types as used in the 24.24M crossovers.

Butterworth:

Butterworth filters individually are always -3dB at the displayed crossover frequency and are used because

they have a "maximally flat" passband and sharpest transition to the stopband. When a Butterworth HPF and LPF of the same crossover frequency are summed, the combined response is always +3dB. With 12dB per octave Butterworth crossover filters, one of the outputs must be inverted or else the combined response will result in a large notch at the crossover frequency.

Bessel:

These filters, as implemented in Protea NE, are always -3dB at the displayed crossover frequency. Bessel filters are used because they have a maximally flat group delay. Stated another way, Bessel filters have the most linear phase response. When a Bessel HPF and LPF of the same crossover frequency are summed, the combined response is +3dB for 12dB/oct, 18dB/oct, and 48dB/oct Bessel filters, and -2dB for 24dB/oct Bessel filters. One of the outputs must be inverted when using either 12dB/oct or 18dB/oct Bessel crossover filters or else the combined response will have a large notch.

Linkwitz-Riley

The 12 dB/oct, 24dB/oct, and 48dB/oct Linkwitz-Riley filters individually are always -6dB at the displayed crossover frequency, however the 18dB/oct Linkwitz filters individually are always -3dB at the displayed crossover frequency.

The reason for this is that Linkwitz-Riley filters are defined in terms of performance criterion on the summing of two adjacent crossover HPF and LPF filters, rather than defined in terms of the pole-zero characteristics of individual filters.

The 18dB/oct Linkwitz-Riley individually are 18dB/oct Butterworth filters in that they have Butterworth pole-zero characteristics and also satisfy the criterion for Linkwitz-Riley filters.

When a Linkwitz-Riley HPF and LPF of the same crossover frequency are summed, the combined response is always flat. With 12dB/oct Linkwitz-Riley crossover filters, one of the outputs must be inverted or else the combined response will have a large notch at the crossover frequency.

Delay (Base/ Extra) Block



Description:

Both the Base Delay and Extra Delay allow the signal to be delayed by a given number of samples (and thus a given amount of time). The only differences between the Base Delay and extra delay is the total amount of delay available and also the amount of DSP Usage used by each block. The base delay uses far less DSP than the Extra delay and should be used if possible. Since the delay, delays a given number of samples, the delay and total delay available will change when the Sampling frequency is changed. Also for convenience if both the standard and extra delay are required they can be presented on a single control screen. For [linking](#) purposes Base Delays are linked only to Base Delays, while Extra Delays are linked only to Extra delays.



Parameters:

Delay – delay added to the signal (presented in ms, ft or meters), note distances are Temp. dependant

Temp – local temperature, used to calculate the speed of sound through air and thus the distance.

Link Group – the link group to which the delay block will be linked

Show Detailed Channel Information – if both base and extra delay are present, when checked will show both controls in a single window.

Audio Meter Block



Description:

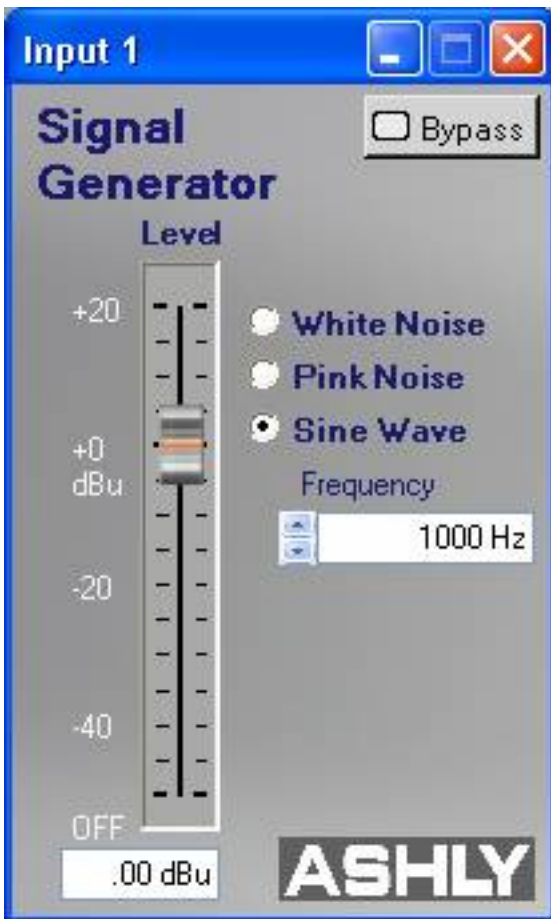
The audio meter does Not affect the audio signal however simply provides a user placed meter to view the current audio level at a critical stage.

Signal Generator Block



Description:

The “Signal Generator DSP Block” is a unique block in it will provide its own output regardless of its input when engaged. When bypassed however it will allow the input signal to pass to the output of the block unaffected. There are several types of signals that can be generated. Pink Noise, White Noise and Sine Wave (20Hz to 20Khz). These signals may be generated at any level from -50dBu to +20dBu.



Parameters:

Mode – (Pink Noise, White Noise, Sine Wave) signal to be generated

Frequency – Frequency of Sine Wave to be generated

Level – Level of signal Generated

Bypass – Bypass Signal Generator and pass input signal to output unaffected.

Details:

Pink Noise – contains every frequency in the human hearing range (20Hz-12kHz) however it is filtered such that composite sound wave has equal energy at every octave (-6 dB per octave). This accounts for the increasing number of frequencies in the higher octaves.

White Noise – similar to pink noise only without the filtering.

White noise contains equal amounts of every frequency in the human hearing range (20Hz- 12kHz).

Sine Wave – Produces a signal frequency at the given level as wave.