



VoiceBox 4 I/OP User Manual

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IMPORTANT SAFETY INSTRUCTIONS

The symbols below are internationally accepted symbols that warn of potential hazards with electrical products.

 This symbol, wherever it appears, alerts you to the presence of uninsulated dangerous voltage inside the enclosure -- voltage that may be sufficient to constitute a risk of shock.

 This symbol, wherever it appears, alerts you to important operating and maintenance instructions in the accompanying literature. Please read the manual.

1. Read these instructions.
2. Keep these instructions.
3. Heed all warnings.
4. Follow all instructions.
5. Do not use this apparatus near water.
6. Clean only with a dry cloth.
7. Do not block any ventilation openings. Install in accordance with the manufacturer's instructions.
8. Do not install near any heat sources such as radiators, heat registers, stoves, or other apparatus (including amplifiers) that produce heat.
9. Protect the power cord from being walked on or pinched particularly at plugs, convenience receptacles, and the point where they exit from the apparatus.
10. Only use attachments/accessories specified by Attero Tech
11. Unplug this apparatus during lightning storms or when unused for long periods of time.
12. Refer all servicing to qualified service personnel. Servicing is required when the apparatus has been damaged in any way, such as power-supply cord or plug is damaged, liquid has been spilled or objects have fallen into the apparatus, the apparatus has been exposed to rain or moisture, does not operate normally, or has been dropped.
13. This apparatus shall be connected to a mains socket outlet with a protective earthing connection.
14. When permanently connected, on all-pole mains switch with a contact separation of at least 3mm in each pole shall be incorporated in the electrical installation of the building.
15. If rack mounting, provide adequate ventilation. Equipment may be located above or below this apparatus but some equipment (like large power amplifiers) may cause an unacceptable amount of hum or may generate too much heat and degrade the performance of this apparatus,
16. This apparatus may be installed in an industry standard equipment rack. Use screws through all mounting holes to provide the best support.

 TO REDUCE THE RISK OF FIRE OR ELECTRIC SHOCK, DO NOT EXPOSE THIS APPARATUS TO RAIN OR MOISTURE.

Note: This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to Part 15 of the FCC Rules and EN55022. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference, in which case the user will be required to correct the interference at his own expense.

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1 - Overview

The VoiceBox 4 I/OP CobraNet Audio Interface is a full featured 4 input, 4 output mic/line preamp with an integrated CobraNet interface for sending and receiving digital audio over a CobraNet network. In addition, the VoiceBox 4 I/OP provides a dedicated DSP for compression, equalization, and gain on each input channel as well as limiting on each output channel. Audio is routed using an internal matrix mixer that allows flexible audio routing between all available local and CobraNet I/O.

The VoiceBox 4 I/OP features the following connectivity:

- Audio I/O - 4 balanced inputs, all with gain and +12 V phantom power selectable on a per channel basis, and 4 line level balanced outputs.
- CobraNet Interface Port - Connects the VoiceBox 4 I/OP to a CobraNet network using standard CAT-5 cable.
- RS-232 Port - Enables the transmission of serial data from any VoiceBox 4 I/OP to any other VoiceBox 4 I/OP on a CobraNet system.

It can be powered by either using PoE from a standard 802.3af compatible PoE network switch or injector, or by using an external +12VDC source.

Configuration of the VoiceBox 4 I/OP is done using the Attero Tech Control Center software. This control software allows real-time manipulation of all signal routing and signal processing.

1.1 – What's in the Box?

The VoiceBox 4 I/OP comes supplied with the following

- VoiceBox 4 I/OP device

1.2 – Optional Extras

All of the following are available as options for the VoiceBox 4 I/OP:

- A 12 V DC power supply is available if PoE power is not required or available.
- Single device 1U rack mount kit.
- Dual device 1U rack mount kit

Each optional item required must be ordered separately to the VoiceBox itself.

1.3 – System Signal Flow

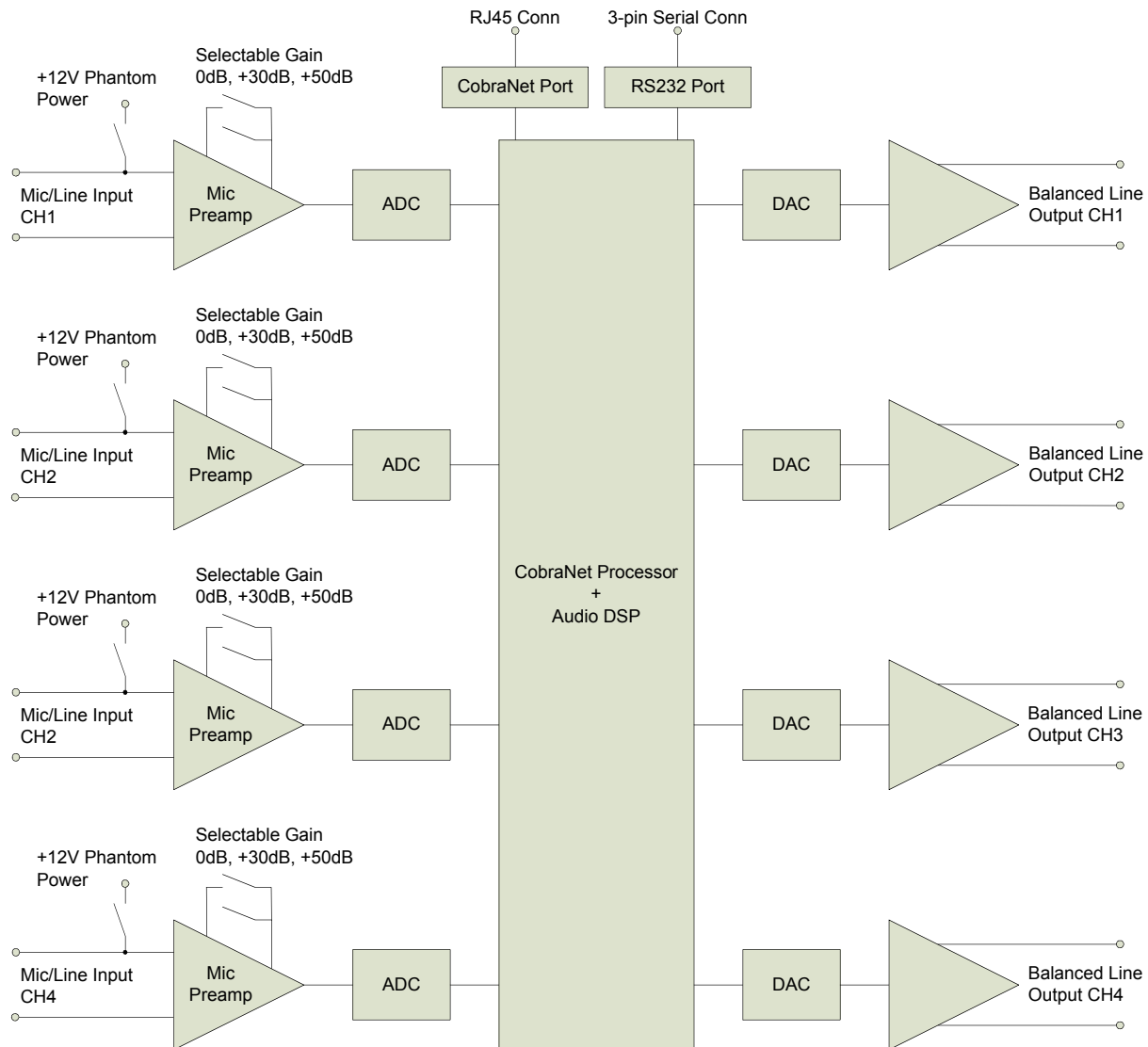


Figure 1 - Signal Flow within the VoiceBox 4 I/OP

System signal flow is shown in Figure 1. All four input channels utilize discrete preamplifier circuitry for optimum signal to noise ratio with low distortion. Each preamplifier is equipped with 3 selectable analog gain settings, 0 dB for line level sources, +30 dB for condenser microphones, and +50 dB for dynamic microphones. Each channel also has selectable +12 V phantom power for powering most electret or small diaphragm condenser microphones.

After the mic/line preamp stage, the analog signal is converted using a high quality 24-bit/48 kHz A-D converter. After conversion, the digital signal is then sent to the CobraNet processor and DSP. The DSP applies independent and adjustable equalization (low cut, low shelf, mid sweep, high shelf, and high cut filters), compression, and gain to each input channel.

A fully configurable thirteen input, twelve output matrix mixer allows the processed input channels to be sent out to the CobraNet network and/or routed back to the local outputs. The matrix mixer also provides the ability to route audio received from the CobraNet network to the local outputs and/or be looped back to the CobraNet outputs. A further input into the matrix mixer is provided from a configurable sine wave generator for testing and diagnostic purposes.

Each of the four local output channels also features five bands of fully adjustable parametric EQ, a high pass filter, a fully adjustable limiter, additional gain/attenuation and a delay element with up to 20ms of delay. The digital audio is then sent to a high quality D-A converter. The analog signal is then converted to a balanced line level output and sent to the local audio output connectors.

1.4 – Signal Processing

All signal processing after the analog preamplifier circuitry is implemented in the digital domain using powerful 32-bit DSP hardware and high quality algorithms. All signal processing parameters mentioned are configurable using the Attero Tech Control Center software.

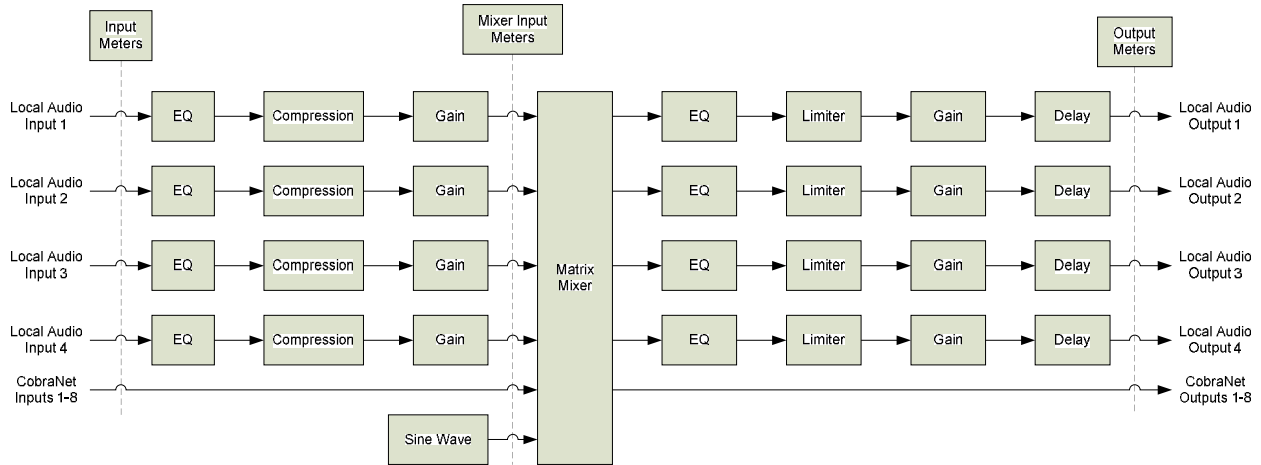


Figure 2 - DSP Architecture

Input EQ

The EQ stage consists of five cascaded filters: A single pole low cut filter, a 2-pole high cut filter, a low frequency shelf filter, a mid sweep filter, and a high frequency shelf filter. The low and high cut filters both have adjustable frequencies and have a bypass option. The low frequency shelf filter is set at 400 Hz and has adjustable gain. The high frequency shelving filter is set at 3 kHz and also has adjustable gain. The midspan equalizer has adjustable center frequency and gain. Gain for each filter is adjustable from +12 dB to -12 dB.

Compressor

The compressor stage provides full control over the compression ratio, threshold, attack, and release times. A gain reduction meter is shown also that shows the amount of compression taking place. The compressor function has a bypass option.

Input Gain

The input gain stage provides an adjustable range from +12 dB gain to -100 dB of attenuation in 1 dB steps.

Matrix mixer

The matrix mixer provides 12 inputs (4 local, 8 CobraNet) and 12 outputs (4 local, 8 CobraNet). There is also an additional input from an internal sine wave generator. Any mix of local inputs, network inputs and sine wave generator input can be directed to any of the local or network outputs.

Output EQ

The output EQ stage consists of five cascaded parametric filters and a high pass filter. Each parametric filter has adjustable gain, bandwidth and frequency and the 2-pole high pass filter has adjustable frequency. Gain for each parametric filter is adjustable from +12 dB to -12 dB.

Output Limiter

The output limiter provides limiting for each output channel independently. Limiter ratio is fixed at 100:1. Threshold, attack, and release time parameters are configurable. A gain reduction meter shows the amount of limiting taking place. The limiter function may be bypassed.

Output Gain

The gain stage provides a range from +12 dB gain to -100 dB of attenuation. Gain settings are adjustable in 1 dB steps.

Output Delay

Each output has its own delay control with up to 20ms of delay available.

Metering

Metering is available from several points in the audio path: Raw local input, Modified local input, Local output. CobraNet input and CobraNet output.

2- Installation

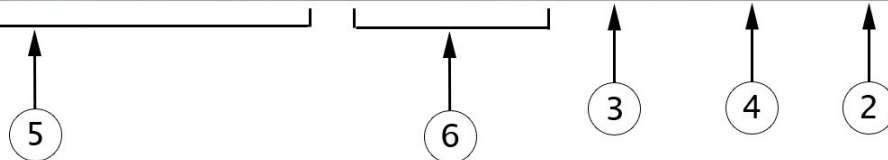
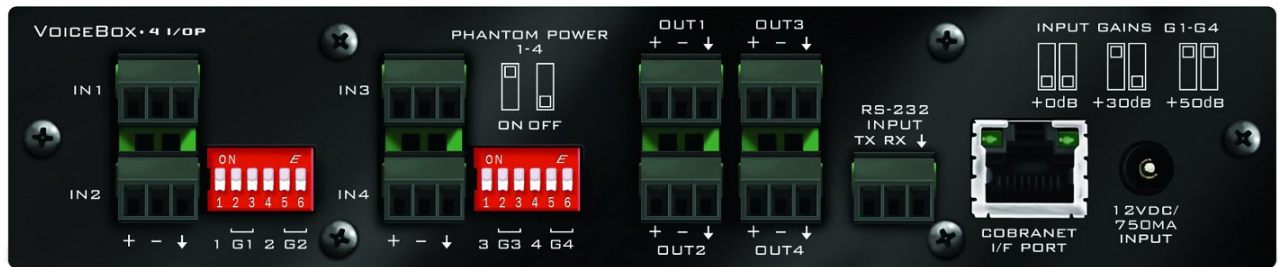


Figure 3 - VoiceBox 4 I/OP Connectors and Indicators

- ① Power LED
- ② Power Socket - Use with optional 12 V DC wall wart only.
- ③ Serial Bridge interface connector
- ④ CobraNet Ethernet interface connector and indicators
- ⑤ 4 x Balanced audio inputs and associated mic/line and phantom power switches
- ⑥ 4 x Balanced Audio outputs

Installation of the VoiceBox is very straight forward. All connections to the VoiceBox should be made before the power is applied.

- Attach any audio sources that will be used to the inputs. The inputs are balanced so be sure to check what output type the source is in order to find how to connect it correctly (see section 2.1 – Hardware Connections).
- Select phantom power and/or the level of input gain required as necessary.
- Attach the outputs to the required audio devices. The outputs are balanced so be sure to check what input type the device requires in order to find how to connect it correctly (see section 2.1 – Hardware Connections).

When powering using PoE:

- Attach the CobraNet I/F port to a spare PoE-enabled port on a PoE switch using a CAT-5 cable. If a midspan injector is being used, connect a spare input port to the CobraNet network switch using a CAT-5 cable, and then connect the corresponding output port to the CobraNet I/F of the VoiceBox.

When powering using an optional external supply:

- Attach the CobraNet I/F port to a spare port on the CobraNet network switch using a CAT-5 cable.
- Attach the power supply to the power input jack and then power up the external supply.

If all steps are performed correctly, the power light on the front should be lit.

There may also be some activity on the VoiceBox CobraNet I/F LED indicators. With no CobraNet network, the LED on the right of the CobraNet I/F port will be on. If another active CobraNet device is detected on the network or the Attero Tech Control Center application or other similar CobraNet application program is active on the network, the left hand LED will be flashing at a rate around three times per second and the right hand LED may either also flash or remain steady.

2.1 – Hardware Connections

The VoiceBox 4 I/OP accepts and drives either unbalanced or balanced audio devices. Refer to the following diagrams and instructions for connecting different types of audio devices. Professional grade audio cabling is recommended to achieve the best audio performance throughout the system.

2.1.1 – Input from an Unbalanced Source

To connect a 2-wire unbalanced source to the VoiceBox 4 I/OP, connect the positive output of the unbalanced source to the positive input of the VoiceBox 4 I/OP. Connect both the source and VoiceBox 4 I/OP input grounds together, and short the negative input of the VoiceBox 4 I/OP to ground at the input of the VoiceBox 4 I/OP.

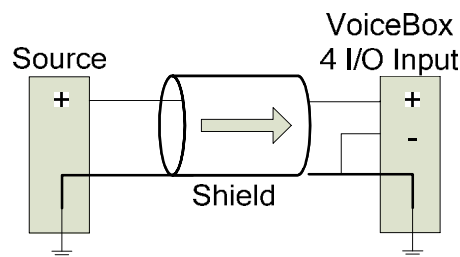


Figure 4 - 2-Wire Unbalanced Source Connection

To connect unbalanced sources with a 3-wire connection, short the negative conductor to the shield at the source connection.

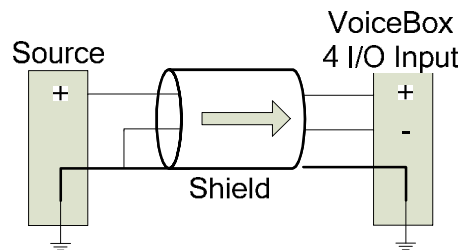


Figure 5 - 3-Wire Unbalanced Source Connection

2.1.2 – Input from a Balanced Source

To connect balanced sources to the VoiceBox 4 I/OP, connect positive output to positive input, negative output to negative input, and connect the grounds together through the cable shield.

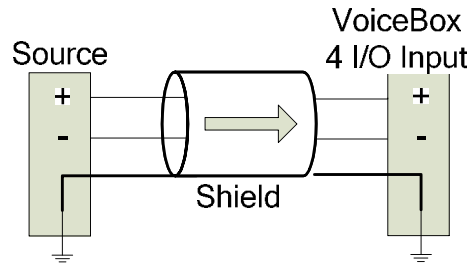


Figure 6 - Balanced Source Connection

2.1.3 – Output to a Balanced Destination

To connect to a balanced input on a destination device, connect the positive, negative, and ground connections of both the VoiceBox 4 I/OP output to the destination input respectively.

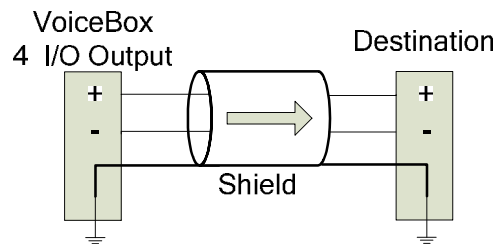


Figure 7 - Out to Balanced Destination Connection

2.1.4 – Output to an Unbalanced Destination

To connect the VoiceBox 4 I/OP outputs to a 2-wire unbalanced input, connect the positive output to the destination's positive input and connect the grounds together through the cable shield. Leave the VoiceBox 4 I/OP negative output floating.

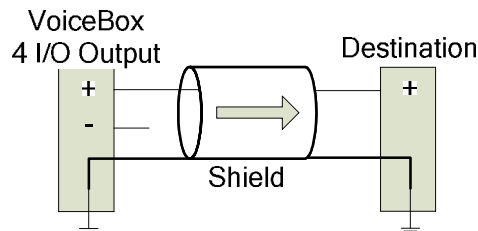


Figure 8 – Output to Unbalanced Destination

2.1.5 – RS-232 Serial Bridge Port

The RS-232 Serial Bridge Port located on the rear panel of the VoiceBox 4 I/OP implements RX, TX, and GND connections. The 3-Pin de-pluggable connector can be used to interface with other RS-232 enabled equipment - for example, a computer serial port. The TX signal is an output from the VoiceBox 4 I/OP and the RX signal is an input. Connections to other RS-232 hardware should be made accordingly.

Note: The RS-232 port does not communicate locally with the VoiceBox 4 I/OP, but is a conduit to other devices on the CobraNet network. The Serial Bridge feature of CobraNet allows serial data to be sent across the network to other serial-enabled devices. This feature is useful for extending control data to remote locations. The settings for the serial bridge can be configured using the Attero Tech Control Center software.

2.2 – Phantom Power and Gain Switches

The VoiceBox has the ability to provide phantom power and input gain on each input. Switches are provided for controlling both these features independently on each input. G1 to G4 are 4 groups of switches controlling input gain on inputs 1 to 4 respectively. The remaining four switches, number 1 through 4 are the four phantom power switches for inputs 1 to 4 respectively.

The two gain switches per input provide a choice of line level input (0 dB), +30 dB gain, or 50 dB gain. To use a line level input, ensure both gain switches for that channel are in the down position. To set +30 dB of input gain, move the left hand switch of the gain group to the up position. To set the input gain to 50 dB, move both switches in the gain group to the up position.

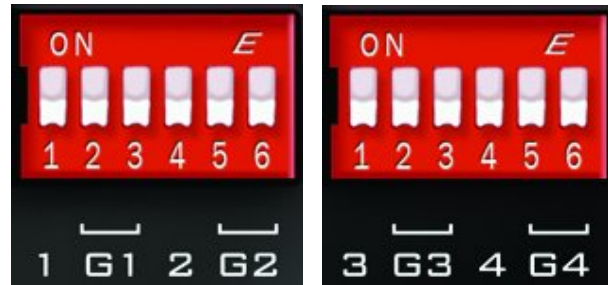


Figure 9 - Gain and Phantom Power switches

Phantom power enables the VoiceBox 4 I/OP to provide +12 V DC at very low current over the audio input lines to power an active microphone. With the switch in the down position, (as they all are in Figure 9), phantom power is not enabled. Move the switch to the up position to activate it.

Note: The positions of the switches for both gain and phantom power are also shown on the back panel itself for easy reference.

3 – Software Installation and Setup

The Attero Tech Control Center application should be used to examine and modify device configuration. This includes not only the standard CobraNet features such as audio routing but also the device specific controls of the signal processing blocks the VoiceBox uses.

Refer to the Attero Tech Control Center User’s Guide for installation and setup instructions.

Note: Other third party software may be used to set up audio routing but the signal processing blocks can only be controlled using the Attero Tech Control Center application.

3.1 – DSP Overview

The following section describes the operation and controls available in VoiceBox 4I/O PoE interface V1.1.1.133.

Note: Control Center version 1.0.2.12 and later shows the version number for each interface on the About form. Please be aware of the version of interface being used, as a different version of the interface may contain different controls and have different operation.

The VoiceBox 4 I/OP utilizes available DSP resources within its CobraNet device to provide audio signal processing (see Figure 10). The local inputs of the VoiceBox are fed through a Compressor stage, an EQ stage, and a Gain stage. The processed signals are then presented along with eight incoming channels of CobraNet audio to a matrix mixer. After the Mixer stage, the local outputs pass through a Limiter stage before being fed to the D/A convertors.

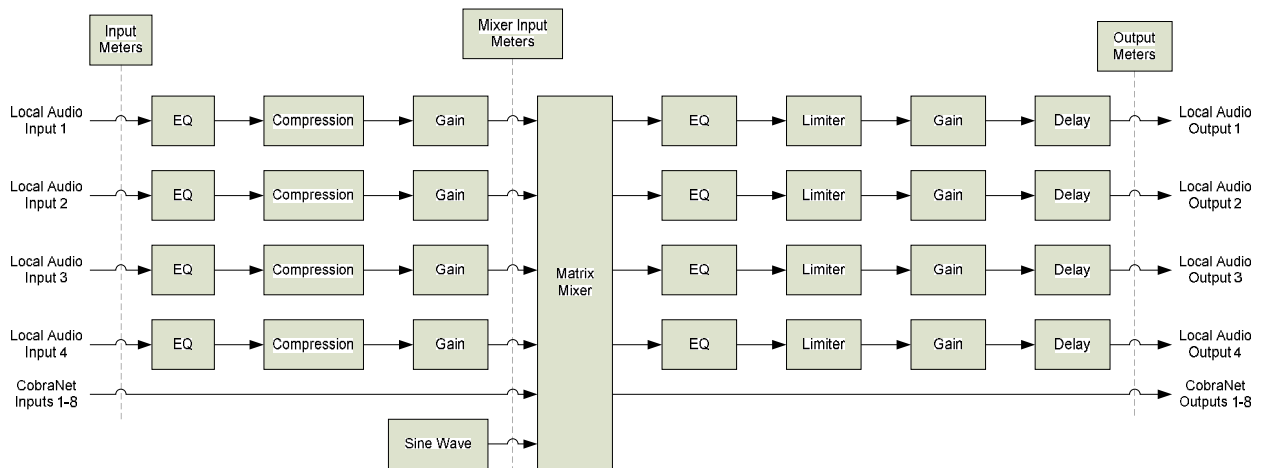


Figure 10 – VoiceBox 4 I/O DSP Architecture

The interface for the VoiceBox 4 I/OP allows configuration of all aspects of the signal processing blocks shown in Figure 10. The processing is split into three sections: Input Processing, Matrix Mixer, and Output Processing.

The input processing affects signals input into the local inputs on the VoiceBox only. The Matrix Mixer takes the results of the input processing, CobraNet audio received by the VoiceBox and also an internally generated sine wave and mixes them together to produce four local output channels and eight CobraNet output channels. The CobraNet output channels are ready for immediate transmission elsewhere with no further processing. However, the local output channels are then subjected to the output processing before being passed to the outputs themselves.

3.2 – Input Processing

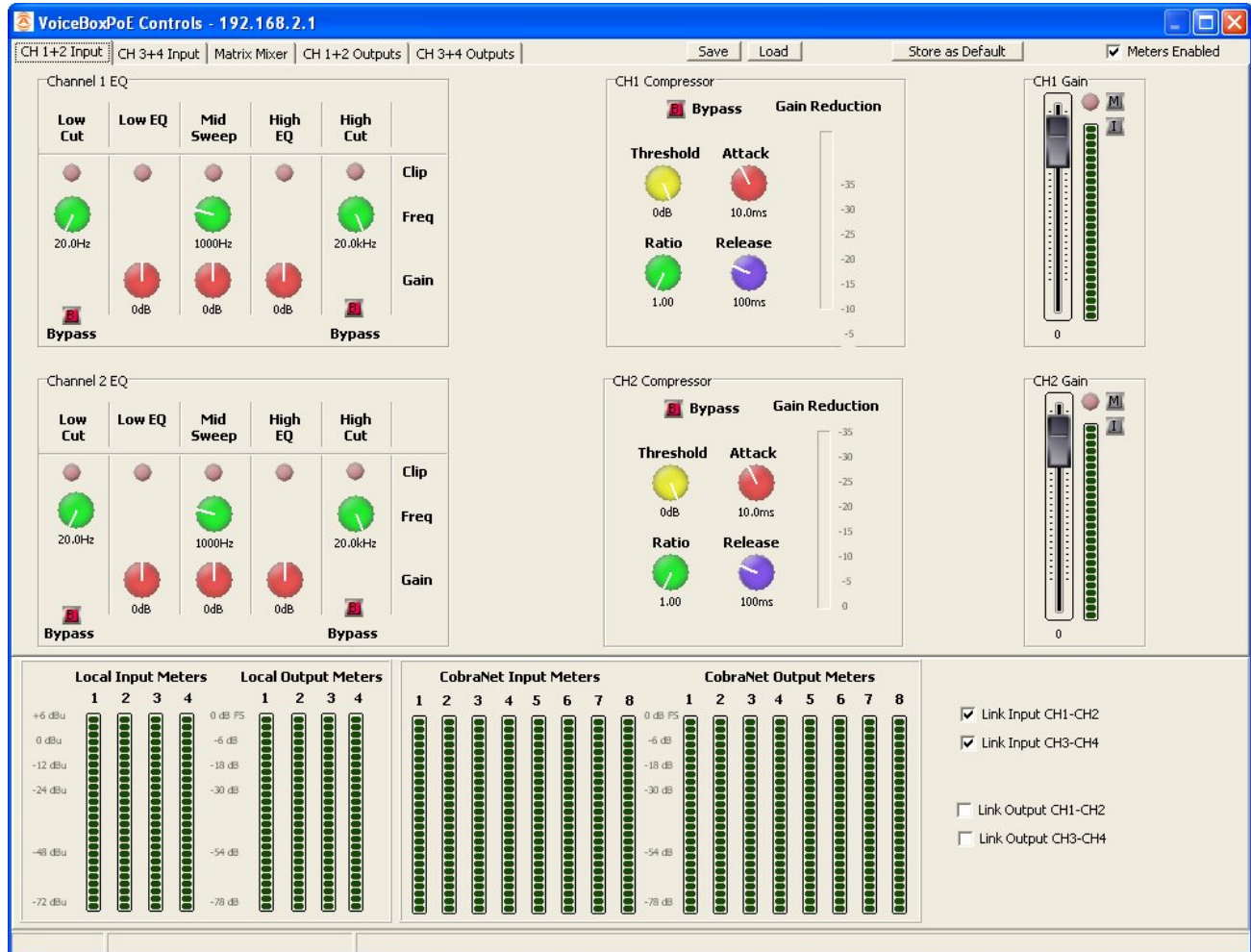


Figure 11 – Local Input Channel Processing for Channels 1 and 2

Figure 11 shows the general layout of the form. It contains five tabs. The first two tabs are dedicated to local input processing with two input channels on each tab. The third is dedicated to the Matrix Mixer. The final two tabs are dedicated to local output processing with two output channels on each tab.

For controls that utilize either a control knob or a fader, the value can be altered by clicking and holding the left mouse button down on the control and dragging the mouse forward or backward to alter the value upward or downward respectively.

The value can also be typed in if the controls have their value shown underneath. Clicking on the control will highlight the complete text and typing a new value will overwrite the current text. The controls accept shorthand notation such as 10k for 10000 and 10m for 0.01. They will also accept the correct unit if entered, though it is not necessary to include it. Once the required value is entered, press the Enter key to complete the process and accept the new value. If a valid number is entered, the control will change to the newly entered number or as close to it as the control's range allows. The control will revert back to its previous value if an invalid number is entered.

Notes:

- If the Enter key is not pressed after entering a new value and a different control is selected, the control that was being edited will revert to its previous value.
- When a fader or knob control value is changed, its value is sent to the device only once the Enter key is pressed or the mouse is released.

The other type of control is an indicator button. These are either grey for off or colored (red, green, or yellow) for on. To change the state of the button from on to off or from off to on, left-click on it.

3.2.1 – Input EQ

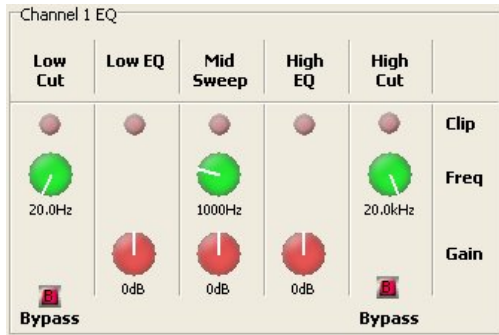


Figure 12 - Input EQ Controls

The input EQ consists of several cascaded filters: a low and high cut, a low EQ, a mid sweep and a High EQ.

The low cut is a 12dB / Octave filter with adjustable frequency. The high cut is a 6dB / Octave filter with adjustable frequency. The low EQ is a low shelf filter set at 400 Hz with adjustable gain. The mid sweep is a parametric EQ with a fixed bandwidth of 1 Octave but adjustable frequency and gain. The high EQ is a high shelf filter set at 3 kHz with adjustable gain.

In each case the frequency can be adjusted from 20 Hz to 20 kHz and the gain controls work from -12 dB to +12 dB. The high cut and low cut filters also have a bypass facility which, when active, will cause that filter to be bypassed and have no affect on the audio signal.

Each stage also has a clipping LED which lights red when clipping occurs.

3.2.2 – Input Compressor

The Input Compressor consists of four controls, a Bypass button, and a Gain Reduction meter.

The Bypass button allows the compressor to be completely bypassed and its settings ignored. Click on the button to toggle the state of the bypass. If the red indicator is on, the bypass is activated.

The Threshold value ranges from -100 dB to 0 dB. The Ratio value ranges from 1:1 to 100:1. The Attack value ranges from 1ms to 1s and the Release value ranges from 10 ms to 30 s.

If any compression takes place, the Gain Reduction meter will show the amount of signal attenuation.

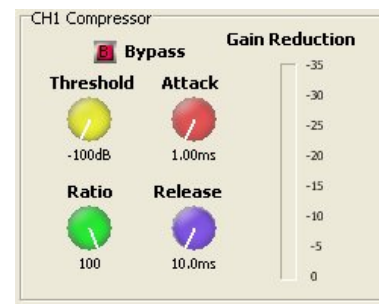


Figure 13 - Input Compressor Controls

3.2.3 – Input Gain

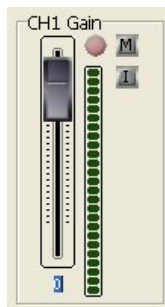


Figure 14 - Input Gain Controls

The Input Gain control allows the overall level of the signal to be adjusted before it is fed into the mixer.

The fader sets the gain from -100 dB to +12 dB. The meter alongside shows the current level after all the input processing has taken place, including the gain factor. There is also a clipping LED.

The gain control also has two toggle indicators. The upper one marked with an “M” is the “Mute” button. The lower button marked with an “I” is the “Polarity Invert” button.

3.3 – Matrix Mixer

The Matrix Mixer (shown in Figure 15) combines a number of controls. Each of the Local and CobraNet inputs are shown in columns with the send controls running across in rows.

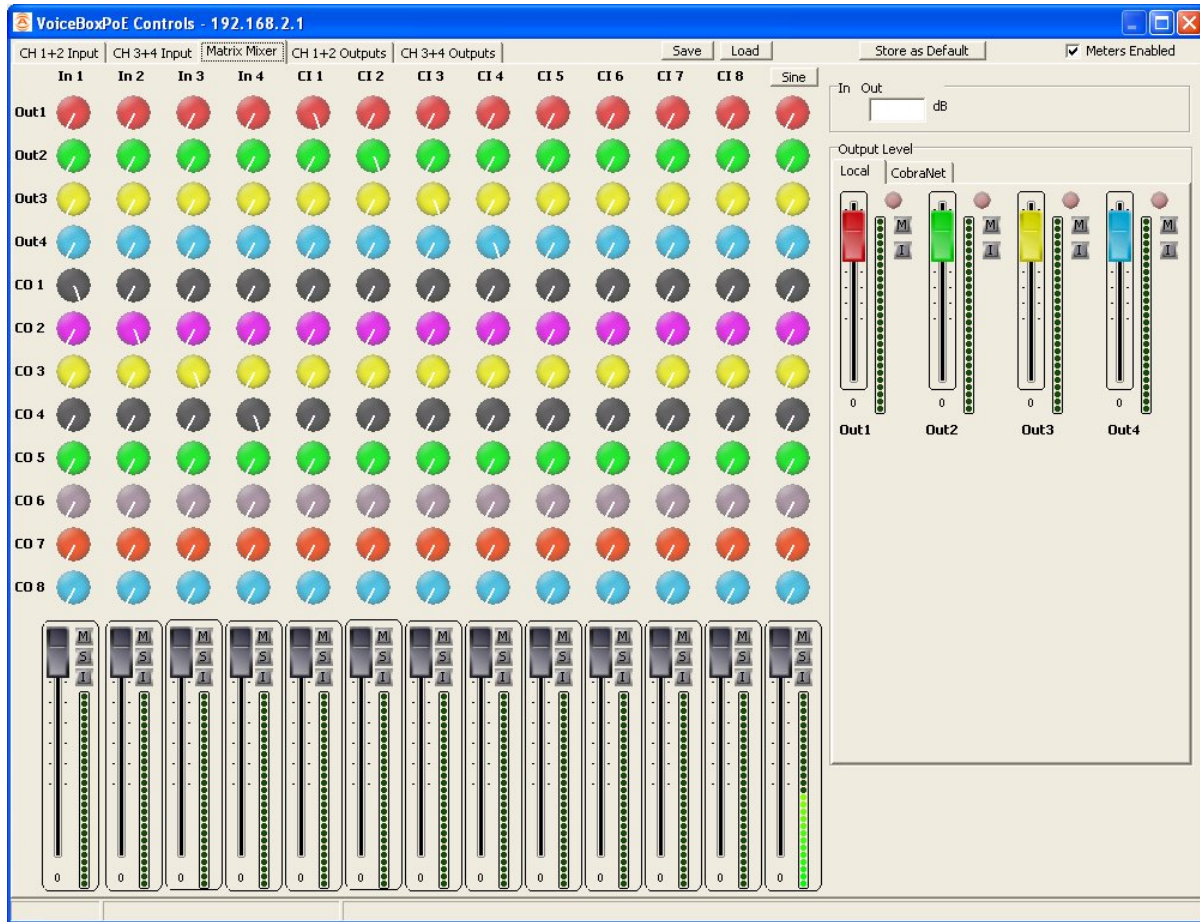


Figure 15 – Matrix Mixer Tab of VoiceBox Interface

3.3.1 – Matrix Mixer Input Controls

Each input has its own set of controls at the bottom of the column. There is an input fader that ranges from -100 dB to 0 dB as well as “Mute”, “Invert” and “Solo” buttons.

Note: The Matrix Mixer input meters can only show the levels prior to entering the Matrix Mixer. Changing any of the Matrix Mixer input controls will not affect the level of the accompanying input meter.

3.3.2 – Crosspoint Controls

Each input has a set of corresponding crosspoint knob controls which feed the audio (or a degree of it) to a particular output. Each knob control ranges from -100 dB to 0 dB. Unlike other knob controls elsewhere on the form, there is no text below each control. Instead, there is an edit box at the top right hand corner of the matrix. A single left click on a particular crosspoint will show the control’s value in the edit box. This can also be used to type in a specific value for the selected control.

3.3.3 – Matrix Mixer Output Controls

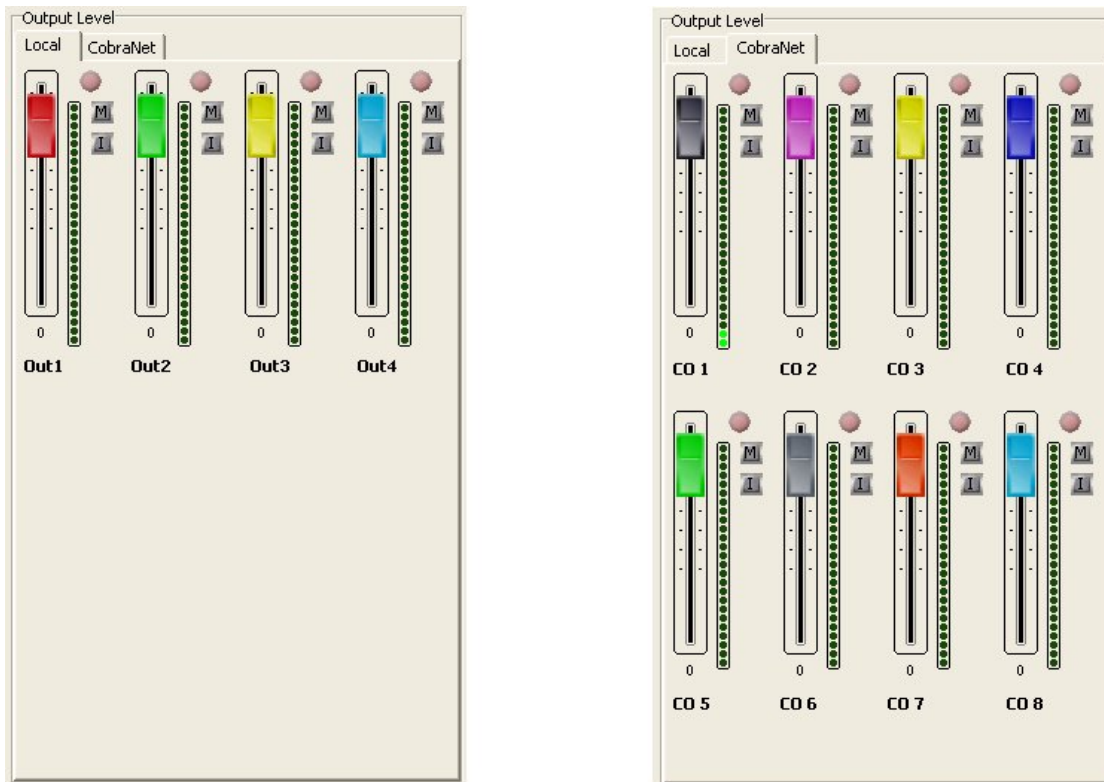


Figure 16 - Matrix Mixer Output Controls: Local (left) and CobraNet (right)

The output section of the Matrix Mixer is split into two tabs. The first tab shows the local outputs. The second tab shows the CobraNet outputs.

The mixer sends are colored according to the corresponding output level fader. The output faders range from -100 dB to +12 dB.

There are also a number of complementary controls: a clipping LED, a “Mute” button, and a “Polarity Invert” button.

The meters on the CobraNet outputs show the output levels from the Matrix Mixer that are available to the network. However, the meters on the local outputs show the levels after any output processing has been applied rather than the direct output of the matrix mixer.

3.3.4 – Sine Wave Generator

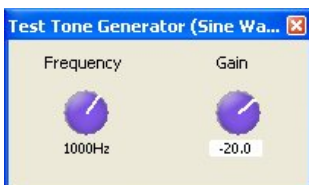


Figure 17 - Sine Wave Control Form

The Test Tone or Sine Wave Generator can be used to help diagnose problems. It is generated internally by the on-board DSP of the VoiceBox. The output of the generator is fed into the Matrix Mixer (furthest input to the right). Above the column is the “Sine” button which, when clicked on, accesses the Sine Wave Control form. The form has a Gain control to set the amplitude of the signal between -100 dB and 20 dB. There is also a Frequency control to set the frequency of the test tone which ranges from 20 Hz to 20 kHz.

3.4 – Output Processing

The audio for the local outputs is passed from the matrix mixer to the output processing stage. Figure 18 below shows the form for outputs 1 and 2. The layout is identical for outputs 3 and 4.

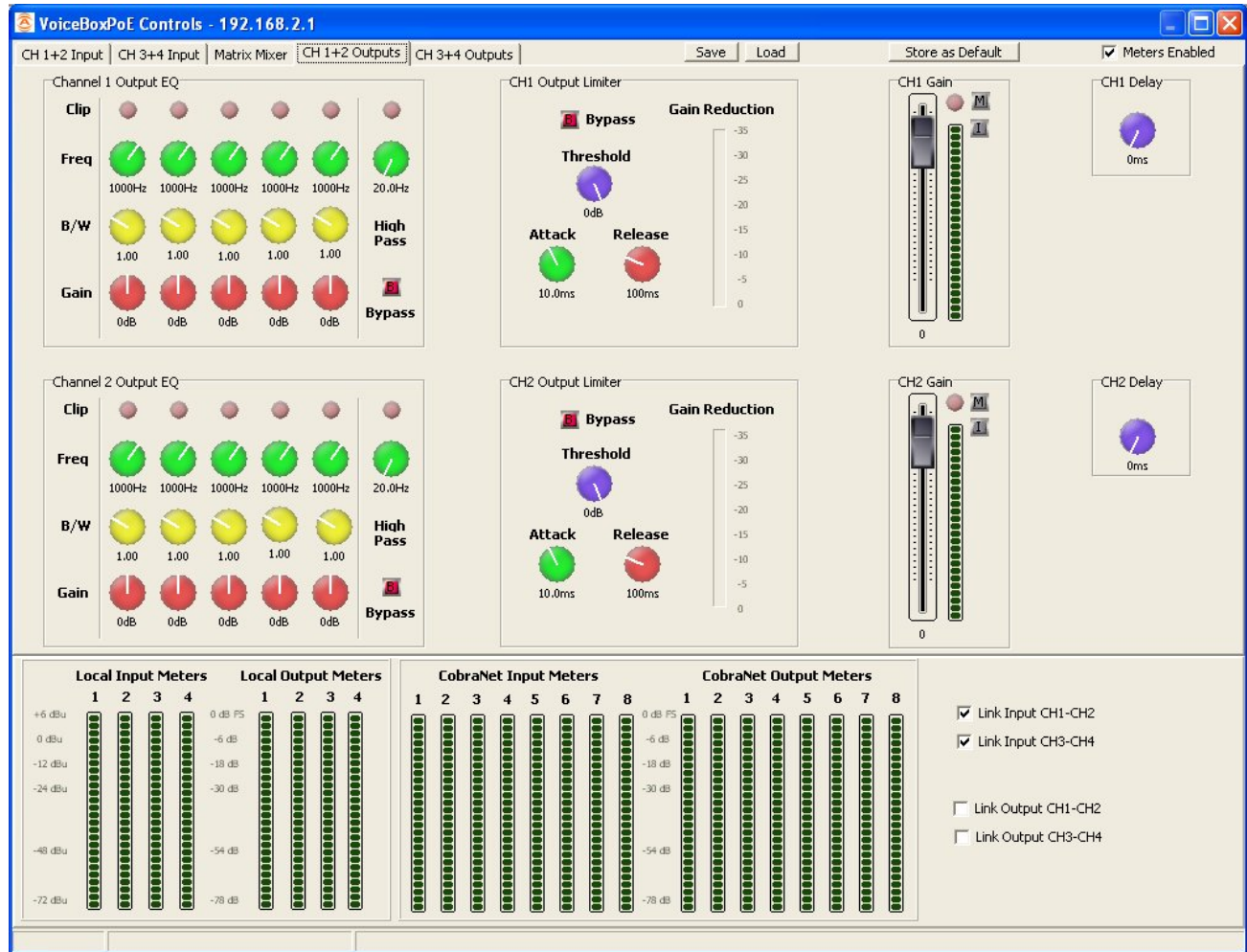


Figure 18 - Local output processing for Channels 1 and 2

3.4.1 – Output EQ

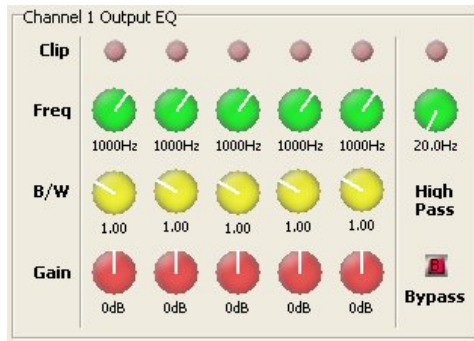


Figure 19 - Local Output EQ Controls

The output EQ stage has five cascaded bands of parametric EQ followed by a high pass filter. Each parametric EQ has adjustable frequency (20 Hz to 20 kHz), bandwidth (0.1 to 3.0) and gain (-12 dB to +12 dB). Each stage has its own clip LED to warn if the signal at the stage is clipping or not.

The high pass filter is a 12 dB / Octave filter with adjustable Frequency from 20 Hz to 20 kHz. This filter also has a clip indicator. It also has a bypass facility that, when active, will allow the audio to ignore this filter.

3.4.2 – Output Limiter

The output limiter has four controls are available for each output.

The bypass button allows the limiter to be completely bypassed and its settings ignored. Click on the button to toggle the state of the bypass. If the red indicator is on, the bypass is activated.

The remaining controls control the specific aspects of the limiter. The threshold control ranges from -100 dB to 0 dB, the attack control ranges from 1 ms to 1 s, and the release control that ranges from 10 ms to 30 s.

There is also a gain reduction meter to given an approximate indication of the amount of limiting that is actually taking place.

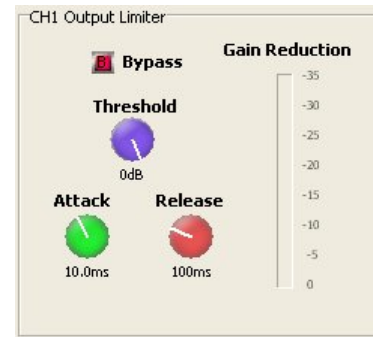


Figure 20 - Local Output Limiter Controls

3.4.3 – Output Gain

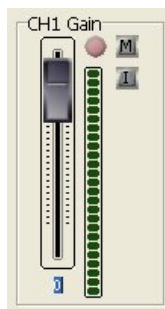


Figure 21 - Output Gain Controls

The Output Gain control allows the overall level of the signal to be adjusted before it is fed to the local outputs

The fader sets the gain from -100 dB to +12 dB. The meter alongside shows the current level after all the input processing has taken place, including the gain factor. There is also a clipping LED.

The gain control also has two toggle indicators. The upper one marked with an “M” is the “Mute” button. The lower button marked with an “I” is the “Polarity Invert” button.

3.4.4 – Output Delay

The delay control adds buffering to delay the audio signal. If the control is set to 0, the control is essentially bypassed.

A maximum delay of up to 20 ms can be added in total.

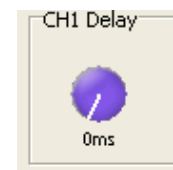


Figure 22 - Output Delay Control

3.5 – General Controls

3.5.1 – Stereo Linking

The local input channel controls have the option to be linked together to form stereo pairs. This same feature, called stereo linking, is also available for the local output channels also. Channels 1 & 2, may be linked and channels 3 & 4 may be linked. Both links may be active at one time. There is no provision for the settings in channels 3 and 4 to be linked in any way to inputs 1 & 2 and vice versa.

Linking the channels in this way will mean that any changes to one channel will be reflected in the other. When linking takes place, depending on which channels are linked, the parameters for channel 2 will change to match those set for channel 1 or the parameters for channel 4 will change to match those set for channel 3. This happens as soon as the link is enabled. From then on, all changes to one channel will update the other channel and vice versa.

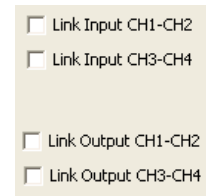


Figure 23 - Stereo Link Controls

3.5.2 – Store New Power-up Default

Although the settings of the controls are applied immediately to the controlled device, the values are not permanent and will be lost if the device is powered down. In order to ensure that the changes are retained, they must be committed to non-volatile memory. Clicking on the Store as Default button on the device setup form commits the current settings in the device to memory.

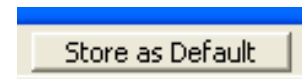


Figure 24 – Store as Default Button

Note: This operation takes a few seconds to complete and no other actions can be done until it is complete. The status bar will show the progress of operation and also a message once the operation is complete.

3.5.3 – Saving / Loading VoiceBox Configuration Presets

Once all the parameters have been set correctly, the complete configuration can be saved to a file so the settings may be easily transferred to another VoiceBox. Click the Save button at the top of the form (as shown in Figure 25) to initiate this. The Configuration Save dialog will appear allowing you to select a folder and a file name. Clicking on OK will save the configuration using the chosen filename in the chosen location. The filename will, by default, use a .DCF extension.

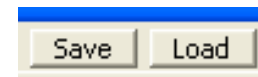


Figure 25 - Save/Load Buttons

Once the save is complete, the application will ask if the current device should have its configuration “saved as default” to ensure that it will power up with the parameters just saved. Click “Yes” to confirm the save or “No” to skip it. A final message will then appear to confirm the save is complete.

To load a previously saved configuration, click the Load button. The Load Configuration dialog will appear so the desired configuration file can be located. Locate and select the file to be loaded and click OK. A confirmation message will then be shown as a reminder that the current device configuration will be overwritten. Click “Yes” to continue.

Once correctly loaded, the application will ask if the new configuration just loaded to the device should be saved as the default. Click “Yes” to store the loaded parameters as the new defaults for power up. Click “No” to skip.

Note: The saved configuration is device specific so only VoiceBox configuration files can be loaded into a VoiceBox. The revision of VoiceBox firmware is also taken into account. A warning will be shown and the file will not be loaded if either the device type or the firmware revision does not match the device being set up.

3.5.4 – Metering

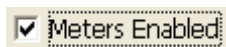


Figure 26 - Meters Enabled Checkbox

The VoiceBox interface can constantly monitor audio levels while the interface is open. Meters on the input tabs and on the Matrix Mixer tab show various audio levels whilst numerous clipping LEDs advise if the digital signal gets too large.

The metering function polls the VoiceBox numerous times a second to get constant updates for the meters and clip LEDs, but doing so creates an increased amount of network traffic. If this additional traffic becomes a problem or the metering is not required, it can be turned off.

Click the Meters Enabled check box to toggle the metering on or off.

Note: The meter and clip LED monitoring is only active whilst the VoiceBox interface is open. Metering starts every time the VoiceBox interface is opened and any active metering ceases as soon as the interface form is closed.

4 – Troubleshooting

4.1 – Checking the Network Connection

The CobraNet network connection can be diagnosed by noting the status of the LEDs just above the Ethernet connector on the VoiceBox. These LEDs will be on, off, or flashing depending on the current state of the network connection. Below is a table showing the states of the LEDs and what device status they represent.

LED status	Device Status
Left LED on permanently Right LED off	No Ethernet Connection
Left LED on permanently Right LED on but flashes off every 4 seconds	Ethernet connected but no devices found
Left LED on permanently Right LED flashing 3 times a second	Connected and CobraNet devices found Device is a performer
Left LED on flashing 3 times a second Right LED flashing 3 times a second	Connected and CobraNet devices found Device is the conductor

Table 1 - CobraNet Interface LED Status

4.2 – Common Faults

Below are some common faults. Accompanying each one is a list of the most common solutions. While the lists of solutions are not exhaustive, they represent the majority of the likely failures.

The VoiceBox does not show up in Control Center.

- If using PoE, check the VoiceBox is attached to a PoE-enabled port on a switch or the output port on a midspan injector and the power light on the switch/injector is on.
- If using the optional wall wart, Check power is supplied and the power light is on.
- Ensure the PC is running Windows 98SE or later (Windows 95 is not suitable).
- If running Vista, ensure a Vista compatible version of Control Center is running (1.1.0 onwards).
- Check the Ethernet cable is correctly connected.
- If connected to a switch ensure the link indicator is lit and activity flashes at least occasionally. If not, ensure the correct type of cable is used for the switch. Some switches require standard patch cables. Others can use standard or crossover cables.
- If connected directly to a PC, ensure the Ethernet cable is a crossover type.

5 – ARCHITECTS & ENGINEERS SPECIFICATION

The CobraNet interface unit shall provide four mic/line analog inputs on rear panel pluggable 3-pin barrier strips. Selectable gain of 0 dB, +30 dB, +50 dB, and +12 V phantom power shall be provided on rear panel DIP switches for each analog input channel. The internal analog to digital signal conversion shall be performed at 16, 20, or 24-bit resolution with a sampling frequency of 48 kHz. Eight channels of CobraNet inputs and eight channels of CobraNet outputs are supplied via the rear panel RJ-45 connector. Four channels of balanced line-level outputs shall be provided on the rear panel pluggable barrier strips. Interface power shall be provided by either an 802.3af compliant PoE network switch or a +12 V DC external power source.

The CobraNet interface shall provide DSP capabilities for independent channel gain, equalization and compression on all four local input channels. Each local output channel shall have DSP capabilities for equalization, output limiting, gain, and delay. The DSP shall also provide a fully configurable 13-input, 12-output matrix mixer with an input from a configurable sine wave generator.

The CobraNet Interface shall be compatible with Attero Tech Control Center software for flexible control and monitoring in system applications. The CobraNet Interface shall include a UL listed power source. The CobraNet Interface shall be compliant with the RoHS directive.

The CobraNet Interface shall be the Attero Tech VoiceBox 4 I/OP.

5.1 – Device Specifications

Audio Inputs	
Gain:	Selectable 0 dB, +30 dB, +50 dB
Input Type:	Balanced and RF filtered 3-pin pluggable
Input Impedance:	> 1.5K ohms, any gain
Maximum Input Levels:	+9 dBu @ 0 dB gain, -23 dBu @ +30 dB gain, -42 dBu @ +50 dB gain
Phantom Power:	+12 V, selectable per channel
Audio Outputs	
Output Type:	Balanced 3-pin pluggable with automatic muting upon loss of CobraNet signal
Output Impedance:	200 ohms balanced, 100 ohms unbalanced
Maximum output level:	+10 dBu
Audio Performance:	
EIN:	-128 dBu
Noise:	< -86 dBu @ 0 dB gain
System THD+N:	< 0.05% at any gain input signal 3 dB below max
Frequency Response	20 Hz - 20 kHz, +/- 1 dB
Sample Rate	48 kHz only

CobraNet Network	
Physical Level:	Standard Ethernet
Connector:	Single RJ-45
Cable Quality:	CAT-5
Transmission Speed:	100 Mbps
RS-232 Port	
Physical Levels:	Standard RS-232
Connector:	3-Pin pluggable Phoenix style
Power Requirements	Class 2 802.3af PoE PD Compliant or Optional 12 V DC @ 1A
Power Consumption	11 Watts maximum
Dimensions	1.5" H x 7.25" W x 6.5" D
Weight	27 oz, 760 g
Compliance	RoHS, UL Listed power source, FCC Part 15 Class A Compliant

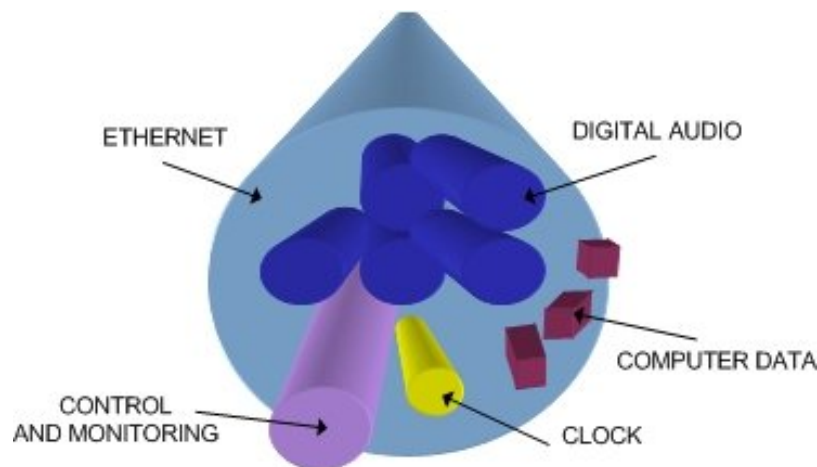
APPENDIX A – Introduction to CobraNet

CobraNet is an audio networking technology for delivery and distribution of real-time, high quality, uncompressed digital audio using a standard Ethernet network. It is implemented using a combination of hardware, firmware, and the CobraNet protocol.

Unlike other audio networking or distribution technologies, CobraNet is a true network and exists on standard Ethernet networks using standard Ethernet hardware. Since it is a true network, audio routing is highly flexible between network nodes and can be used in a variety of audio distribution applications.

In addition to the high degree of routing flexibility that CobraNet provides, the technology also incorporates the ability to monitor and control CobraNet devices remotely. This is a key feature that is highly important in fixed installation applications where the audio distribution equipment may not be readily accessible. All CobraNet devices on the network can be controlled and monitored from a central location by sending control commands and monitoring device specific parameters.

CobraNet provides this capability by implementing Simple Network Management Protocol (SNMP). SNMP is a standard protocol typically used for monitoring network devices such as Ethernet switches. In the case of CobraNet, it allows users to communicate with any CobraNet device using standard SNMP tools or a customized user interface designed specifically for CobraNet, such as Attero Tech’s Control Center application.



The figure above represents the types of data that coexist on a CobraNet network.

Before a CobraNet system can be configured, it is important to first understand how CobraNet distributes audio between devices.

Audio is sent in "bundles" on a CobraNet system. Each bundle is capable of holding up to 8 logical audio channels. Every CobraNet device has a number of bundle transmitters and bundle receivers. These transmitters and receivers are the mechanism used to send and receive bundles between devices.

For a transmitted bundle, audio may be sourced either directly from the local audio inputs of the device or from internal audio via the on-board DSP¹, but not both simultaneously. Internal audio from the onboard DSP could have originally been sourced from the local device inputs, sent from another CobraNet device or even generated by the DSP itself. Combinations of the local or internal audio may exist within a bundle in any order. Additionally, a single audio source in a device may be used multiple times in a single transmitter bundle or across multiple transmitter bundles.

For a received bundle, the received network audio may be routed directly to the device’s local outputs, the internal DSP¹ or simply ignored.

Once the contents of a bundle have been decided, the next step is to pass the bundle to another CobraNet device. To do this, every CobraNet device has up to 4 bundle transmitters. Each bundle transmitter has a transmit mode that must first be selected. This affects how many devices may receive that particular bundle at a time.

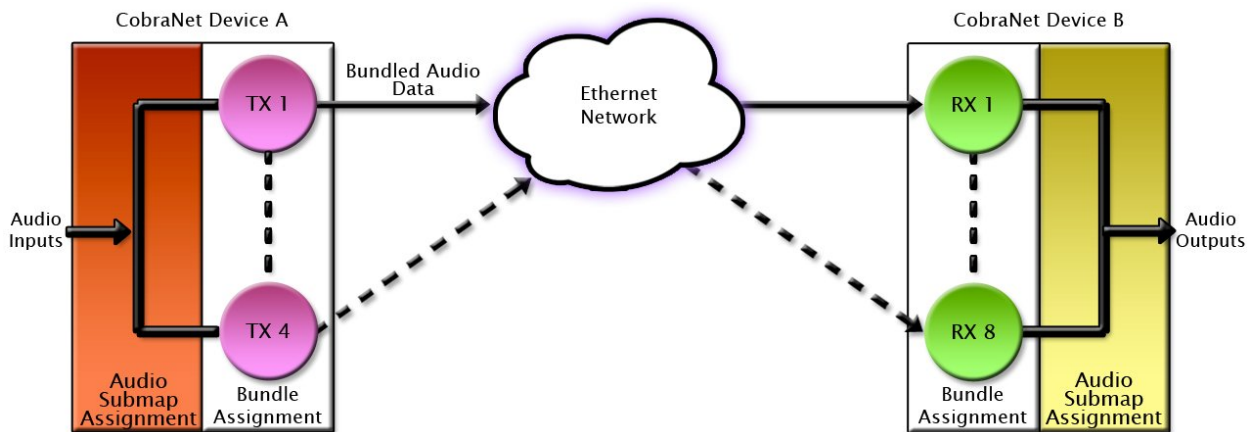
¹ Not available on all devices – CS496xxx devices only

The modes are as follows:

- Unicast – Used for one-to-one connections. In this mode, only one receiver at a time can receive this bundle. Once a link is established from this transmitted bundle to a receiver, any future requests for that bundle from other potential receivers will fail.
- Multicast – Used for one-to-many connections. This mode broadcasts its contents over the entire network. There is no restriction on the number of receivers. However, the downside is that CobraNet packets are distributed to all nodes on the network, whether they need them or not thus creating possible network bandwidth issues.
- Multi-unicasts – Another one-to-many mode. Whilst this is the most efficient method for getting a bundle to multiple receivers in terms of network bandwidth, it requires more processing power on the CobraNet device so in this mode there is a maximum limit of four receiver connections (this can be reduced if required). If more connections are required than the limit, the node can be configured to automatically switches to multicast.

Note: When a bundle must be transmitted to multiple receivers, multi-unicast transmissions should be used where possible.

Once the mode is selected, to enable a device to transmit the bundle, simply allocate the particular transmitter bundle a non-zero number. Since this number identifies all the network packets sent out by that transmitter, each transmit bundle number must be unique on a network².



Now that the transmitter is set up, it is time to set up the receivers. In order to receive bundles, each CobraNet device has up to eight bundle receivers. To enable a device to receive a bundle, simply allocate one of that device's bundle receivers the same bundle number as a transmitted bundle. By doing so, a virtual link is created and audio should now be passed from one device to the other. It should be noted that no knowledge of a device's network topology or connection is thus required in order to configure audio connections. The only restriction to this is that a device cannot be set up to receive a bundle it is also transmitting.

The above case creates a simple, one-to-one, unidirectional link. If more devices are required to receive that bundle, allocate the same transmitted bundle number to a bundle receiver on the other CobraNet devices.

It is also important to note that CobraNet supports simultaneous bidirectional audio distribution in each device. Not only could audio be sent from Device A to Device B but at the same time, should it be needed, audio could also be sent from Device B to Device A. The exact bundle and routing configuration will be determined by the needs of each individual installation. An installation may have multiple units transmitting multiple bundles. The only restriction is the bandwidth available on the network to transfer the audio.

CobraNet does more than just transfer audio data. It can be used to pass serial information as well. A feature called serial bridging has been incorporated that allows the passage of serial data between nodes. Each node can pass serial data to a specific node or multicast the data to multiple nodes. A node can also receive data from either a single source or multiple sources. Baud rates, data bits, stop bits, parity, and so on are all configurable. There is also support for multi-drop serial buses as well.

Finally, CobraNet has the capability to alter all of the above options in real time making the whole system completely dynamic. By use of control software, all of the bundle assignment parameters can be configured with no need to change cables, switch out connectors, or pull new wiring. Most importantly, this control capability can be implemented from a single location!

² Bundle numbers range from 1 through 65535. A value of 0 represents an inactive bundle. Numbers 1-255 are reserved for multicast mode transmissions only.

APPENDIX B- Reference Documents

The following table lists the relevant reference documents.

Document Title	Revision
CobraNet Programmer's Reference (Cirrus Logic)	v2.5