## 7.2 Connecting the ne24.24M to a computer using Ethernet

Ethernet is the preferred data protocol for computer control. A standard RJ-45 jack is used for connecting directly to a PC, PDA, or 10/100 Base-T network.

## 7.3 Connecting the ne24.24M to a computer using the RS-232 Dataport

The ne24.24M is optimally designed for use with ethernet control, however it remains backward compatible to systems using RS-232 control. In other words, older Protes System Software (or other) based installs using RS-232 will work fine with a new ne24.24M. If, on the other hand, the ne24.24M needs RS-232 control while still in a networked ethernet system using Protea NE Software, add a 24.24M (not the ne24.24M) from the <Add Item> menu in Protea NE Software and set the unit up accordingly using the self-launching older Protea System Software. RS-232 and ethernet will run concurrent on the same unit.

For RS-232, a D-Sub nine pin female to male connector cable is used to connect to the ne24.24M. An active USB to RS-232 converter can be purchased at computer stores if necessary. Here are the suggested steps for connecting the ne24.24M using RS-232:

1) Plug the RS-232 cable into an available serial port on the PC or control unit.

2) Plug the other end of the cable into either the front or back RS-232 Dataport on the ne24.24M.

3) Turn on the power to the ne24.24M.

4) Open Protea System Software (version 6 .4 or higher), *not* the Ethernet based Protea NE Software.

5) Go to the <Devices> menu and select <ne24.24M>.

6) Select the appropriate COMM port (Comm 1-16).

7) Select desired Baud rate (9600bps or 38,400bps). Note that 38,400bps is only supported by the ne24.24M.

8) Go to the <Communications> menu and select <Enable Communications>.

9) The connection should now be established. The unit will communicate on any device channel (Channel 1-16) in the Protea System Software (see sec. 4.2.)

# 8. AUDIO FUNCTIONS

Editing of audio controls is done primarily in Protea System Software. Input and output expansion cards are autodetected, and the software automatically updates to display the current ne24.24M status. *Note: Accidental, potentially destructive loudspeaker damage can occur when abrupt changes are made to EQ, filter, or level controls, so plan carefully before making radical changes to a live sound system.*

## 8.1 Input Audio Functions


The following functions are available on all inputs; Mute, Preamp Gain, Input Gain, Delay, EQ, Noise Gate, Autoleveler, Ducker, and Matrix Routing.

**8.1a Input Mute**

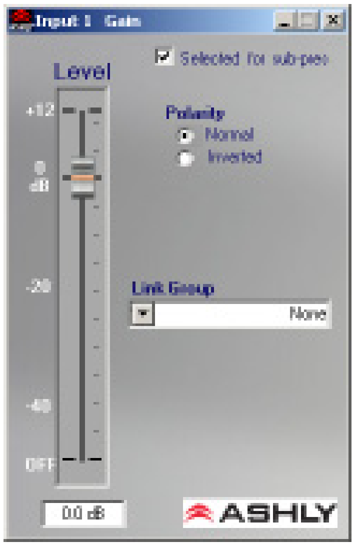
This turns off the input channel without changing gain settings. When an input channel is muted, that channel's red mute LED on the face panel is lit.

**8.1b Input Preamp**

This determines the up-front analog gain to an input signal. A good rule of thumb is to allow 20dB of headroom above the nominal input signal level. Clipping occurs at +20dBu, so a microphone nominally generating a -40dBu signal should have +40dB of gain, a 0dBu line level input would be set to 0dB gain, etc.

### 8.1c Phantom Power

+48V Phantom power for condenser microphones can be provided to individual channels by checking the <+48V Phantom Power> box in the preamp block for a given channel.



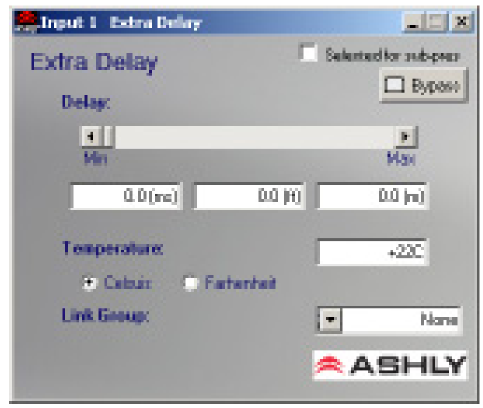

### 8.1c Push To Talk Mic

This check box and drop down menu selects the back panel logic input to serve as a contact closure input channel engage.

### 8.1d Input Gain

Different from the Preamp block, input Gain adjusts signal level from +12dB to Off. Due to limitations within the graphical interface, fine tuning gain settings in 0.1dB steps is accomplished using the <up/down arrows> on the keyboard. Changes of 3dB are quickly accomplished using <PageUp/PageDown> buttons. To instantly return a gain setting to 0dB, press <Ctrl + click>.

### 8.1e Input Delay




In large installations or outdoor venues there are often many speakers in various locations to get the best coverage possible. Since sound travels relatively slow through air (1130 ft/s at 20 deg. C), multiple loudspeaker locations can create a situation where the original audio signal, simultaneously leaving all loudspeakers, arrives at a single point in the venue at several different times. Needless to say this causes problems, and what may be crystal clear sound directly in front of any one loudspeaker can be unintelligible at the farther reaches of the venue with direct line-of-sound to multiple loudspeaker sources.

The solution is to delay the audio signal to the loudspeakers located further away from the primary source, so that sound comes out of the distant loudspeakers at the exact time that sound from the main loudspeakers arrives. Within the Protea ne24.24M, up to 682 milliseconds of time delay are available on each input channel, allowing secondary loudspeakers to be time aligned with the primary speakers up to 771 feet (235m) away. Set the TEMP text box to the current air temperature to get the most accurate display of delay distance.

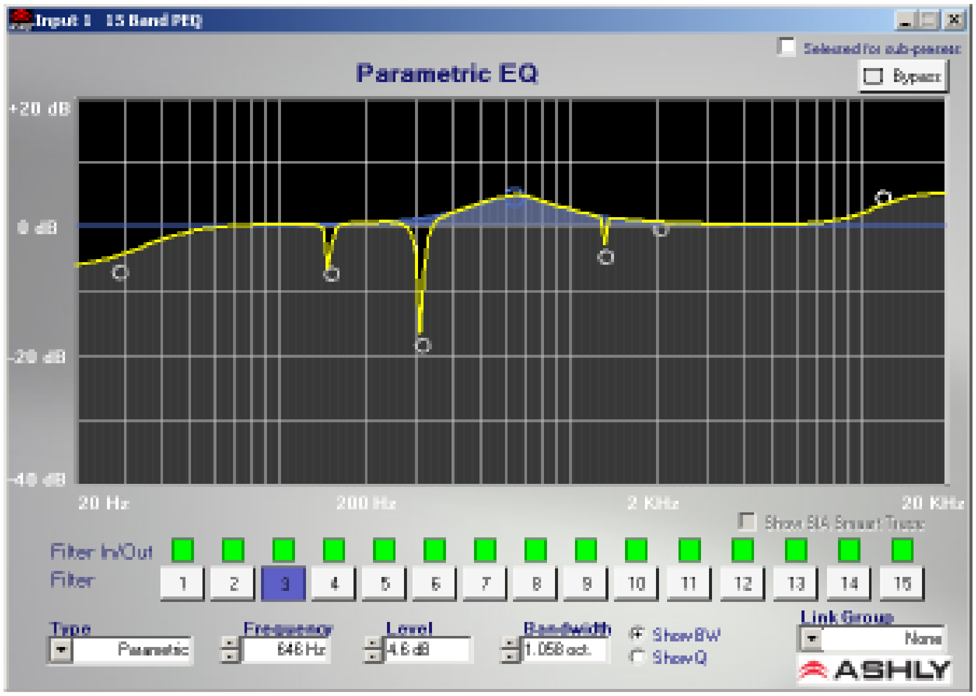
Fine tuning delay settings is accomplished using the <up/down arrows> on the keyboard. Course changes of 1mS are quickly accomplished using <PageUp/PageDown> buttons. For an explanation of short time delay uses, see section 9.2c.

### 8.1f Input EQ

The Protea ne24.24M input EQ section offers 15 custom filters per input. Types of filters available for each band include parametric, 1st order and 2nd order low shelf, 1st order and 2nd order high shelf, and all-pass.

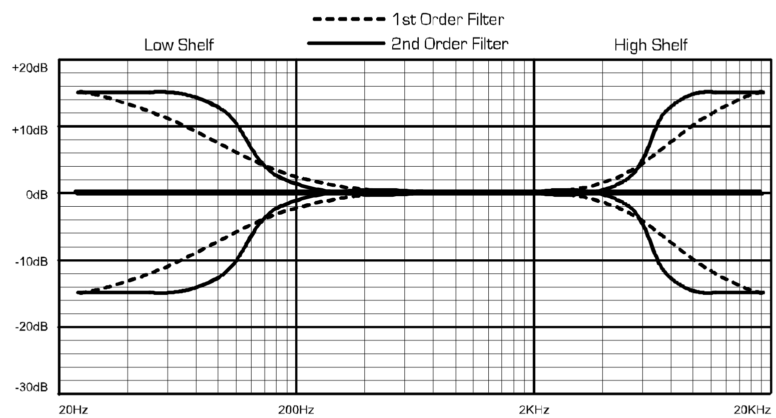
**Shelving EQ filters:** 1st order filters use a gentle 6dB per octave slope, while 2nd order filters use a 12dB per octave slope for a more pronounced boost or cut. All shelving filters have a boost/cut range of +/-15dB. Low shelving filters have a frequency range from 20Hz through 2kHz, and the high shelving filters range from 3.886kHz through 20kHz. Shelving filters are most useful as broad tone controls to boost or cut the high end or low end of an audio signal's frequency content. Because they affect a wider spectrum of audio, they are not as suitable for feedback control as parametric filters. Course and fine tuning of shelving filters can be performed using <page up/page down> and the <up/ down arrows> on the keyboard.

**Parametric EQ filters:** These are peak filters with the ability to control boost or cut, frequency center, and bandwidth, also called "Q" for this type of filter. Think of one band of parametric EQ as a single graphic equalizer fader except that the frequency is variable, not fixed, and that the bandwidth, or how "wide" the filter affects the frequency spectrum at the center frequency, is completely variable. The smaller the bandwidth (higher Q), the less the audio signal on either side of the frequency center is boost or cut, whereas a larger "wider" bandwidth (lower Q) produces an audible change to the overall tone of a signal. Parametric filters are best used to hunt down and eliminate problem feedback frequencies, add or remove a characteristic "hot spot" from microphones, or clean up room resonance situations. It is well worth the time becoming proficient with parametric EQ filters, as they offer the best solution to many EQ problems.

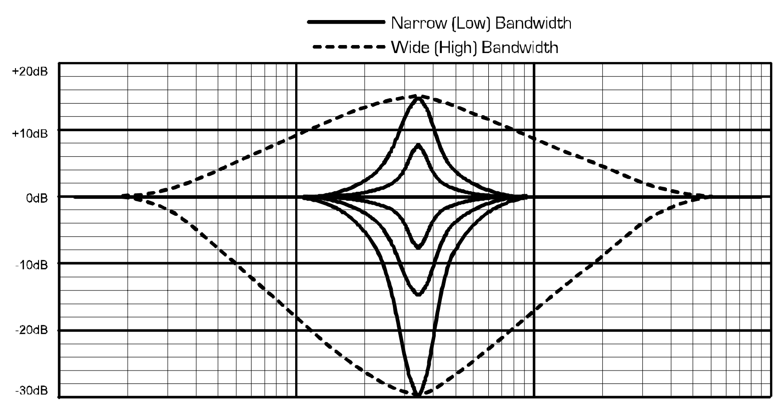



Protea ne24.24M parametric filters have a boost/cut range of +15dB to -30dB. There is more cut than boost because one of the more common uses for parametric filters is to dramatically cut, or "notch out", very narrow frequencies (low bandwidth) in order to eliminate system feedback problems.

Every parametric EQ filter has a center frequency. The factory default is 1kHz, but each filter is adjustable from 20Hz to 20kHz in 1Hz steps. Carefully sweeping a narrow bandwidth filter through a problem feedback area, with just a slight boost, is a quick way to find the exact frequency causing feedback trouble. Once the offensive frequency has been found, cut the filter's level, and then adjust the bandwidth as narrow as possible while still eliminating the feedback problem. Bandwidth is adjustable from about 1/64 octave to four octaves, and the lower the bandwidth (higher Q), the less audible the filter action will be. Finding the problem frequency is relatively easy, but finding the best combination of cut and bandwidth takes a little practice. Again, it is well worth the time getting comfortable with the notching procedure, so that problems can be quickly addressed with a sufficient but minimal amount of corrective EQ.

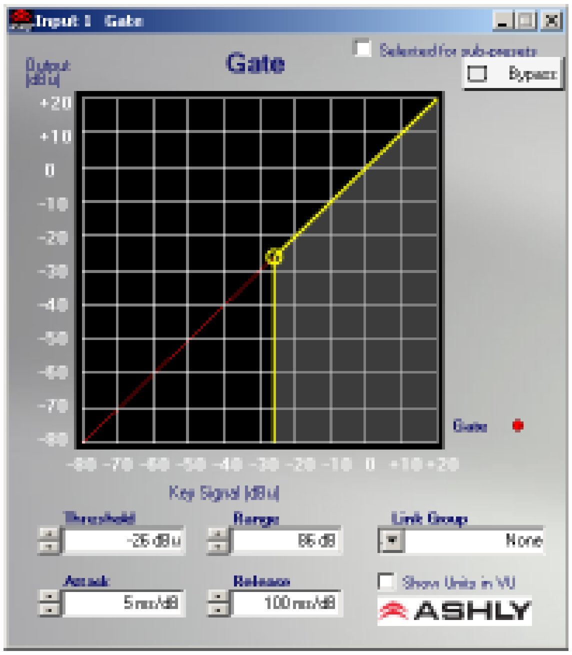



24.24M Shelving Filters




24.24M Parametric Filters

**All Pass filters:** All pass filters have no effect on frequency amplitude, but rather are used to adjust the phase response of the signal at a given frequency, and are often used in conjunction with a frequency-domain filter to correct phase changes. At low frequencies, there




is 0 degree phase shift, at the All Pass filter center frequency there is -180 degrees of phase shift, and at high frequencies there is -360 degrees of phase shift.

Each input channel has an EQ On/Off button for all filters, and in turn each filter band has its own bypass button. The Flatten Curve function returns all filters to 0dB, but preserves the frequency and bandwidth of any used filters.

### 8.1g Noise Gate

Noise gates are used to minimize unwanted or ambient low level signal from an individual mic input. THRESHOLD is the level above which an input signal will pass through, and below which its signal is turned off. RANGE is the amount of attenuation in dB which the noise gate attenuates the signal when the gate is off. ATTACK and RELEASE control the time characteristics of the gating action. Attack sets the amount of time it takes for the gate to open or gated signal to turn on. Release sets the time required for the gate to close back up when the input signal falls below threshold.

### 8.1h Autoleveler

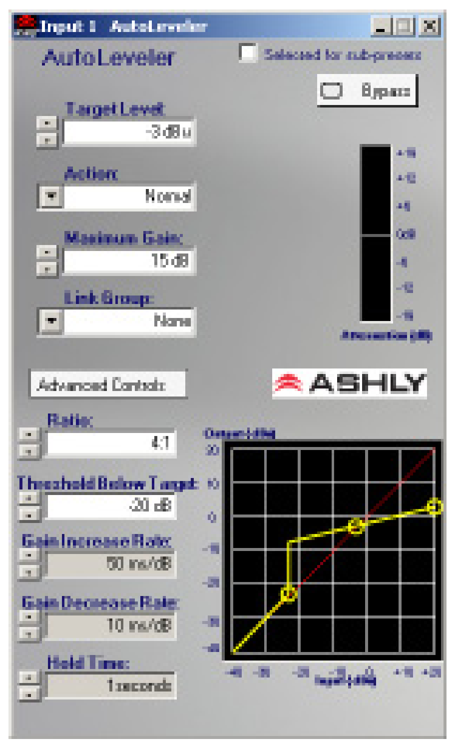
An Autoleveler is a dynamics processor used to automatically boost or cut a signal to a user defined target level. TARGET LEVEL is the primary setting in an autoleveler, as it determines the desired constant level to which an input is boost or cut. Both basic mode and advanced mode utilize the target level control, but basic mode simplifies setup. In BASIC MODE, target level, action, and maximum gain are the available controls. ACTION is selectable to gentle, normal, or aggressive, and automatically adjusts the following controls found in advanced mode:

| Action | Ratio | Hold Time | Gain Incr. Rate | Gain Dec. Rate |
|---|---|---|---|---|
| Aggressive | 10:1 | 0 Sec | 20ms/dB | 5 ms/dB |
| Normal | 4:1 | 1 Sec | 50ms/dB | 10 ms/dB |
| Gentle | 2:1 | 2 Sec | 100ms/dB | 20 ms/dB |

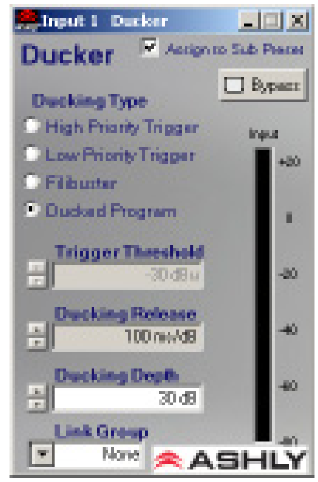
$$\text{Thr} = \frac{\text{Max Gain}}{(1/\text{ratio} - 1)}$$

MAXIMUM GAIN controls the threshold below target using the following formula:

In ADVANCED MODE, THRESHOLD BELOW TARGET determines the input signal level relative to the target level, above which the autoleveler increases gain, and below which no action is taken. RATIO is defined as the relationship of input level change in dB to output level change in dB. It is a measure of how aggressively the autoleveler changes the gain to maintain a constant output target level. GAIN INCREASE RATE and GAIN DECREASE RATE are used to prevent sudden, choppy sounding level changes to an input signal having a wide dynamic range. HOLD TIME is used in conjunction with gain change rate, and is defined as the time after the input signal falls below the threshold during which the autolever's gain is held constant before it returns to unity gain. The purpose is to reduce the amount of gain "chatter" and abrupt signal cutoff when the input signal is hovering around the threshold level. Advanced settings which are different than the three Basic action settings results in a USER DEFINED display in the basic action control.



**8.1i Ducker**




A Ducker is used to attenuate the level of selected input channels when one or more "Trigger" inputs have signal present. Input channels can be set to one of four ducker modes:

1.) BYPASS - no ducking action applies.

2.) DUCKED PROGRAM - signal present on other "Trigger" selected input channels act to attenuate this input channel.

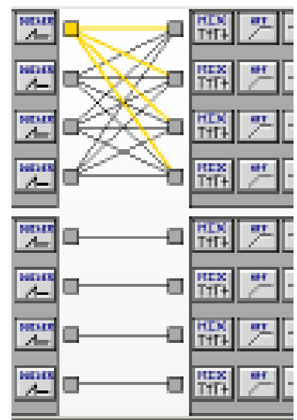
3.) LOW PRIORITY TRIGGER - signal present on this input channel attenuates all other inputs set to "Ducked Program".

4.) HIGH PRIORITY TRIGGER - signal present on this input channel fully attenuates all other inputs set to "Ducked Program", as well as all other channels set to "Low Priority Trigger".

5.) FILIBUSTER - signal present on this channel fully attenuates all other channels until the signal drops below its threshold to allow other channels to come on.

TRIGGER THRESHOLD determines the signal level on a trigger input at which point the ducked program is attenuated. DUCKING DEPTH determines how much attenuation is applied to input channels set up as ducked program. DUCKING RELEASE determines the rate at which the attenuation returns to 0dB after the trigger input signal has stopped.

**8.1j Matrix Routing**

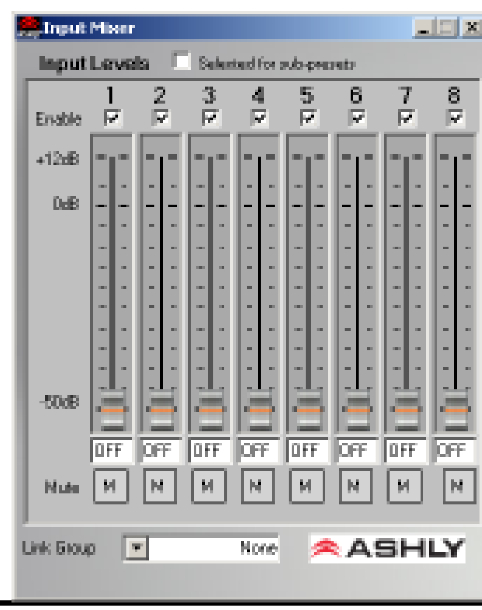


Any input channel can be routed to any or all output channels, likewise any output channel can have as its source any or all inputs, limited only by the installed capacity of input or output expansion modules. To route an input to an output, click and drag from the input connect box to the desired output connect box (click and drag routing works in both directions). To delete a connection, click on the connect line and press <Delete> on the keyboard, or right click on the connection line for the delete prompt. To quickly perform multiple routing connections or disconnections to the current input channel, hold <Ctrl> and click on desired individual output channel connect boxes to toggle them on or off. Anytime a connection is made, the corresponding output mixer level is set to “-inf”, (off).

**8.1k Remote Level Control**

Each channel can be remotely attenuated through the back panel 0-5V Remote Level Control pins, or by an Ashly RD8C active remote attenuator connected to the 4-pin data connector on the back. The Remote Level Controls dialogue box then allows arbitrary assignment of the remote’s attenuator channels to the ne24.24M’s audio channel.

## 8.2 Output Audio Functions

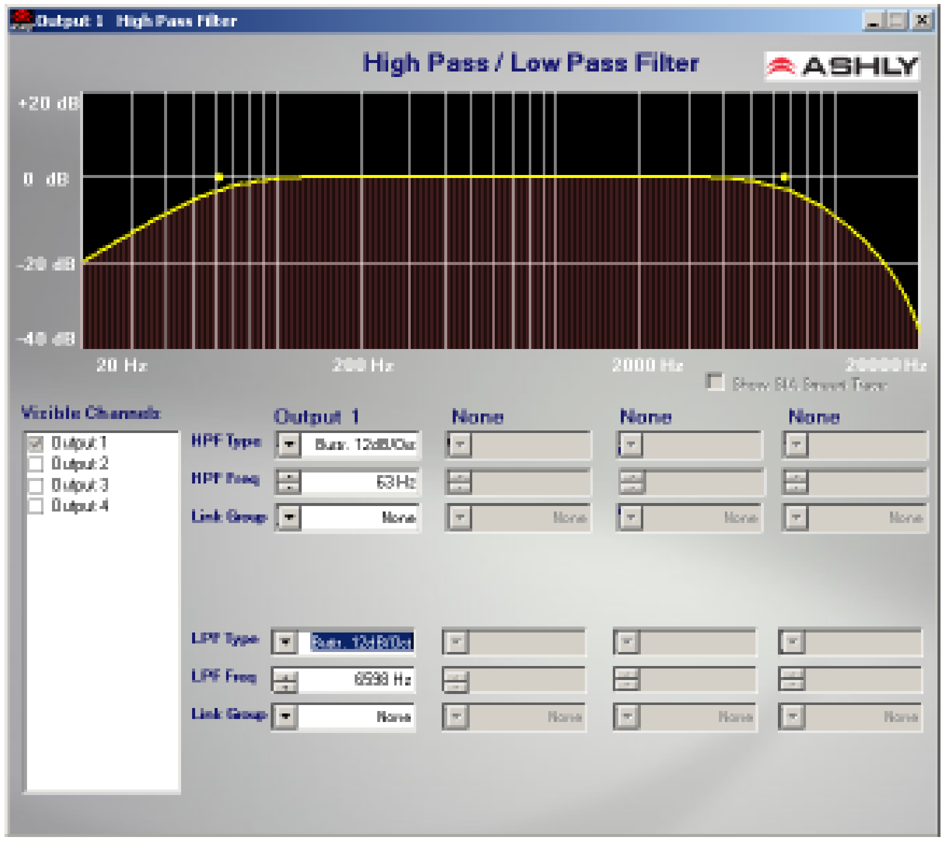



The following functions are available on all outputs: Mixer, High Pass/Low Pass Filter, Delay, EQ, Gain, Limiter, and Mute.

**8.2a Output Matrix Mixer**

Output channels can have any or all inputs as sources, and the output matrix mixer allows the level of each connected input to be adjusted in the “matrix mix” to each output. Note that for a given output, unlinked inputs are disabled in the mixer. Adjustment range is from +12dB to off. To set a level control to unity (0) within the matrix mixer, select an input level control by clicking on it and pressing <ctrl> + <click> on the level adjustment control. Individual input channels in the matrix mixer can be muted independently from the main input or output mute blocks.

### 8.2b Hpf/Lpf - Crossover




Bandpass or crossover functions on the Protea ne24.24M are available only on the output channels. Every channel's crossover consists of a high pass filter (HPF) and a low pass filter (LPF), along with the frequencies and filter types used. Each output's crossover section is essentially a bandpass filter, making it necessary for the user to map out ahead of time which outputs will be used for the various frequency bands, and set the overlapping filter frequencies and types accordingly. *Note: The 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.*

The frequency range for the high pass filter (HPF) is 20Hz to 20kHz, with an option to turn the filter off at the low end of the frequency selection. The low pass filter (LPF) offers the same frequency range, with the "off" option at the high end of the frequency selection.

There are eleven types of filters available in the crossover section, each suited to a specific preference or purpose. The slope of each filter type is defined by the first characters in the filter type, 12dB, 18dB, 24dB, or 48dB per octave. The steeper the slope, the more abruptly the "edges" of the pass band will drop off. There is no best filter slope for every application, so experiment to see which one sounds most pleasing in a specific system. The Ashly default crossover filter is 24dB/octave Linkwitz-Riley, but of course they can be changed to suit the application.

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 a summary of the three filter types as used in the ne24.24M 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 on the ne24.24M, 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.

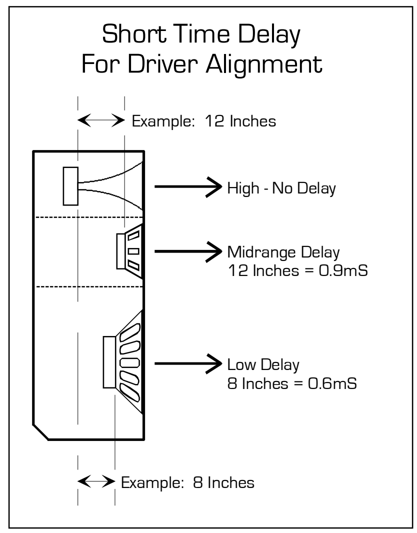
**Linkwitz-Riley**

The 12 dB/oct, 24dB/oct, an 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.

**8.2c Output Delay**

Output delay can be used to time align discrete drivers within a cabinet or cluster using short delay times, or align multiple drivers in different locations using longer delay times. For a thorough long-delay explanation, see section 9.1e. The following example illustrates a use of short delay to time align speakers within a group: A typical three way speaker cluster has low end, midrange, and high frequency drivers all located near one another. The different drivers for each frequency band are not necessarily the same physical depth with respect to the front of the loudspeaker cluster, so there exists the problem of the same signals (at the crossover points) arriving at the cluster "wavefront" at different times, creating undesirable wave interaction such as frequency peaks or cancellation. The solution in this case, rather than fixing the frequency anomalies with EQ, is to slightly delay the signal to the drivers closest to the cluster front.




Using the location of the driver diaphragm farthest back as a reference point, measure the distance to other drivers in the cluster, and set the output delay for each accordingly, with the driver diaphragm closest to the front getting the longest delay and the driver at the very back getting no delay at all. The minimum adjustment is 0.02 milliseconds, or about 1/4 inch. When appropriate, always time align the loudspeakers before applying EQ to the outputs of the ne24.24M.

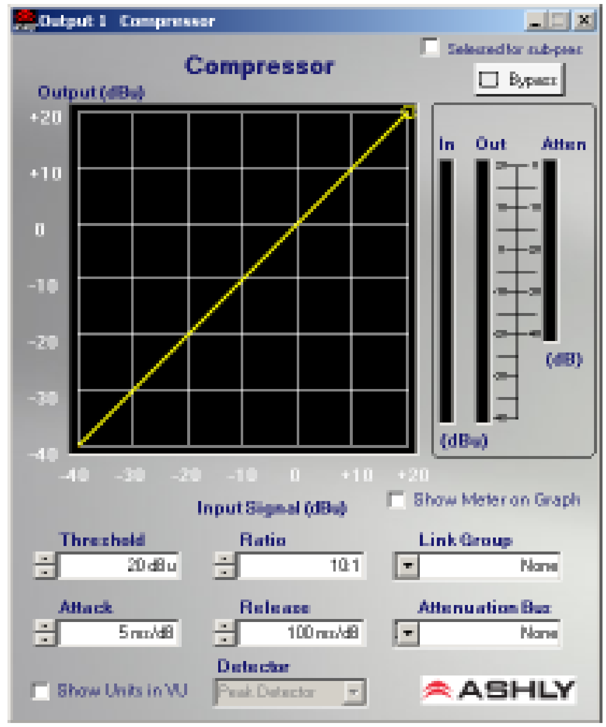
**8.2d Output EQ**

The Protea ne24.24M Output EQ section is the same as the input EQ (see section 9.1f), with the exception of the ability to view the combined effect of input EQ for each installed and linked input channel to a given output channel. Within the output EQ frame, each installed and linked input channel has its own <Overlay Input EQ> check box, through which the interaction between input and output EQ is displayed.

**8.2e Output Gain**

Output Gain operates in the same manner as Input Gain (section 9.1d), ranging from +12dB to Off, with an option to reverse polarity.

### 8.2f Output Compressor/Limiter




A full function compressor/limiter is included on each output channel. A limiter is commonly used to prevent transient audio signal spikes from damaging loudspeakers, manage analog and digital recording levels, optimize broadcast levels, or "thicken" the sound of an audio source (compression). The adjustable parameters include Limiter In/Out, Threshold, Ratio, Attack Time, Release Time, and Link Group, and Attenuation Bus.

The ne24.24M limiter threshold range is -20dBu to +20dBu, or -24VU to +16VU if the metering option is selected to VU. The Threshold control determines the signal level above which gain reduction begins, and is indicated by an amber LED (S/L) on the ne24.24M face panel, as well as indicated in the Matrix Meters in software. Increases in audio level above the threshold will be reduced according to the ratio settings.

The Ratio control determines the amount of gain reduction above limiter threshold. Ratio ranges from a gentle 1.2:1 to a very abrupt INF:1. To illustrate how the ratio control works, imagine a commonly used loudspeaker protection ratio of 10:1, which means that for every input signal increase of 10 dB above threshold, the output level will only increase by 1dB. The higher the ratio, the more pronounced the audio effect, so use the lowest ratio possible to sufficiently address the problem.

Attack and Release settings adjust the time it takes the limiter to engage and then disengage when the signal increases above threshold and then subsequently falls back below threshold. Attack time is adjustable from 0.2ms/dB through 50ms/dB, while release time ranges from 5ms/dB through 1s/dB. A very fast attack time can sound unnatural, while a very long attack time can miss some of the initial transient. Similarly, a very short release time can make the audio sound uneven, while a very long release time can create "pumping", or "breathing" characteristics depending on the kind of signal. Experiment to find the best solution for a given application.

The Attenuation Bus allows up to four output channels within a group to share a threshold detector, so that any channel with a transient signal above threshold will apply equal gain reduction to all other channels within that group which is assigned to the link bus. The channel which furthest exceeds threshold will determine the resulting reduction on all channels selected to the attenuation bus. The channel limiter attenuation bus is particularly useful when processing stereo signals.

### 8.2g Output Mute

Output Mute turns off an output channel. When an output channel is muted, that channel's red mute LED on the face panel is lit. To mute or unmute all outputs at once, go to the <Mute> menu heading.

# 9. OTHER SOFTWARE FUNCTIONS

## 9.1 Channel Preset (and Sub Preset) File Management

The Protea ne24.24M will store up to 31 named internal presets, each preset storing control data for all channels and audio functions. Preset names must be 20 characters or less. While working in Protea NE Software, changes to an individual preset can be saved to the ne24.24M using <File/Save Preset To Protea>, or saved to the PC using <File/Save Preset To Disk>. Individual preset files use the extension (*.pne).

**Sub Presets:** Instead of saving or recalling an entire preset affecting all functions, a sub-preset affecting only a sub-set of functions may be used. To save a sub-preset, check the boxed labelled <Selected for sub-presets> in the audio control functions to be stored in the sub preset, and then click <preset options>, then <Save Sub Preset>.