VP-8IP

DIGITAL MULTIMODE AUDIO PROCESSOR

TECHNICAL MANUAL

Wheatstone Corporation

600 Industrial Drive, New Bern, North Carolina, USA 28562

VORSIS ULTRA-HIGH RESOLUTION PROCESSING

VP-8 IP Digital Multimode Audio Processor Technical Manual - *VP8IP GUI version 4.X.X.exe*

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Attention!

Federal Communications Commission (FCC) Compliance Notice:
Radio Frequency Notice

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. 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.

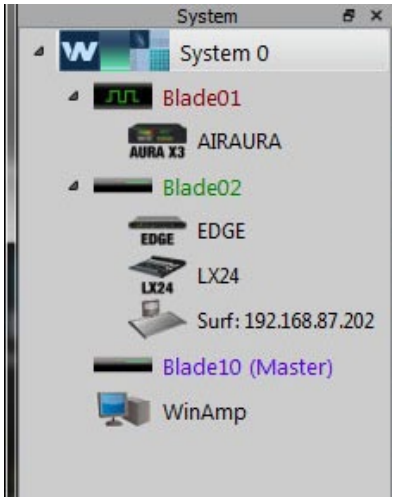
This is a Class A product. In a domestic environment, this product may cause radio interference, in which case, the user may be required to take appropriate measures.

This equipment must be installed and wired properly in order to assure compliance with FCC regulations.

Caution! ***Any modifications not expressly approved in writing by Wheatstone could void the user's authority to operate this equipment.***

Adding VP-8IP To The Peripheral Devices* Tab

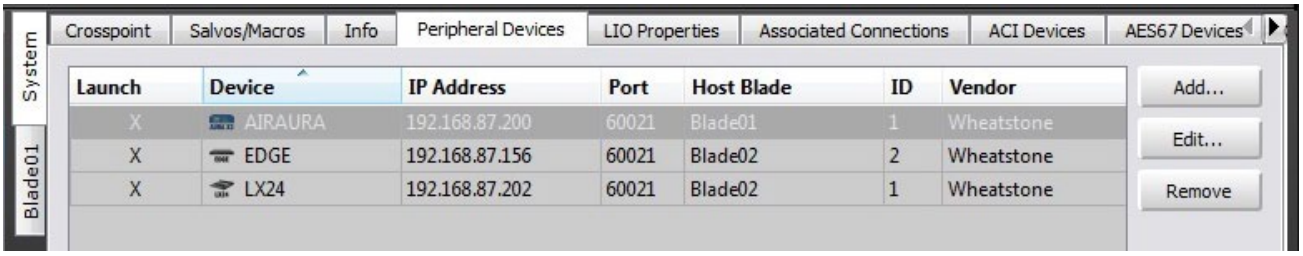
In order to utilize all the features of the VP-8IP Digital Audio Processor the device must be added to the System *Peripheral Devices* tab in the Wheatstone WheatNet-IP Navigator program (aka the Navigator GUI). This sheet shows you the basics of that procedure. Refer to the *WheatNet-IP BLADE3 Audio Over IP Network Technical Manual* for additional details.



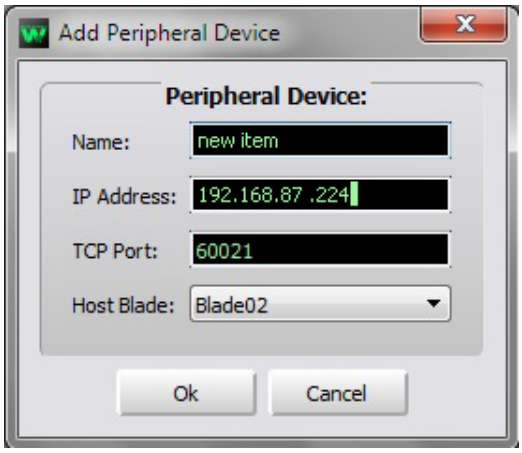
You will need to know the IP address of the device being added, so you will want to find that out before you start.

Launch the Navigator GUI and make sure that **System 0** is selected in the *System* pane. You will see something like this:

Now select the *Peripheral Devices* tab.



Click the *Add* button to bring up the *Add Peripheral Device* dialog:



Type in a convenient *Name* and insert the *IP Address* of the device being added. Leave the *TCP Port* at the default setting of **60021**. From the *Host BLADE* drop down select the BLADE that you want to associate the Peripheral device with. Click *Ok*.

This completes the process of adding the device to the *Peripheral Devices* tab. The added device should show up in the *System* pane under the BLADE you added it to. If it does not show up, or if it shows up but has a yellow question mark on it, then there is either a network issue that needs attention, or the device is not connected to the network at all, or one or more steps have been omitted or done incorrectly in the configuration process.

* In previous Navigator GUI versions this was the **System 3rd Party Devices** tab.

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Wheatstone Corporation

VORSIS ULTRA-HIGH RESOLUTION PROCESSING

VP-8IP Digital Audio Processor

General Information

Introduction

The Wheatstone VP-8IP takes a completely new approach to broadcast audio processing through new and surgically accurate audio processing algorithms created by Wheatstone's Vorsis product design team. The result is far more audio processing power as well as complete user control over that power.

Our new "Audio Processing Guru®" software is included with each VP-8IP and its six simple controls intuitively and intelligently adjust the over 200 controls that are available in the full control "Pro" version of the GUI. No PhD in audio processing is required when using the Guru – just move a few sliders until you achieve your perfect sound.

Speaking of a Pro GUI version, we make a full control GUI available that is intended for the audio processing experts. In that version there are no hidden controls and no secret "back door" to hide things from the end user. Every control is available and labeled for exactly what they do so prior experience with audio processing adjustment is definitely recommended. Obtaining the full control Pro GUI is simple and it's free. Simply register for it at our website http://wheatstone-processing.com.

Regardless of which GUI version you choose to use, if you can think of the on-air sound you want your station to have, the VP-8IP can create it for you. This extreme flexibility is made possible by our entirely new approach to the user interface and clever new DSP audio processing algorithms. The VP-8IP has clarity, power, and accuracy of sound not found in any other broadcast audio processor in its price range.

The VP-8IP can be remotely controlled from anywhere via a TCP/IP network connection and Windows-based GUI (Graphical User Interface) software applications.

Wheatstone Corporate Headquarters

New Bern, North Carolina, USA

Wheatstone Corporation is a world-leader in broadcast audio equipment design and manufacturing. Our products include audio consoles for the radio and television industries, audio control surfaces and digital audio networks, and our award winning digital audio processors.

Originally founded in 1975 as the Audioarts Company, Wheatstone is known worldwide for broadcast audio equipment of the highest precision. Every facet of our products from concept to customer support is accomplished at our state-of-the-art facility in New Bern, North Carolina, USA. To ensure that Wheatstone products always meet our extremely demanding quality standards we never rely on offshore or contract manufacturing.

As audio professionals and audio perfectionists we believe that our customers should have not only the right, but also the power, to shape their OWN sound – one with a completely unique sonic signature that's theirs, not the one created by a processor manufacturer. To get there, it required our team to take a completely fresh approach to audio processor design. To achieve that goal we developed Vorsis Ultra-High Resolution Processing technology to provide a completely clean, high quality "draw you in" sound that even the best ears in the business love to hear.

Wheatstone's proven and award winning Vorsis Ultra-High Resolution Processing technology is embedded in every audio processor we make, allowing you to create the cleanest possible sound – as loud as you want it – if ultimate loudness and great sound is your goal.

Overview

The VP-8IP is the industry's first true multi-mode audio processor, having six built-in operating modes for: FM, AM, FM-HD, AM-HD, MP3/AAC>48k, and MP3/AAC<48k. In each mode, special processing algorithms designed by Vorsis condition the audio appropriately for the selected transport medium.

Audio is first preprocessed by a phase-linear four-band AGC, then by an eight band limiter and final limiter or clipper. The eight band and final limiter section operation is entirely different in each of the operating modes due to special signal conditioning algorithms that have been specifically designed and tuned for the particular needs of the operating mode. Changing operating modes is as simple as selecting a new mode from within the PC-based Windows Graphical User Interface; no reboot of the unit is required.

Feature Highlights

- Selectable phase rotator for making voice energy more symmetrical.
- Adjustable input high-pass filter may operate in Stereo or M/S modes.
- Separate audio input gains for analog and digital inputs.
- Four band parametric EQ may be placed before or after multiband section.
- Four band linear phase crossover with adjustable crossover points.
- Exclusive Wheatstone Multiband Dynamics Processing.
- Precision eight-band multiband limiter.
- Specialized codec preconditioning tailored for each of the four CODEC modes.
- Advanced Distortion Masked Clipper for absolute FM and AM peak control.
- Reference-grade stereo encoder available in FM mode.
- Adjustable positive asymmetry, LF tilt correction, and dual transmitter outputs in AM mode.
- 4.5kHz, 5kHz, 5.5kHz, 6kHz CCIR, and 6.5kHz, 7.5kHz, 10kHz NRSC low-pass filters supported in AM mode.

At the heart of the VP-8IP is a specialized DSP farm containing both floating and fixed-point processors. Advanced signal routing capability allows restructuring of the processing chain as desired and on-the-fly. Confidence monitoring via headphones can be patched to any processing section at will. The flexibility of the headphone monitoring also allows a backup audio source that is attached to the VP-8IP but not even on the air to be monitored!

The most salient feature of the VP-8IP is its eight band final processing section, which is completely reconfigured for each of the six possible operating modes. This innovative design allows for far more flexible peak control and audio tailoring for each of the specialized six modes than the antiquated limiters having fewer bands found in other audio processors on the market today.

The eight-band limiters are equipped with fully adjustable thresholds and attack and release times for each of the bands. Because the VP-8IP limiters operate with precision, time-aligned bandpass filters, its dynamic operation is nearly invisible to the ear. The resulting on-air sound can be carefully tailored to create exactly that desired. The eight bands of peak limiting utilized in the VP-8IP allows tuning flexibility previously unheard of in any broadcast audio processor. The eight-band limiter section is followed by its own four-band parametric equalization stage that can be used to further fine tune the sound of each signal path.

The FM signal path is equipped with an adjustable diversity delay of up to ten seconds, corresponding to the requirements of the North American (iBiquity®) HD Radio system. A reference-grade stereo encoder is also available in FM mode to provide the highest possible quality on-air signal.

Rack Mounting

The VP-8IP is designed to fit into an industry standard 19" equipment rack, and requires one rack unit (1.75 inches) of vertical space. The VP-8IP has a depth of 14" behind the rack rails (including chassis connectors). An additional five inches of space is required for wiring cables to pass through. The chassis has a width of 17-3/8". Space needed in front of rack rails is 3/4". Ideally, four screws should be used to mount the unit. If only two screws are being used they must be used in the bottom holes in order to provide proper support.

The VP-8IP does not have top or bottom cover vent holes. Latent heat is vented out of the enclosure by natural convection through slots in the top of the rear panel. Cooler air is drawn into the unit through vertical slots positioned lower in the side panels. There is no fan inside the VP-8IP because its power consumption is low enough to not require one.

The VP-8IP may be mounted between other devices in the equipment rack and in accordance with good engineering practice should not be mounted directly above devices that generate significant amounts of heat. If such a location is unavoidable then it is advisable to utilize an extra 1RU blank rack panel between the VP-8IP and devices immediately above and/or below it.

WARNING! **Under no circumstances should the VP-8IP unit be opened! The unit contains high voltage circuits that are hazardous and potentially harmful. The unit has no user-serviceable parts inside! If you have a problem the unit must be returned to Wheatstone Corporation for repair.**

Installation Tips:

- Place any surge protection circuits as close as possible to the VP-8IP or other device being protected.
- Establish a low impedance common ground in your facility and try to route all grounds to that point.
- Choose the best power conditioning / UPS units that you can afford and suitable for your equipment – focus on the features and options you need. The better UPS products can prevent thousands of dollars in equipment damage – some even come with an external equipment damage warranty.

- *Unbalanced* audio connections to the VP-8IP should be made with shielded two conductor cable such as Belden 8451 or 9451 as if connecting a balanced source. At the unbalanced source's output connect the + Output to the HI input wire and connect the source GND wire to the LO wire. Connect the shield at the VP-8IP end only.
- For digital audio connections always use a good quality digital audio cable with a characteristic impedance of 110 ohms.

The AES/EBU specification, with its broad impedance tolerance, allows for cables with impedances from 88 ohms to 132 ohms – 110 ohms is ideal. Twisted pair cable should be shielded, and in the case of multi-pair cable, each pair should be individually shielded. Foil shielding is recommended for permanent installations and foil shield plus overall braid should be used in applications where frequent flexing of the cable will occur. One cable pair is capable of carrying two channels of digital audio.

Generic "audio" cable such as Belden 8451 *may* be used for interconnecting AES3 digital audio devices but **only** for distances of less than about 25 feet. The actual cable length that will work satisfactorily in an installation is primarily determined by the error correction and jitter tolerance of the AES3 receiver device and the cable used.

The impedance of most "analog" cables ranges from 40 ohms to 70 ohms and represents a large impedance mismatch from the nominal 110 ohms required in the AES3 standard. Such mismatch will result in signal reflections causing bit errors at the AES3 receiver. The higher capacitance of generic analog cables also slows down the rise time of the digital data signals, impairing the ability of the AES3 receiver to accurately detect digital signal transitions. This may result in increased jitter.

Where to Install the VP-8IP

The recommended location for the VP-8IP is at the transmitter site when it is in its FM or AM processing mode. A transmitter site installation will always ensure a more faithfully modulated signal on the air because there is better control of intervening devices passing the peak-controlled output of the VP-8IP.

Additional notes regarding FM and AM mode when the unit cannot be installed at the transmitter site:

Analog STL

Older analog STL's can suffer from an inability to control high frequency audio peaks because of inadequate bandwidth in their IF circuits, or inadequate low frequency performance which will result in tilt. Also, some STL designs suffer from bounce in their AFC loops when handling processed low frequency material, which can rob modulation capability.

Analog Phone Lines:

We cannot recommend any form of discrete (left/right) analog "phone line" type STL because of the inability of many service providers to guarantee flat frequency response and proper phase matching between circuits. Furthermore, in many countries analog circuits such as these are being discontinued or the cost has been raised to prohibitive points.

Digital STL:

If using a Digital STL that employs audio compression, then it is highly recommended that the VP-8IP be placed at the transmitter site. This is primarily because the encoding schemes used in such STL's will not accurately pass the well-defined peak levels created by the VP-8IP.

The VP-8IP can be placed at the studio end of a compressed STL with at least two caveats:

- You will not be able to use the composite clipper in the VP-8IP when it is operating in FM mode. While most digital exciters offer a composite clipper function, they are typically quite crude in their operation by comparison to what is inside the VP-8IP and are *never* the optimum choice when sound *quality* is important.
- Compressed (data reduced) STL's do not perform well when competitively processed audio is presented to their inputs. This is primarily because when dense audio is presented to these codecs they have fewer "opportunities" to remove redundant audio information and mask that removal. Therefore, codec operation may be much more obvious – potentially more so than what might occur with the processing located *after* the codec where the masked artifacts are simply unmasked by the increased "gain" due to processing.

Energizing

Assuming the VP-8IP Digital Audio Processor is correctly rackmounted, you may now energize it. There is no power switch. The AC line input voltage is permitted to be between 90 and 260 VAC, 50 or 60Hz. Power consumption is under 100VA.

Aggressive AC input filtering is utilized at the AC input of the VP-8IP; however it is always advisable to use external surge protection and/or an uninterruptible power supply (UPS), especially where AC power quality is questionable, such as at a remote transmitter site.

Power conditioning, surge suppression, and even power backup devices are wise investments when using sensitive modern electronic devices that use an internal computer.

Use of a UPS (uninterruptible power supply) is a good idea and will protect the VP-8IP from short duration power interruptions which may cause it to reboot. During boot up, audio is interrupted for approximately 20 seconds.

I/O Connections

All audio input and output, control, Ethernet, and power supply connections are made via various connectors mounted on the VP-8IP's rear panel.

Six XLR connectors are provided for analog and digital audio input and output connections.

Four BNC connectors are provided for SCA in and TX (transmitter) out connections.

An RJ-45 connector is provided for Ethernet connection, which can be used to connect a Windows® PC running the VP-8IP GUI.

The two DB-9 connectors provide eight GPI inputs for the first eight presets and four GPO outputs.

The pinout drawings on pages 1-15 through 1-18 summarize all wiring connections.

Audio Inputs

The VP-8IP accepts three types of audio input sources:

- Balanced analog line level left/right audio;
- Digital AES3 compliant left/right audio with sample rates between 32kHz and 96kHz;
- Optional WheatNet-IP via 100BaseT Ethernet connection to a WheatNet-IP audio network.

Input audio can be applied to all inputs simultaneously.

Automatic audio failover from analog to digital or vice versa is supported.

Automatic failover from AES3 or WheatNet-IP to analog is instantaneous and based on invalid or missing bits in the AES3 or WheatNet-IP stream, or after 30 seconds of "silence" (level below -48dBFS).

Automatic failover from analog to AES3 is based on silence sense responding to audio on both channels being below -48dBFS for more than 30 seconds.

Failover from AES3 to WheatNet-IP (or vice versa) is not available.

Analog In – XLR-F

Pin 1 XLR LT SH – LINE LT IN SH
Pin 2 XLR LT HI – LINE LT IN HI
Pin 3 XLR LT LO – LINE LT IN LO

Pin 1 XLR RT SH – LINE RT IN SH
Pin 2 XLR RT HI – LINE RT IN HI
Pin 3 XLR RT LO – LINE RT IN LO

AES In – XLR-F

Pin 1 XLR SH – AES IN SH
Pin 2 XLR HI – AES IN HI
Pin 3 XLR LO – AES IN LO

SCA In – BNC

Pin 1 BNC 1 HI – SCA 1 IN HI
Pin 2 BNC 1 SH – SCA 1 IN SH

Pin 1 BNC 2 HI – SCA 2 IN HI
Pin 2 BNC 2 SH – SCA 2 IN SH

Audio Outputs

Output audio is available as:

- Analog balanced left/right stereo.
- Analog balanced left/right stereo and de-emphasized according to any pre-emphasis used.
- AES3 digital, either pre or post diversity delay, and/or de-emphasized according to any pre-emphasis used.
- Balanced line level composite stereo.
- Composite stereo on two rear panel BNC female connectors.
- WheatNet-IP audio network.

Analog Out – XLR-M

Pin 1 XLR LT SH – LINE LT OUT SH
Pin 2 XLR LT HI – LINE LT OUT HI
Pin 3 XLR LT LO – LINE LT OUT LO

Pin 1 XLR RT SH – LINE RT OUT SH
Pin 2 XLR RT HI – LINE RT OUT HI
Pin 3 XLR RT LO – LINE RT OUT LO

AES / baseband192 Out – XLR-M

Pin 1 XLR SH – AES / BASEBAND192 OUT SH
Pin 2 XLR HI – AES / BASEBAND192 OUT HI
Pin 3 XLR LO – AES / BASEBAND192 OUT LO

TX Out – BNC

Pin 1 BNC 1 HI – TX 1 OUT HI
Pin 2 BNC 1 SH – TX 1 OUT SH

Pin 1 BNC 2 HI – TX 2 OUT HI
Pin 2 BNC 2 SH – TX 2 OUT SH

Headphones

A high-power, overload protected headphone output is available on the front panel for local monitoring. The headphone source can be selected in software to be from one of six important signal points inside the processing algorithm, including both analog and AES inputs even if those inputs are not selected to feed the audio processing chain.

Network Connection

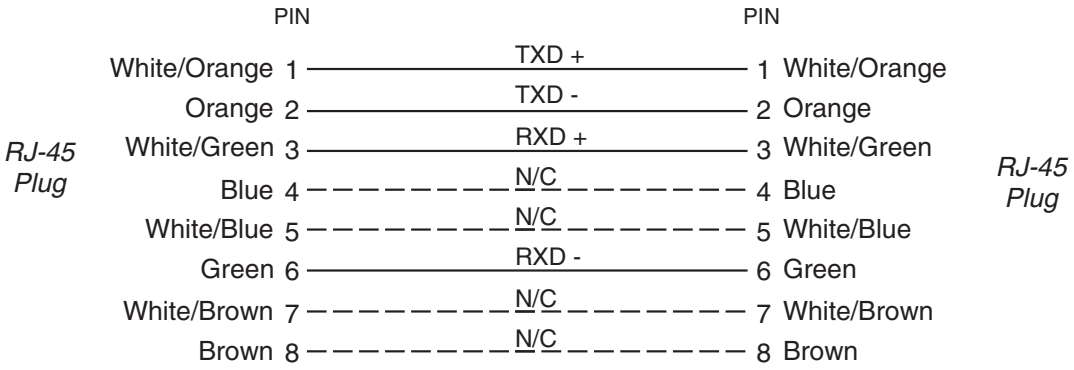
The VP-8IP contains its own internal 100Base-T Ethernet switch with one port available on the outside of the product reserved for user connections. The Ethernet switch is equipped with auto-sensing and therefore this port will accept either a straight-through or crossover cable.

Networked systems are normally connected to the VP-8IP via "straight" (pin to pin) CAT5 cable. For typical CAT5 cable pinouts see below. These connections are for communicating with the configuration computer via a network. If you are connecting directly between the computer and the VP-8IP with no network in between, it is normal to use a crossover cable.

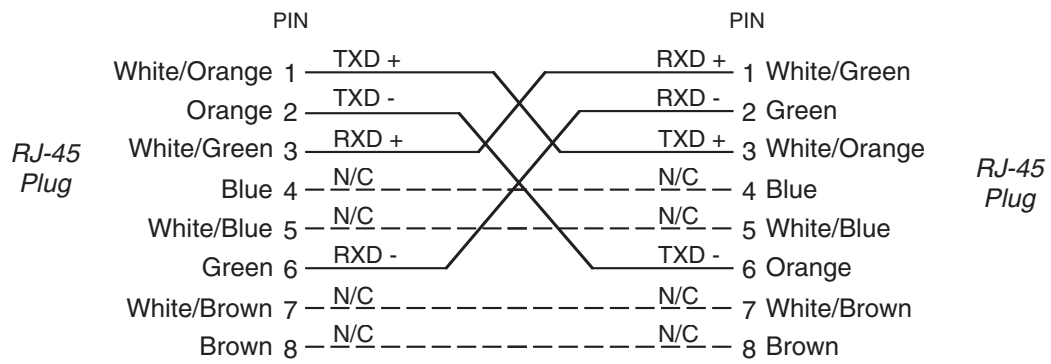
Ethernet – RJ-45

Pin 1 – TXD +
Pin 2 – TXD -
Pin 3 – RXD +
Pin 4 – N/C
Pin 5 – N/C
Pin 6 – RXD -
Pin 7 – N/C
Pin 8 – N/C

TYPICAL ETHERNET CABLE



TYPICAL CROSSOVER CABL E



General Purpose Interface (GPI/GPO)

The VP-8IP is equipped with eight General Purpose Input (GPI) control input ports on one female DB-9 connector, and four General Purpose Output (GPO) tally outputs on a second female DB-9 connector. All GPI and GPO connections are optically-isolated to prevent external ground loops and prevent dangerous voltages from being introduced into the VP-8IP.

GPI

The GPI interface provides eight separate General Purpose Inputs, sharing a common return. The connector is a female DB-9 connector with the following pin assignments. Note that all pins are isolated from VP-8IP internal circuitry!

Pin 1 – GPI COM (-)
Pin 6 – GPI 1 IN (+)
Pin 2 – GPI 2 IN (+)
Pin 7 – GPI 3 IN (+)
Pin 3 – GPI 4 IN (+)
Pin 8 – GPI 5 IN (+)
Pin 4 – GPI 6 IN (+)
Pin 9 – GPI 7 IN (+)
Pin 5 – GPI 8 IN (+)

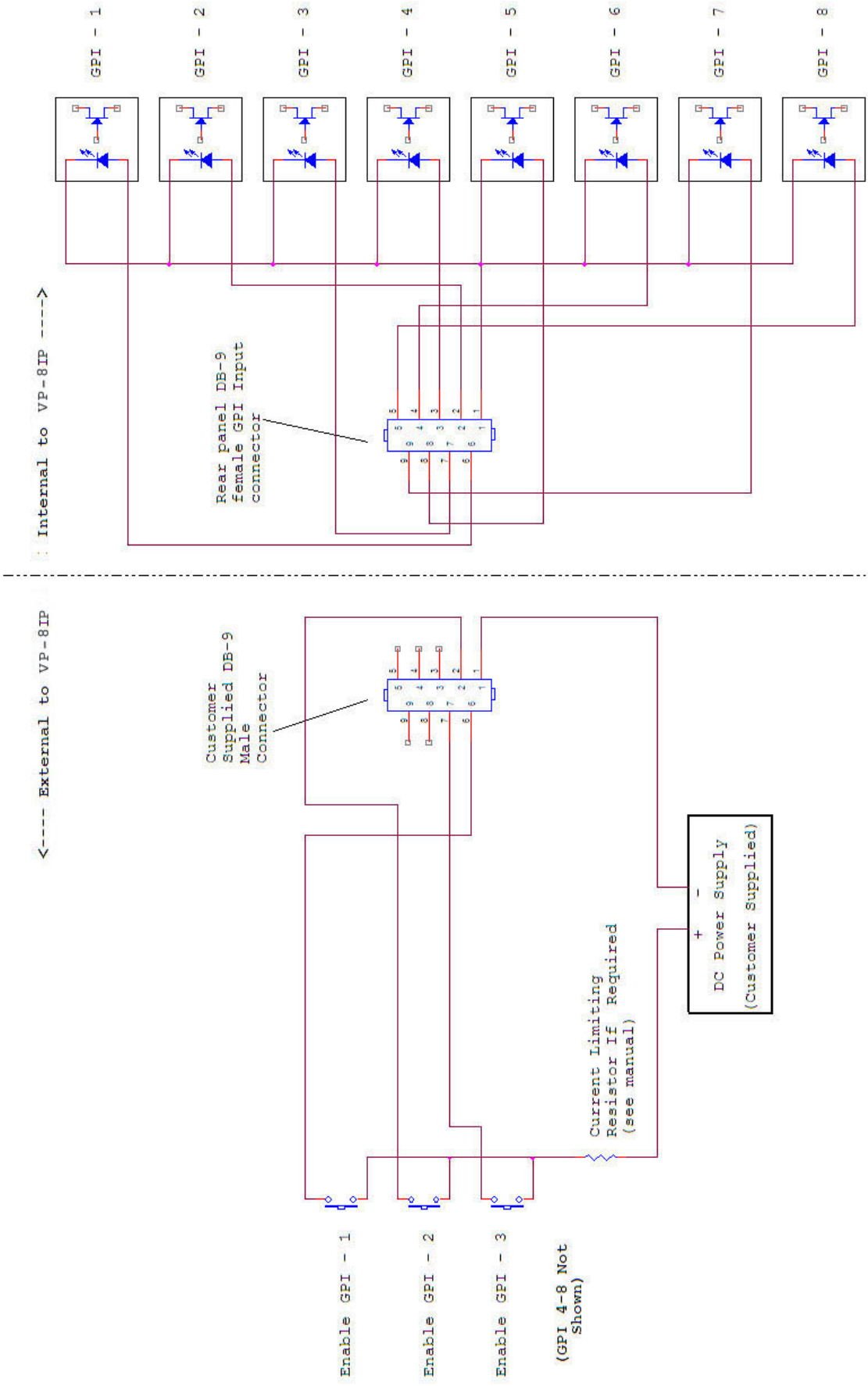
The GPI input circuits are optoisolated by devices with a 1,500VAC isolation rating.

Each of the GPI inputs is current limited by a 475 ohm resistance. The LED device inside the optoisolators is rated at a maximum forward current of 50mA with a *normal* operating current of 5mA; therefore it is necessary to determine if an additional outboard series resistance will be required in your particular application.

The following table lists the value of external resistance required when the external voltage applied to the GPI exceeds 3.3VDC. In all applications up to and including 48VDC, a one-half watt resistor is sufficient.

Applied Voltage to GPI	External Resistance
3.3 VDC	None
5.0 VDC	330 Ohms
6.0 VDC	470 Ohms
7.5 VDC	820 Ohms
10 VDC	1.3 kOhms
12 VDC	1.8 kOhms
15 VDC	2.4 kOhms
24 VDC	3.9 kOhms
30 VDC	5.1 kOhms
48 VDC	9.1 kOhms

Example Schematic of General Purpose Inputs



User Presets

The VP-8IP comes with approximately 50 factory presets and can hold a total of 80 in its onboard memory. You can design and save additional presets within the VP-8IP's onboard memory and can store an unlimited number of presets on the PC that is hosting the remote control GUI software. The factory pre-sets can be readjusted and then saved to new names in order to create your own unique and individual sound.

Note: The factory presets are write-protected and changes made to them cannot be written back to the same memory location. Factory presets whose control settings have been modified are considered "user" presets and therefore must be saved as a new name and to a new preset number. Please start with a factory preset that has the on air sound closest to what you need, and then make any necessary changes to that preset to get to your final sound.

NOTE: User presets stored in locations 1 through 8 are logically assigned to the remote GPI function! Two examples of presets that might be stored here are:

- A preset that has all processing turned off and has special input/output level calibrations (Bypass or EAS Preset).
- A preset that has all processing enabled but has the Stereo Pilot turned off (Mono Preset).

Preset storage is typically as follows:

- Presets 1 - 8 are user presets that can be quickly selected from the GPI connector.
- Factory presets are installed starting at Preset 9, with Preset 9 being a "bypass" preset. The factory presets cannot be written over or deleted.
- User presets, other than the GPI selectable presets 1 - 8, are stored above the highest factory preset. The number of available slots for user presets depends on how many factory presets were installed.
- User presets may be locked by the user (see Chapter 3). Factory presets, on the other hand, are locked at the factory. Whereas a user can unlock user-locked presets, the same user can not unlock factory presets.

GPO

The VP-8IP provides four General Purpose Output (GPO) circuits that may be used in a variety of ways. Like the GPI circuits, the GPO utilizes optoisolators in order to provide high electrical isolation from the outside world.

Unlike the GPI circuits, though, each of the GPO's has completely isolated return circuits – that is, they do not share a common return. Because of this each GPO may be used in any way desired without concern of cross coupling between unrelated external circuits.

The optoisolators on the GPO are rated at a maximum of 350 Volts AC or DC and with a maximum load current of 100mA. Because of the AC rating, external circuit polarity is unimportant and therefore we will simply define the GPO pins as "Source" and "Return."

The pin assignment for the rear panel DB-9 GPO connector is as follows:

Pin 1 – N/C

Pin 2 – GPO 1 Return
Pin 6 – GPO 1 Source

Pin 3 – GPO 2 Return
Pin 7 – GPO 2 Source

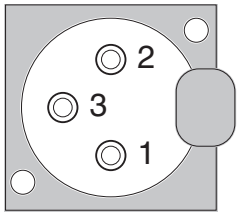
Pin 4 – GPO 3 Return
Pin 8 – GPO 3 Source

Pin 5 – GPO 4 Return
Pin 9 – GPO 4 Source

Analog XLR Connections

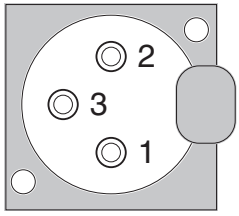
Analog In - XLR-F

XLR LT-F



PIN 1 XLR LT SH - LINE LT IN SH
PIN 2 XLR LT HI - LINE LT IN HI
PIN 3 XLR LT LO - LINE LT IN LO

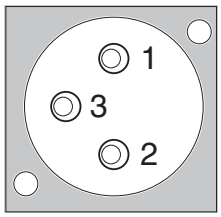
XLR RT-F



PIN 1 XLR RT SH - LINE RT IN SH
PIN 2 XLR RT HI - LINE RT IN HI
PIN 3 XLR RT LO - LINE RT IN LO

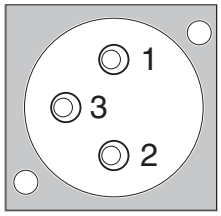
Analog Out - XLR-M

XLR LT-M



PIN 1 XLR LT SH - LINE LT OUT SH
PIN 2 XLR LT HI - LINE LT OUT HI
PIN 3 XLR LT LO - LINE LT OUT LO

XLR RT-M

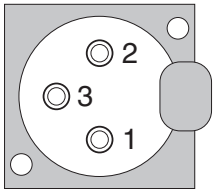


PIN 1 XLR RT SH - LINE RT OUT SH
PIN 2 XLR RT HI - LINE RT OUT HI
PIN 3 XLR RT LO - LINE RT OUT LO

Digital XLR Connections

AES In - XLR-F

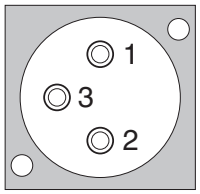
XLR-F



PIN 1 XLR SH - AES IN SH
PIN 2 XLR HI - AES IN HI
PIN 3 XLR LO - AES IN LO

AES Out / baseband192 - XLR-M

XLR-M



PIN 1 XLR SH - AES / BASEBAND192 OUT SH
PIN 2 XLR HI - AES / BASEBAND192 OUT HI
PIN 3 XLR LO - AES / BASEBAND192 OUT LO

BNC Connections

BNC - SCA

1

PIN 1 BNC 1 HI - SCA 1 IN HI - CENTER PIN
PIN 2 BNC 1 SH - SCA 1 IN SH - SHELL

2

PIN 1 BNC 2 HI - SCA 2 IN HI - CENTER PIN
PIN 2 BNC 2 SH - SCA 2 IN SH - SHELL

BNC - TX

1

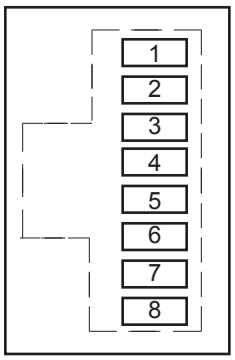
PIN 1 BNC 1 HI - TX 1 OUT HI - CENTER PIN
PIN 2 BNC 1 SH - TX 1 OUT SH - SHELL

2

PIN 1 BNC 2 HI - TX 2 OUT HI - CENTER PIN
PIN 2 BNC 2 SH - TX 2 OUT SH - SHELL

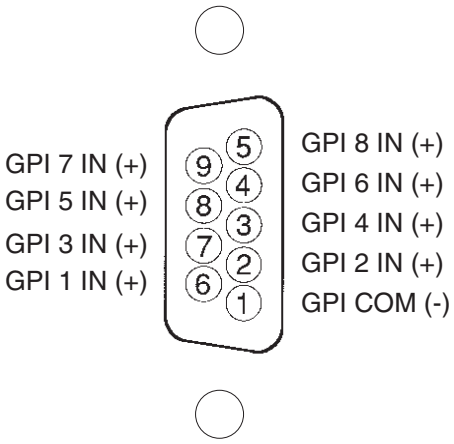
Ethernet - RJ-45

RJ-45 ETH

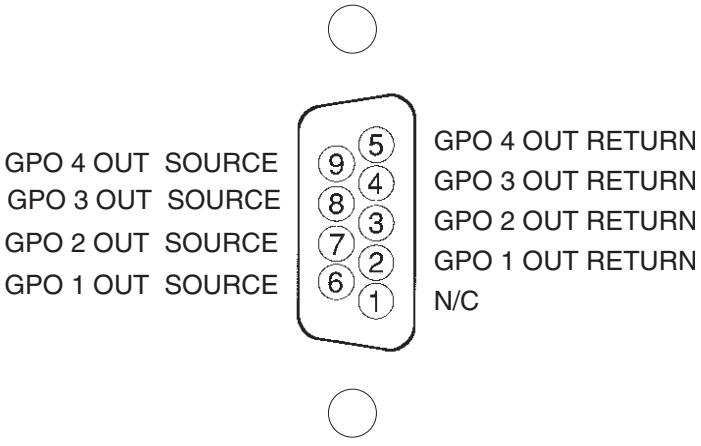


General Purpose Interface - DB-9

GPI - DB-9



GPO - DB-9



VP-8IP Features

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VP-8IP Quick Start Setup Guide

We know that you're probably in a hurry to get on the air with your new VP-8IP. If you have little technical expertise with audio processor fundamentals, the following checklist will get you up and running quickly and without having to know a lot of techie stuff.

This is followed by several pages of information that falls into the category of "stuff I need to know right now because I don't have time to read the whole manual." Please check out the various chapters of this manual if you need some additional information. If for some reason you find that you're really stuck, you can always call our friendly technical support folks at (252) 638-7000.

Please inspect the contents of the package to ensure that the following items are present:

Wheatstone VP-8IP Unit	Quantity 1
VP-8IP Guru Software & Technical Manual CD	Quantity 1
AC Power Cord	Quantity 1
Wheatstone Warranty Card	Quantity 1

The photo below shows the rear panel of the VP-8IP with various connectors associated with an installation:

VP-8IP Rear Panel Connections

Steps to Success

1. Install the VP-8IP Guru software on a host PC. This will allow you to control the various setup options required for successful installation. The default IP address of the VP-8IP is 192.168.1.198. Refer to the Network Installation section for detailed information on reconfiguring the VP-8IP IP address if this will not work for you.
2. Install the VP-8IP in the equipment rack using at least two rack screws. If only two screws are used they must be installed in the bottom holes of the rack ears to prevent undue stress on the front panel and possible bending.
3. Connect the Analog or Digital audio inputs as appropriate for your installation.
4. Make the audio output connections that are appropriate for your installation.
5. Connect local AC power to the unit. There is no power switch and power is applied to the unit as soon as you connect it to a suitable source of AC power. The power supply is universal and accepts line voltages from 90VAC to 260VAC, 50 or 60Hz.
6. Start the VP-8IP Guru, configure a device if you haven't done so already (see page 2-6), and then connect to the unit.
7. Click on the Calibrate Input/Output Menu tab in the Guru and select the input source to match the Input type you connected to the rear panel. If your audio source is already providing an audio feed, you should see meter activity on the VP-8IP front panel Input and Output meters.
8. While still in the Calibrate Input/Output menu and observing the Input meters within the Guru or on the front panel, adjust the Input Gain control (AES or Analog as required) until the input meters are peaking at no more than -12dBFS with normal program levels being applied.
9. If using the Analog outputs pre-emphasized, navigate to the Calibrate Input/Output screen and select the check box for Analog L/R. Then adjust the TX 1 and TX 2 controls to set the correct Left and Right output levels respectively.
10. If using the Analog outputs de-emphasized navigate to the Calibrate Input/Output screen and select the check box for Analog L/R deemph (de-emphasized). Then adjust the TX1 and TX 2 controls to set the correct Left and Right output levels respectively.
11. If using the AES Outputs, connect them to the equipment following the VP-8IP, navigate to the Calibrate Input/Output screen, and adjust the AES Output level appropriately.
12. If using the Composite outputs associated with the rear panel BNC connectors, navigate to the Calibrate Input/Output screen and select the check box for Analog MPX. Then adjust the TX 1 and TX 2 controls to set the correct modulation levels for your system.

When Analog MPX is selected, the balanced Analog Left/Right outputs are also transmitting composite stereo, while the BNC outputs are unbalanced. If you have an exciter that provides a balanced composite input (most do, even though the input is a BNC connector) the best possible signal to noise performance will be obtained by using the VP-8IP balanced outputs.

The easiest way to prepare an interconnecting cable is to use a cable with a BNC already attached at one end. If a BNC is also present on the other end, remove that BNC and replace it with an XLR female connector. The suggested wiring is as follows:

→ Cable shield to XLR Pin #3

→ Cable center conductor to XLR Pin #2

The XLR Pin #1 can be ignored unless operating in a very high RF environment where it may be advisable to connect a 0.01μF capacitor between XLR Pin #1 and XLR Pin #3.

We ship every Wheatstone Audio Processor with the FM Quick Start preset as the factory default (preset 10). This preset provides a well balanced sound and is a good starting point for a variety of program formats.

It's good to remember that our preset names don't necessarily mean that the preset can ONLY be utilized for a format with a similar name. Please feel free to experiment and try each preset we provide to see how it sounds on your station and in your market!

From time to time we'll be adding new presets to our Wheatstone Processing website for you to download and play with. Presets can easily be installed into your VP-8IP using the Preset management features within the remote GUI.

The following pages contain additional startup information. Please excuse us if there is some repetition of information; we know that some of you will be jumping around in the manual rather than reading it sequentially. After all, we're engineers too!

Introducing the VP-8IP

VP-8IP is a *collection* of deep enhancements to the VP-8PLUS product that are included in Version 4.4.0 or higher software. The essential parts of the new software and hardware, and how it is different from the previous release, are:

- Wheatnet-IP I/O
- Baseband192
- Improved front panel setup and configuration capability. No need to have a PC handy for installation and setup;
- Improved processing functions that improve audio consistency, clarity, and overall loudness while lowering distortion.

Adjusting the Headphone Level

When used in conjunction with the rotary encoder, the lighted pushbutton allows the audio level of the front panel headphone jack to be adjusted.

To turn the volume up, press in the lighted pushbutton and hold it while turning the rotary encoder clockwise. The relative position of the control is shown by digits ranging from "0" to "100," with "0" being "Off" and "100" being full volume.

Configuring the IP Address

The VP-8IP is shipped from the factory with a default IP address of 192.168.1.198. Please follow the steps below *noting that if the VP-8IP's default IP address of 192.168.1.198 will work on your network you should skip this procedure.*

The IP address can be set from the front panel of the VP-8IP by navigating the menus. Turn the front panel PRESET-HDPN CONTROL encoder knob until NETWORK shows in the display, then press either the encoder or the illuminated "W" button to drop down a level in the menu. Scroll again to EDIT and press. You will see the IP address with the first octet (192, assuming the factory default address) flashing. For each octet, dial up a new number if necessary, then press to accept. After doing the IP address the subnet mask will be presented for possible editing, followed by the gateway.

Once the VP-8IP IP address is set, proceed to the section below titled **Telling the GUI How to Find the VP-8IP**.

The IP address can also be edited using the Guru, as described in the following section.

Please note that, whether the IP address is changed from the front panel, or changed via the GUI (see page 2-6 or 3-4), or left at the factory default setting, you will still need to set up the GUI with the VP-8IP IP address before the GUI will be able to control the unit.

Operating the VP-8IP Remotely – Installing the GUI Software

For remote operation via the Guru software supplied with the unit, insert the VP-8IP software CD into a Windows computer and follow the on screen instructions to install the software. If the software installation does not automatically start when the CD is inserted into the drive, you can start the installation manually by:

- Click Start
- Then click Run
- Then click Browse
- Browse the "My Computer" device tree to locate the CDROM device and then double click it.
- When the contents of the CDROM drive appear in the window, locate the *VP-8ipGUISetup_x_x_x.exe* file (where x_x_x is the version number) on the CDROM and double click it.
- Follow the on screen instructions to complete the Guru installation.

Configuring the VP-8IP TCP/IP Address*

Once the Guru has been installed you must configure it and the VP-8IP so that they can communicate with each other. This requires configuration for both the VP-8IP (if changing from the factory-programmed default IP address, subnet, and gateway settings) and the remote GUI so that they agree on the networking parameters.

Before starting the configuration procedure, please locate and carefully make a note of the VP-8IP's 12-digit MAC Address which is printed on a label either on the top cover or the rear panel. This address will be in the format 00:50:C2:xx:xx where xx:xx:xx are the digits unique to your VP-8IP. You can also read the MAC Address from the VP-8IP front panel.

* *The VP-8IP does not support DHCP (Dynamic Host Configuration Protocol) and therefore requires a static TCP/IP address on the network. Its presence on the network will not interfere with DHCP addressing of other network connected devices as long as the IP address that is configured for the VP-8IP does not conflict with the address of any other device on the network. Please consult your friendly IT manager if necessary.*

Note that if the MAC address is not entered carefully and correctly in the following steps, the VP-8IP's IP address will not be changed!

- Start the VP-8IP Guru software. Then right click on the control area of the GUI and select Hardware/Assign IP Address. The window on the right (Figure 1) will appear.

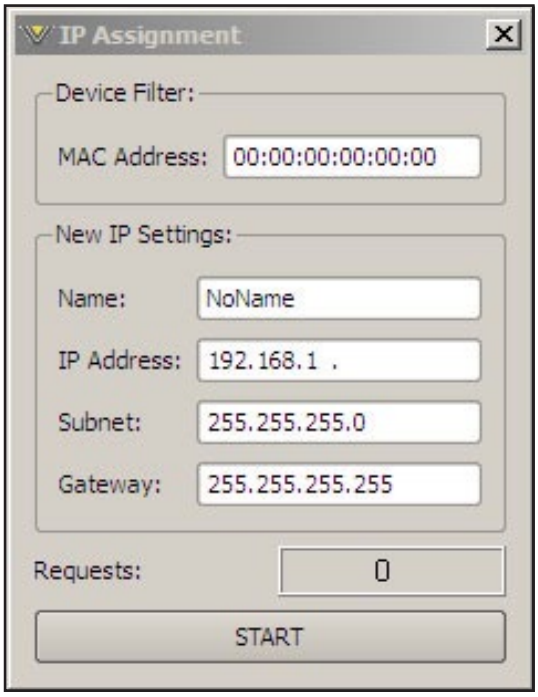
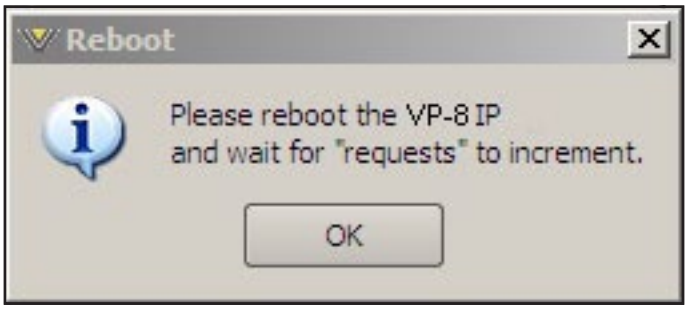


Figure 1
VP-8IP IP Address Configuration

- Next, carefully enter your VP-8IP's MAC address in the MAC Address field. The MAC address must be carefully entered because the GUI will "send" the IP address information to the device that owns the MAC address that was entered in this step. If the MAC address is incorrect, the VP-8IP will never "hear" it.
- Next, enter a pet name for your VP-8IP, like "WABC VP8 2," etc.
- Then, enter the IP address that you wish the VP-8IP to have, noting that this address must be unique if the VP-8IP will be communicating over your network.
- Next enter the desired Subnet Mask and Gateway IP addresses. In some cases (but not all) it is sufficient to use the default values from the form.
- Once this is done, click the *Start* button at the bottom of the IP address configuration box. A message similar to the one below should appear:

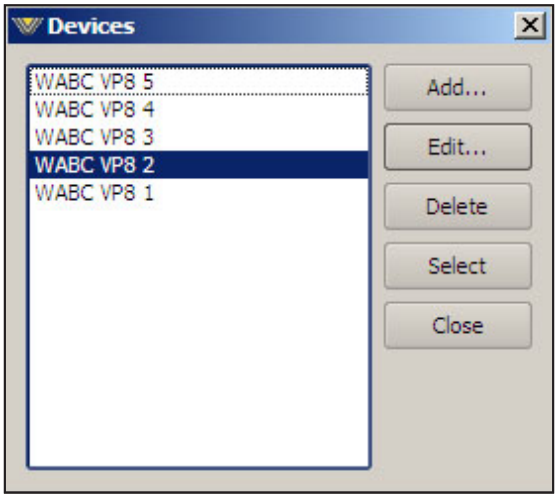


- Click *OK*, then remove power to the VP-8IP for a few seconds, and then reapply it. As the VP-8IP is booting up, the *Requests* number in the IP Address window should increment to something other than zero (please see the bottom of the image in Figure 1). When this occurs the VP-8IP has been programmed with its new IP address and is ready for use.

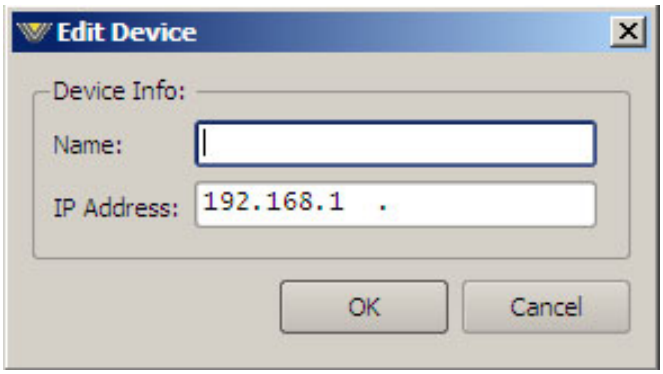
Telling the GUI How to Find the VP-8IP

Now that the VP-8IP has its own TCP/IP identity, we must configure the GUI so that it can talk to the VP-8IP. This is done by adding to the list of devices that the VP-8IP GUI knows about. To do this:

- Right click on the control area of the Guru and select *Devices*.
- The following *Devices* window will pop up:



- Next, click the *Add...* button and the *Edit Device* dialog box will appear – this box is used for adding new devices or modifying the configuration of existing ones.



- First, enter the name that you wish the GUI to display when connected.
- Next enter the IP address that you configured the VP-8IP for in the previous steps. Note that it is always best to completely delete any existing data in the IP Address box and enter the IP address from scratch. We apologize for this inconvenience.
- After this is done you may click *OK* on the *Edit Device* box.
- Then, in the *Devices* dialog box, highlight (single left click) the VP-8IP device that you just added and click on *Select*. This step tells the GUI specifically which VP-8IP you want to connect to.
- Next, click on the *Online/Offline* button just to the left of the *Online Status* indicator.

In the Status window you may see the message *Trying* as the GUI is handshaking with the VP-8IP. Once the handshake is complete and communication protocols are locked in the Status message should revert to *Online*.

If for some reason the GUI cannot connect with the VP-8IP, the *Trying* status message will remain and will occasionally blink as the GUI retries the establishment of a connection. Under these circumstances the configuration of the GUI and VP-8IP should be carefully examined to ensure that the destination TCP/IP address is consistent between the two.

Of particular importance is that the controlling PC and the VP-8IP must be on the same network subnet. For instance, if the VP-8IP was assigned the IP address 192.168.1.194, then the PC's IP address must be between 192.168.1.1 and 192.168.1.254, noting of course that the VP-8IP and GUI PC cannot share the same IP address.

Additional Notes on VP-8IP GUI Connectivity

The VP-8IP uses both TCP port 55890 and the first available UDP port in the range of 60000 to 60010 during communications sessions.

TCP is used for controlling the VP-8IP because TCP works to repair any transmission errors that need to be corrected. This ensures that the VP-8IP's controls will always do exactly what you told them to do from the remote GUI.

UDP, on the other hand, is the default protocol for returning meter data back to the GUI from the VP-8IP. We prefer UDP because of its low overhead and because we don't really mind if a meter data packet is dropped occasionally. They are updated so fast that missed meter packets are inconsequential to the operation of the unit.

What this means is that network traffic on TCP port 55890 and the UDP port range 60000 - 60010 must be allowed on the portion of the network between the computer that the GUI is running on and the actual VP-8IP hardware. If this is not true, then they will not be able to communicate with each other.

TCP may be used when metering data via UDP does not work because unsolicited UDP packets are being blocked. The system menu has the UDP/TCP metering option.

Installation and Setup Using the Front Panel

The VP-8IP is equipped with front panel menu navigation functions to permit installation and configuration without needing a computer.

The menu is navigated by the rotary encoder knob and illuminated "W" button. The items in the list are arranged in tree form in order to make navigation simple and logical. The menu is structured so that the Preset selection item is always at the top of the tree. Logically the navigation list is as follows:

- PRESETS
- MODE
- INPUT
- OUTPUT
- NETWORK
- VERSION
- ACCESS

An item called <<MENU or EXIT is always present at the end of a list to enable backing out of the currently selected item by one level.

The OUTPUT menu item radically changes as the VP-8IP mode is changed due to large differences in the controls present in the various operating modes. In the following pages the front panel navigation will be examined in more detail.

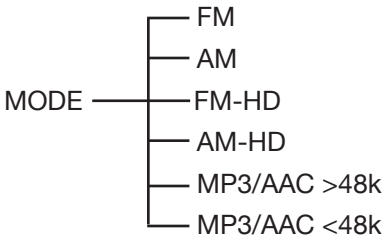
NOTE: Please use the detail shown in the following sections to determine where the appropriate setup controls will be for your particular VP-8IP operating mode and installation type.

PRESETS

The first menu item in the list is PRESETS, which allows scrolling through the list of mode-specific factory (and/or user) presets stored on the VP-8IP hardware. Once a preset is decided upon, pressing either the knob or the lighted "W" pushbutton will select that preset to be active. The active preset will be displayed in the front panel alphanumeric display and will scroll if the name is longer than eight characters. The VP-8IP will respond with TAKEN when the preset is loaded and running.

MODE(s)

The VP-8IP is equipped with *six* completely different operating modes:



- FM, complete with stereo generator and baseband192 digital MPX.
- AM, complete with asymmetrical limiting and selectable lowpass filters and analog transmitter equalization.
- FM-HD, with codec-optimized pre-processing.
- AM-HD, with codec-optimized pre-processing.
- MP3/AAC>48k (specialized processing for codecs utilizing greater than 48kHz bitrates).
- MP3/AAC<48k (specialized processing for codecs utilizing less than 48kHz bitrates).

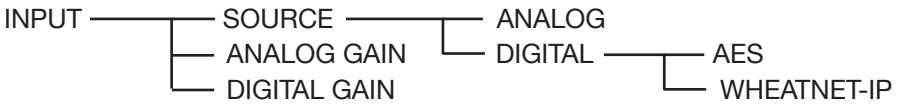
NOTE: THE VP-8IP OUTPUTS WILL MUTE WHILE CHANGING MODES!

When a new operating mode is being loaded into the VP-8IP's memory the audio outputs will go silent and will remain silent until the new algorithm is loaded.

This is *intentional* behavior that prevents clicks, pops and other disturbing audio discontinuities from appearing at the VP-8IP outputs while the new mode's processing algorithms are being loaded into the DSPs and started up.

INPUT

The INPUT menu item allows the appropriate audio input type to be selected, whether Analog or Digital (AES or WheatNet-IP), and also allows independent input gain adjustments to be made for each of the two input types.



We strongly recommend that peak input audio levels are never set so that they peak higher than about -12dBFS as shown on either the GUI or the VP-8IP's front panel meters.

This caution is necessary to allow appropriate headroom for the processing algorithms when they are presented with material that has been highly compressed and possibly also clipped during the mastering process. Such material can achieve far higher peak levels after the first stage of signal processing and those signals may not have enough headroom to avoid clipping within the processor if the input level was already too high.

Said another way, if you hear mysterious distortion or what sounds like clipping on very loudly mastered material it's a good idea to check the input levels to the VP-8IP to ensure they are not spending any significant time at levels of -12dBFS and above. Guru has a "Ref" indicator on its input level meter at the -12dBFS calibration mark as a convenient reminder.

Along with Digital AES3 and Analog inputs VP-8IP is also equipped with WheatNet-IP (WNIP), the Audio Over Internet Protocol (AoIP) used in our Wheatstone BLADE and IP audio network control surface environment. WheatNet-IP permits VP-8IP's input audio and its output to be carried over the same standard CAT5 Ethernet cable as that used for remotely controlling the VP-8IP.

This procedure assumes that your facility is equipped with at least one Wheatstone BLADE and the associated Navigator software. The BLADE used may even be the multiband, eight processor Aura8IP Processing BLADE whose physical inputs and outputs (half analog, half digital) may even be used as a substitute for VP-8IP's physical I/O.

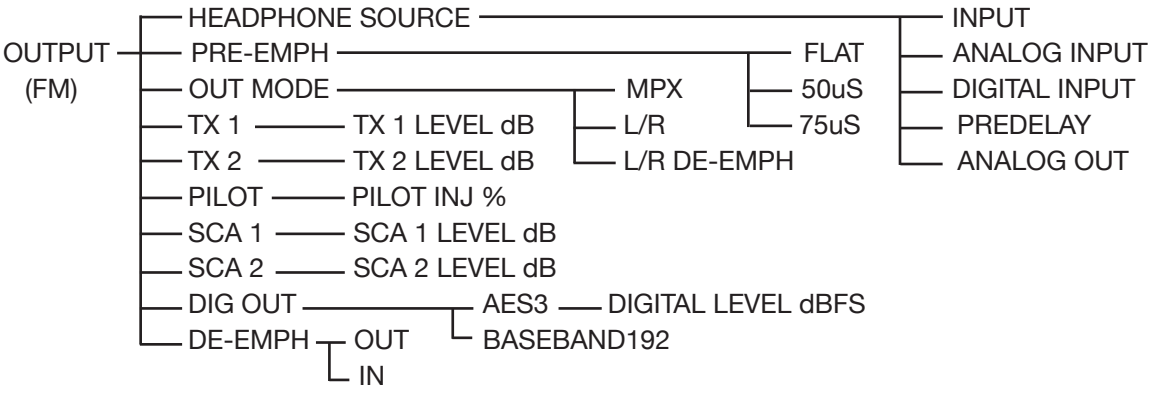
Now it's time to open up WheatNet-IP Navigator to gain access to VP-8IP's WNIP-associated input and output signals.

Once the WheatNet-IP Navigator software is open and you are logged into the system we must create a new device for VP-8IP. An important concept to remember is that VP-8IP is a "third party" device within a WheatNet-IP system. Our Sideboard, an outboard mixer/controller for use with any BLADE's pair of 8-channel utility mixers is another example of a "it's not-a-Blade" third party device. One more example of a third party device is Tieline's Genie codec equipped with WheatNet-IP. These devices, while technically not BLADES, can and do intelligently interact with the WheatNet-IP system and can have signals routed to and from them. What they lack however is the built-in intelligence, pair of 8-channel utility mixers and other specialized technologies that reside inside an actual BLADE.

OUTPUT

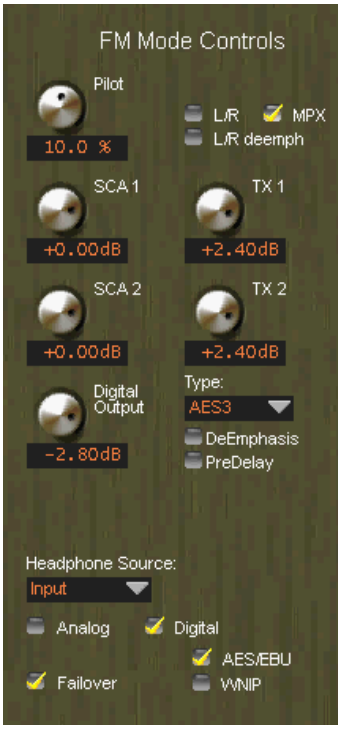
The output menu tree is tailored to the current VP-8IP operating mode, with the most complex being the FM and AM modes due to their specialized processing requirements. The output menu tree for each of the six modes follows along with the corresponding Guru output menu below it for comparison.

FM Mode



MPX Mode

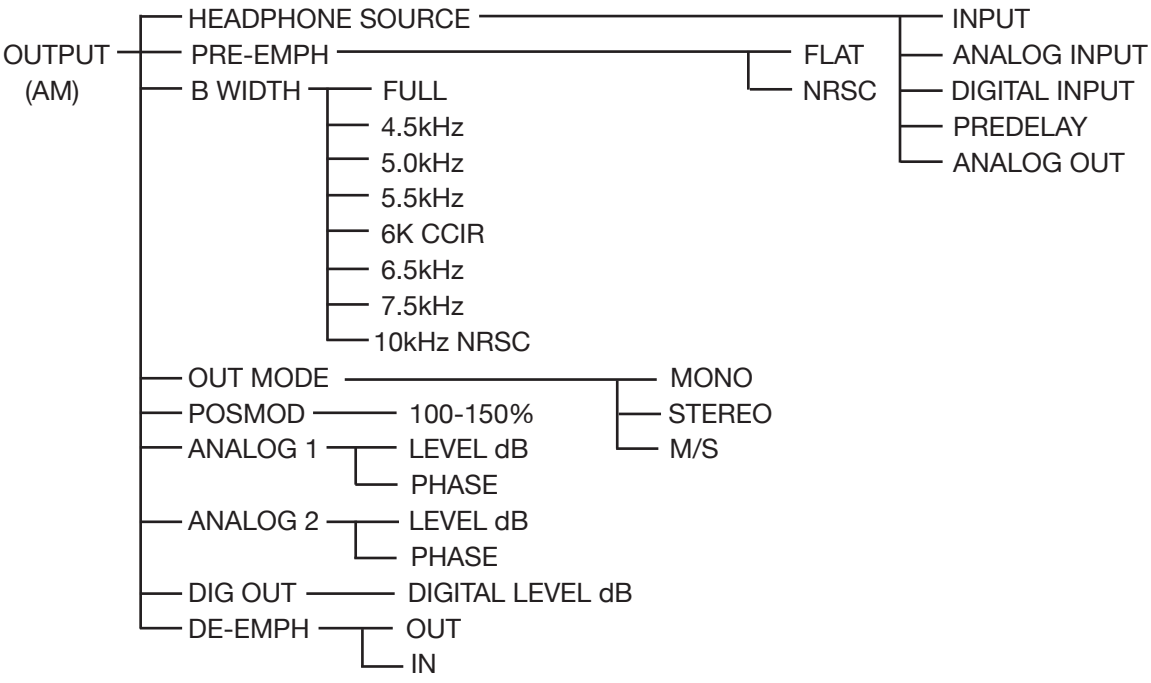
L/R Mode



AES3 Mode

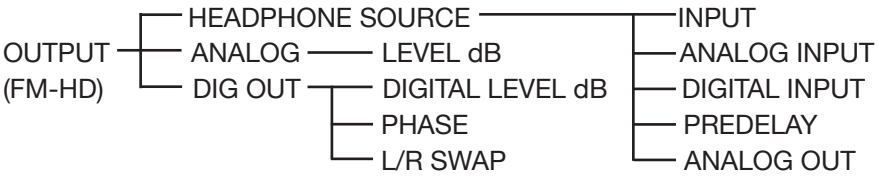
Baseband192 Mode

AM Mode

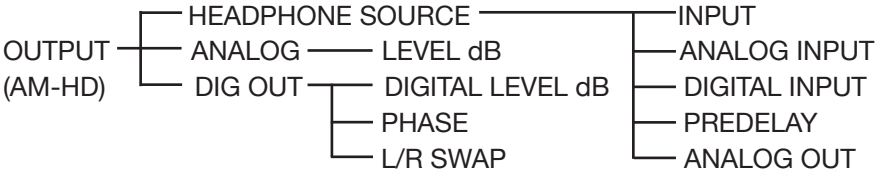

OUTPUT (AM)
HEADPHONE SOURCE
INPUT
ANALOG INPUT
DIGITAL INPUT
PREDELAY
ANALOG OUT
PRE-EMPH
FLAT
NRSC
B WIDTH
FULL
4.5kHz
5.0kHz
5.5kHz
6K CCIR
6.5kHz
7.5kHz
10kHz NRSC
OUT MODE
MONO
STEREO
M/S
POSMOD
100-150%
ANALOG 1
LEVEL dB
PHASE
ANALOG 2
LEVEL dB
PHASE
DIG OUT
DIGITAL LEVEL dB
DE-EMPH
OUT
IN

AM Mode Controls
Bandwidth:
Full
Output Mode:
Mono
Positive Modulation
100%
Analog 1
Phase
+0.00dB
Analog 2
Phase
+0.00dB
Digital Output
+0.00dB
DeEmphasis
PreDelay
Headphone Source:
Input
Analog
Digital
AES/EBU
Failover
VNIP

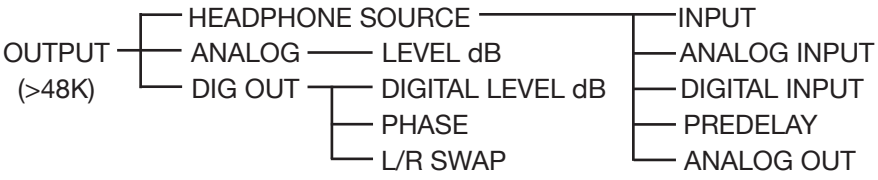
FM-HD Mode



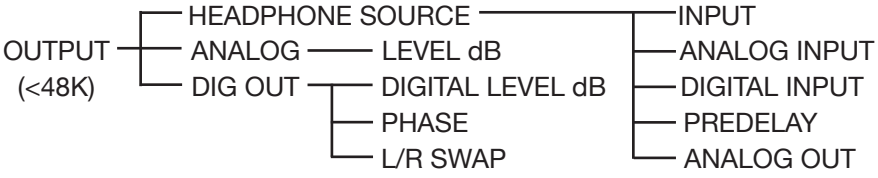
AM-HD Mode



MP3/AAC>48kHz Mode

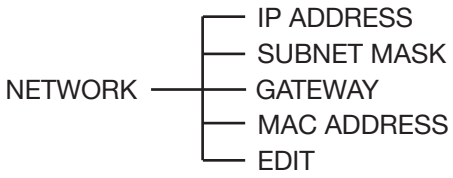


MP3/AAC<48kHz Mode



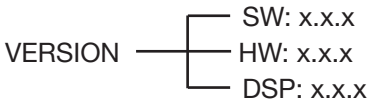
NETWORK

After the Output options the Network Configuration is next in the list.



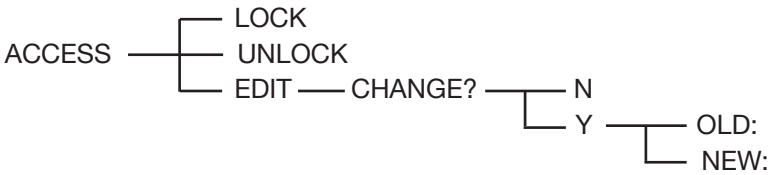
All of the parameters in the Network list can be edited *except* for the MAC address. A MAC address is a *fixed* hardware identification that is intended to be specific *to* and different *for* each networked device. As is the case with most network gear, the MAC address cannot be changed in the field.

VERSION



The version dialog displays the currently loaded versions of operating system, FPGA image, and DSP code, noting that these are displayed values only and cannot be changed without actually changing the version(s) that the VP-8IP is running, i.e., updating it to a new version.

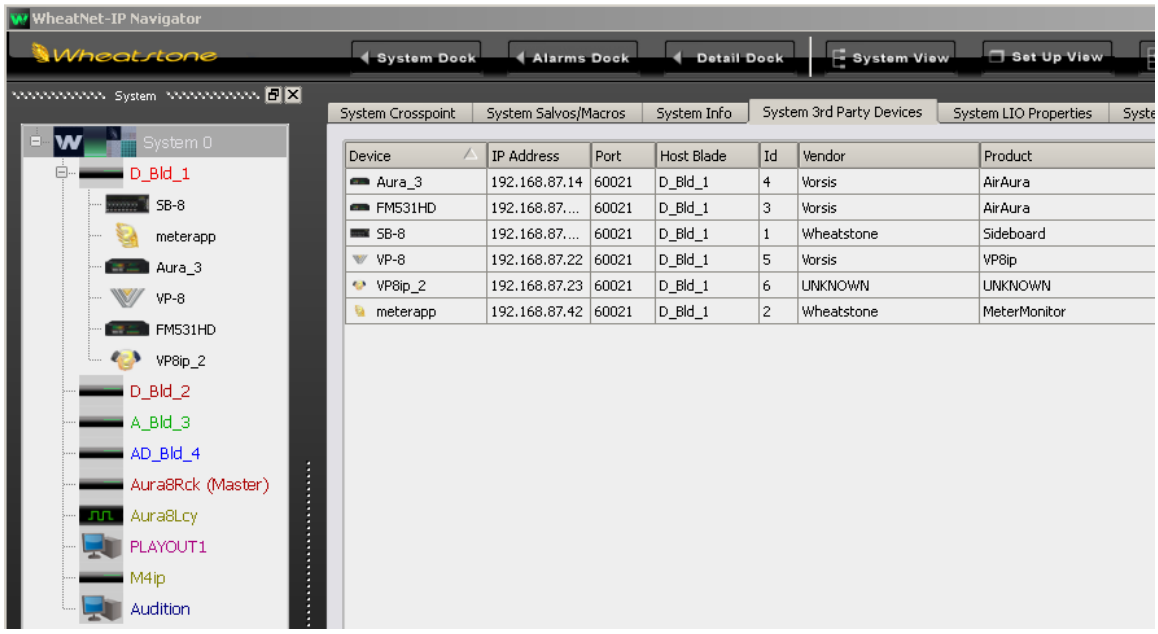
ACCESS



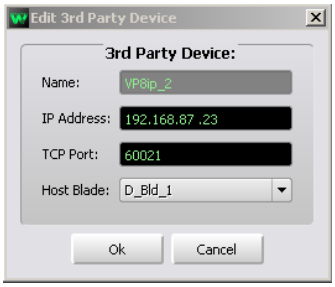
The Access item allows the front panel security features of the VP-8IP to be customized. A front panel password may restrict access to the front panel controls and can be set or changed when necessary to keep curious fingers out of the adjustments.

It is not possible to lock the front panel without first setting a password for it. Likewise, if the front panel has been locked it cannot be accessed without first entering the valid password.

Adding VP-8IP to a WheatNet-IP System

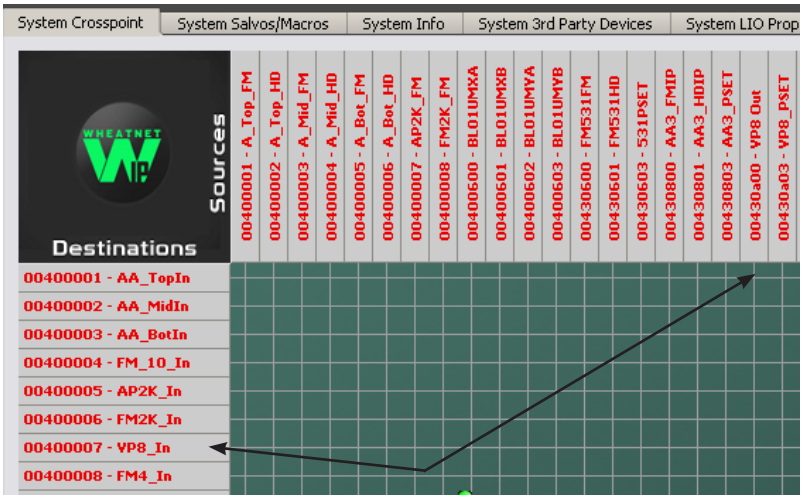


The first thing to do is click on Navigator's *System Third Party Devices* tab which will open a new window listing any third party devices that are already present. In the image above two VP-8IP's already exist (they're in our audio processing lab) but we will be adding another one nonetheless.



Clicking on the *Add* button opens a dialog where a new VP-8IP may be added to the system. Here we will add the new VP-8IP's name, its hardware IP address, and tell the system what BLADE will be its host (while any BLADE can be its host all third party devices must have a host). We do not have to modify the TCP Port setting so we can leave it set to 60021. Once input to the dialog is finished, click on the *Ok* button and it will close.

Next we'll open Navigator's System Crosspoint window and see that our new VP-8IP's Input and output signals are present and available for routing.



Note that as shown in the example at left we've used Navigator's signal name editing capability to assign VP-8IP's input and output signals the names we desire.

Unless or until they are modified by an end user the default input and output signal names appear as “VP8 INP” and “VP8 OUT” in the Navigator software. These signals may be manually crosspointed (and even locked if desired!) to make connections, or I/O routing may be done through the many other methods available within the WheatNet-IP system for routing signals and changing the routes of both audio and logic signals.

More information on configuring and using WheatNet-IP Navigator and the other features and capabilities of the WheatNet-IP system may be found in the WheatNet-IP System product documentation.

Assistance in configuring VP-8IP for WheatNet-IP may be obtained by giving us a call at (252) 638-7000 or by emailing techsupport@wheatstone.com.

Achieving the Desired Sound

First a Few Words About Our Factory Presets

Factory presets are great starting points for many applications, but we'll be the first to admit that they may not be appropriate for every situation.

We at Wheatstone spend a *lot* of time (literally thousands of hours) carefully crafting factory presets that will get you closer to your desired sound quicker than if you had to build them from scratch.

We do our best to provide a lot of good starting points in our factory presets, and we're always improving them, which is why you'll periodically see new VP-8IP presets offered for download on our website.

Each preset has been carefully tuned so that the dynamics control of the audio path through the four-band AGC creates the desired foundation sound for each preset. Differences will then be found in the final limiting schemes of each of the six operating modes.

Each preset was arrived at through *many* hours of listening with appropriate program material, and each is our best personal assessment of the tradeoffs between what the DSP algorithms are capable of and what a typical listener might find appealing. Please feel free to experiment with the presets, as each one has a unique sound, but don't fall into the trap of thinking that (for instance) the "Country" presets are only good for "Country" music formats. The preset that sounds best with your program format is the one to use, either as is, or as the foundation sound on which to add your personalized touches.

What Sound Are You Trying to Achieve?

There are four basic goals that most people want to achieve when they install a new audio processor:

1. They want to be *louder*;
2. They want to be *cleaner*;
3. They want to have *better* bass;
4. They want a *cleaner* and *more open* high end.

About Bass

The VP-8IP contains our specialized Bass Tools™.

Bass Tools™ extracts, reveals, and then controls low frequency nuances in the incoming program that are simply not heard with any other audio processor.

Is There a Caveat?

Of course! The majority of radio listeners don't have extremely high quality monitoring equipment. Therefore care is advised when adjusting the VP-8IP to accentuate the extremes of the audio spectrum.

Program energy at the frequency extremes is typically several dB below that in the more easily heard midrange. Because of the nuances of human hearing, very low and very high frequencies take a *lot* more modulation power to become

audible to the average listener. Over-boosting bass or treble energy that can be heard by only a small percentage of the audience can result in either reduced loudness and/or higher distortion for the majority of the listeners.

What Is Your Short Term Goal?

The usual short term goal that is related to installing new audio processing involves generating large amounts of loudness in order to attract attention to the station as a listener explores his options.

Almost as important as the station playing a listener's favorite record when they've tuned in, a signal that's *LOUD* carries with it a certain "authority" that can make a listener stay tuned long enough to see what's next.

While loudness can attract a listener, so too can it drive them away! Extreme loudness usually carries with it subtle (or not so subtle!) forms of distortion and other artifacts that can subconsciously tire a listener and make them tune away without even realizing why. Loudness then, for loudness' sake, must be approached with a bit of caution.

How Do I Make It Louder?

The VP-8IP is capable of holding its own against virtually any similarly priced audio processor. In general terms, in order to just be louder (we'll cover that first because it's actually a bit easier) you should:

- Increase the density in the four band AGC by operating it with faster attack and release times
- Increase the Drive to the eight band limiters
- Carefully use some equalization in the parametric section to gently boost frequencies that the ear is most sensitive to (1kHz to 4 kHz)
- Drive the limiter section harder to further increase the amount of RMS energy being generated by the processing
- Operate the eight band section with faster attack and release times, in fact sloping them slower for the lows, faster for the highs (see Proportional Drag in the tutorial section for the Pro GUI, page 4-15); this can create additional perceived loudness

The eight band limiters can be very deceptive. Because they operate with comparatively fast attack and release times and in narrow frequency bands they can generate some mighty impressive loudness without the pumping, dynamic artifacts, or distortion typically generated by "broadband" control schemes. The architecture of the VP-8IP sets you free to be as loud as you want. Be aware that excessive limiting depth adds little extra loudness and carries with it the risk of driving away your long-term listeners, especially females!

What Is Your Long Term Goal?

If the long term goal isn't to be the loudest signal on the planet you can still be competitively loud while also sounding better, cleaner, brighter, and punchier than other signals that the listener can choose from.

When loudness isn't the first priority your options open up quickly! Being "cleaner" than the other signals increases a listener's time spent listening.

Being cleaner usually means you're not leaning into the processing as hard, which also means that you can use tools like equalization more freely than a really loud signal in order to craft the sound you want. This is because equalization (EQ) eats up headroom, and if you're trying to be really loud you need all the headroom you can get and therefore less EQ can be used.

Clean and Loud

To accomplish Clean *and* Loud we can use the behavior of the ear and brain (psychoacoustics) to fool it into thinking the sound is louder than it actually is. The following adjustments, together or in combination, can be used to create a sound that's loud and punchy, but also subjectively clean:

- Use slower attack times in the AGC sections
- Use slower release times in the AGC sections
- Use slower attack times in the eight-band limiters
- Leave the eight band release times fairly fast (<100ms)
- Increase the drive to the final limiter (but not too much!)
- Use a fairly low AGC Super Low to Low crossover frequency – perhaps around 70-80Hz
- Use a fairly high Mid High to High band crossover – perhaps 6 or 7kHz
- Be gentle with any mid-frequency equalization. In fact, partially subduing the range (no more than 1dB) between about 600Hz and 2.5kHz can subjectively clean up otherwise harsh-sounding material. To do this, use the parametric equalizer configured for post AGC/Compressor operation with a center frequency of around 1.2kHz, a bandwidth of about 1.5 octaves, and a cut of no more than about 1 to 2 dB.

The above combination will somewhat "relax" the sound because the AGC sections are operating slower, slightly lowering their average output levels, and also because of the slightly reduced levels in the ear's most sensitive region due to the midrange equalization.

On the other hand, you'll note that the drive has been increased to the limiters. The increased limiting depth, albeit only momentary because of the longer AGC attack times, will increase the subjective loudness on larger signal peaks, restoring a sense of loudness and punch and creating a feeling of dynamics and loudness without generating the same kind of distortion artifacts that dense compression and higher duty cycle clipping would create.

With the crossover frequencies at the ends of the audio spectrum carefully tuned to the extremes (within reason!), the dynamic behavior of the AGC/Compressor can create that familiar "smile curve" equalization and lend an overall sense of Hi-Fi to the sound without having to rely on static equalization.

Can I Generate That "Sixties" Compression Sound?

Yes! The way to do this is to purposely drive the multiband section harder and then trade off the slower acting AGC against the faster time constants of the compressors. This is accomplished by operating the AGC Backoff control at negative numbers greater than about -2.0dB to -3.0dB, causing more gain control to be accomplished using the

faster compressor time constants. If you are careful to not run the Backoff control too far negative, the AGC will provide a "platform" that the compressor can work against, preventing rapid suck-up of background noise that might be undesirable.

A word of caution if you're also using reverb to augment that sixties sound… with faster and deeper compression reverb tails will become exaggerated. This artifact can be reduced in several ways:

- Reduce the amount of reverb;
- Operate the AGC Backoff control at slightly less negative numbers;
- Reduce the overall drive to the AGC/Compressor section;
- Raise the Gate Threshold to less negative numbers and decrease the Gate Delay time to something under 100mSec.

You can use these methods alone or in combination to control reverb tails. Alternately, if your station's sound demands a fair amount of reverb during non-speech programming, then announcer microphones will almost certainly become a challenge. In this scenario you may have to come up with a way to reduce the contribution of the reverb, but only when talent microphones are on.

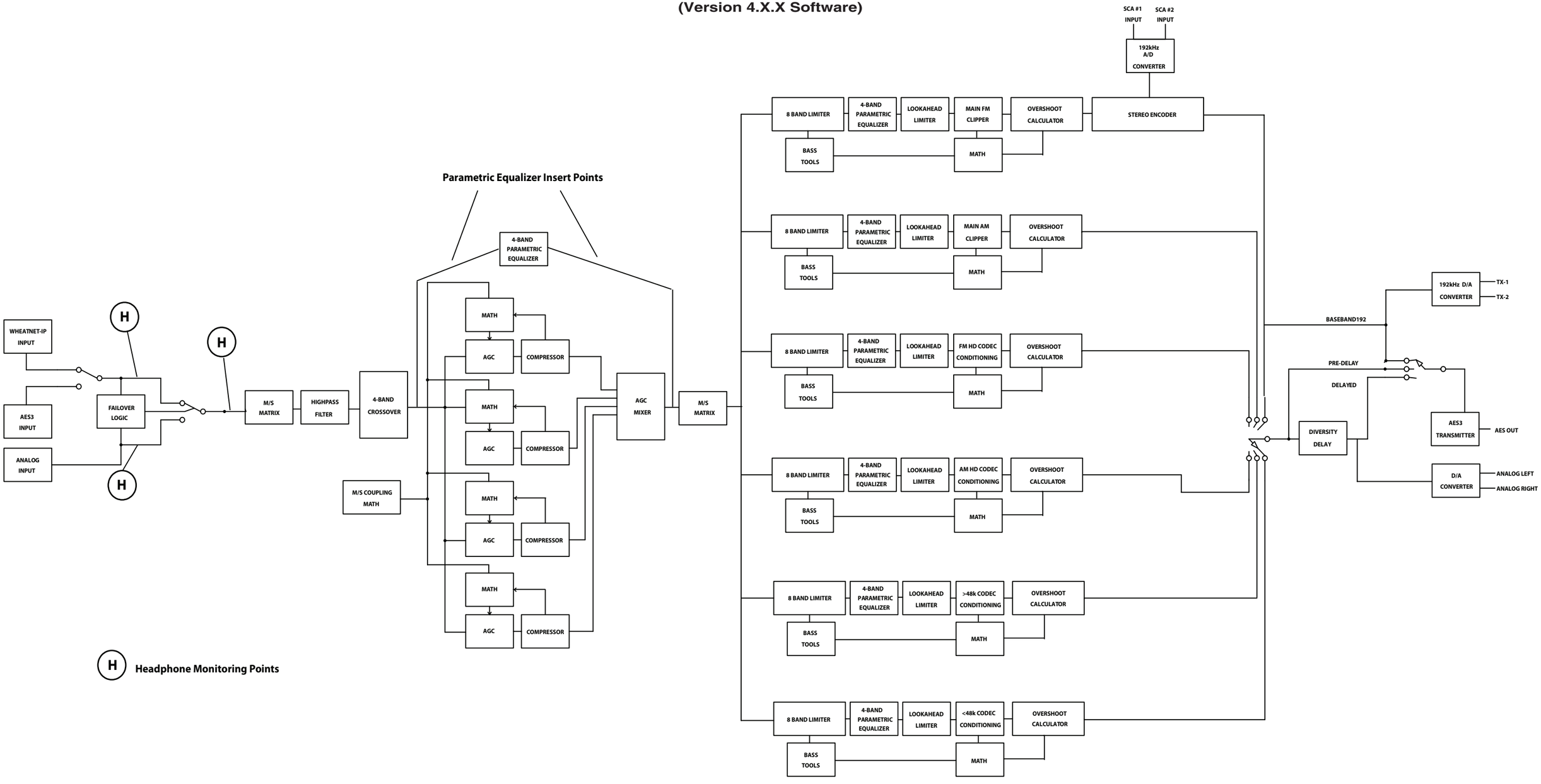
VP-8IP Front Panel Menu

Wheatstone Corporation

VORSIS ULTRA-HIGH RESOLUTION PROCESSING

VP-8IP General SIgnal Flow

(Version 4.X.X Software)



VP-8IP Signal Flow Diagram

VP-8IP
Audio Processing Guru®

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VP-8IP
Audio Processing Guru®

Introduction

Guru is our simplified user interface whose under-the-hood intelligence distills the VP-8IP's more than 230 processing adjustments down to just six very easy to use controls. Finding the perfect sound has never been quicker or easier!

The Guru tool is designed for those who would rather change the sound of our Factory presets without hundreds of tweakable parameters staring back at them.

VP-8IP Guru - Adjust Sound

Guru has only three basic pages hosting controls that an end user might need for installing the processor and adjusting its sound.

- The first page of Guru is called *Adjust Sound* and is solely dedicated to making refinements to our Factory preset's default sound.
- The second page of Guru is the *Calibrate Input/Output* page and hosts the Input and Output Calibrations, Headphone Monitor Source, Input Audio Source and Audio Failover behavior selectors.

- The third and last page of Guru is the *Configuration* page, hosting controls for selecting the processor mode, configuring the GUI to communicate with the host audio processor and programming the Scheduler.

Note that for user convenience we've *repeated* the appearance of some of the controls across various pages noting that the Input, Output and processing-related metering are always visible on all three pages.

Adjust Sound

Calibrate In/Out

Configuration

Installing the Guru Software

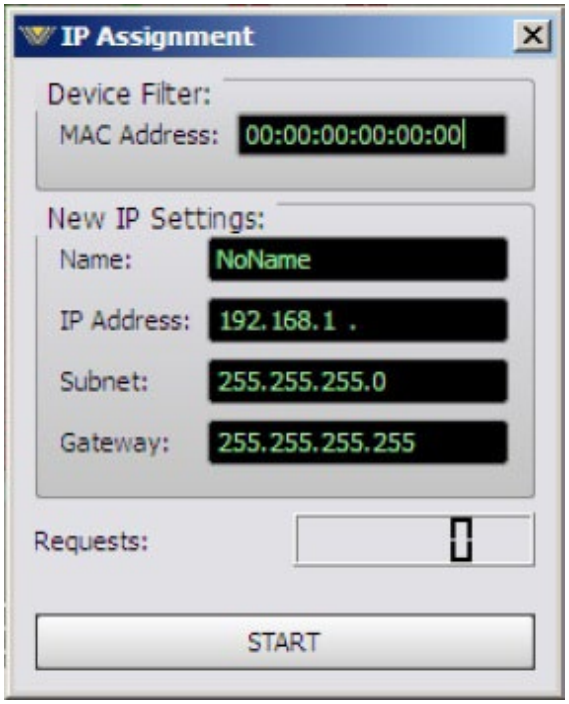
The Guru software is provided to permit access to VP-8IP sound processing controls not accessible via the front panel. To install the Guru software insert the VP-8IP software CD into a Windows computer and:

- Click Start
- Then click Run
- Then click Browse
- Browse the "My Computer" device tree to locate the CDROM device and then double click it.
- When the contents of the CDROM drive appear, locate the *VP-8IP GUI Lite Setup_x_x_x.exe* file (where x_x_x is the version number) on the CDROM and double click it.
- Follow the on screen instructions to complete the GUI installation.

Configuring the VP-8IP TCP/IP Address*

Please note that, whether the IP address is changed from the front panel (see page 2-5), or changed via the GUI, or left at the factory default setting, you will still need to set up the GUI with the VP-8IP IP address before the GUI will be able to control the unit.

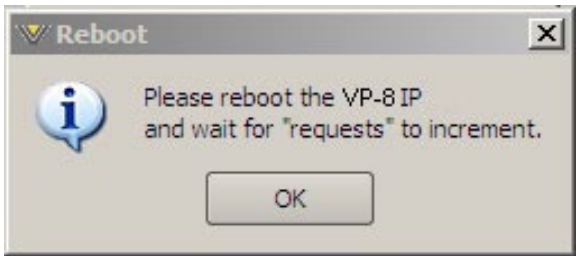
The VP-8IP is shipped from the factory with a default IP address of 192.168.1.198. Please follow the steps below, noting that if the VP-8IP's Factory default IP address of 192.168.1.198 will work on your network you should skip to step 1 on page 3-5.



1. Before starting the configuration procedure, please locate and carefully make a note of the processor's 12-digit MAC Address located on a label either on the top cover or the rear panel. This address will be in the format 00:50:C2:23:xx:xx where xx:xx are the digits unique to your processor. You can also observe the MAC address on the VP-8IP front panel display. Dial NETWORK > MAC ADDRESS.
2. Next, start the VP-8IP Guru software. Then right click on the control area of the GUI and select the *Hardware/Assign IP Address* option. The window to the left should appear.
3. Carefully enter your processor's MAC address in the MAC Address field.*
4. Next, enter a pet name for your VP-8IP, like *WABC VP-8IP 2*, etc.

** The MAC address must be carefully entered because the GUI will "send" the IP address information to the device that owns the MAC address entered in this step. If the MAC address is entered incorrectly, the VP-8IP will never hear it, causing it to remain on its factory default IP address!*

5. Then, enter the IP address that you wish this VP-8IP to have, noting that its address must be unique if the VP-8IP will be communicating over a network with other devices attached to it.
6. Next enter the desired Subnet Mask and Gateway addresses.
7. Once this is done, click the *Start* button at the bottom of the IP address configuration box. A message similar to the one below should appear:
8. Click *OK*, then remove power to the VP-8IP for a few seconds, and then reapply it. As the VP-8IP is booting up, the *Requests* number in the IP Address window should increment to something other than zero (please see the bottom of the image in the *Requests* figure above.
9. When the requests number increments above zero the VP-8IP has been programmed with its new IP address and is now ready for use.



Device Configuration – Connecting to the VP-8IP

1. Now that the VP-8IP has its own TCP/IP identity, we must configure the GUI so that it can talk to the VP-8IP. This is done by adding to the list of devices that the VP-8IP's GUI knows about. To do this:
2. Right click on the control area of the Guru and select *Devices*. The *Devices* window shown on the right will pop up:
3. Next, click the *Add* button and the *Edit Device* dialog box will appear – this box is used for adding new devices or modifying the configuration of existing ones.
4. First, enter the same pet name that you used when you assigned the IP address to the VP-8IP. This step is not absolutely necessary but it makes for easy identification when there are many VP-8IP's in a facility.

5. Next enter the IP address that you configured the VP-8IP for in the previous steps. It is recommended that any existing data in the IP Address box be deleted and new IP address entered from scratch. This ensures that no undesired blanks or spaces creep into the address.
6. After this is done you may click *OK* on the *Edit Device* box.
7. Then, in the *Devices* dialog box, highlight (single left click) the VP-8IP device that you just added and click the *Select* button. This step tells the GUI specifically which VP-8IP you want to connect to next.

8. Next, click on the *Online/Offline* button just to the left of the *Online Status* indicator. In the Status window you may see the message *Trying* as the GUI is handshaking with the VP-8IP. Once the handshake is complete and communication protocols are locked in the *Online Status* message should revert to *Online*.

If for some reason the GUI cannot connect with the VP-8IP, the *Trying* status message will remain and will occasionally blink as the GUI retries the establishment of a connection.

Under these circumstances the configuration of the GUI's networking parameters, those of the VP-8IP, and sometimes even the interconnecting network, should be carefully examined to ensure that the destination TCP/IP address assigned to the VP-8IP and Guru conforms to the expected networking parameters for the host network.

Unless special routing has been configured by the IT department the controlling PC and the VP-8IP must be on the same network subnet. As an example, if the VP-8IP was assigned the IP address 192.168.1.194, then the PC's IP address must be between the addresses 192.168.1.1 and 192.168.1.254, noting that the VP-8IP and GUI PC cannot share the *same* IP address.

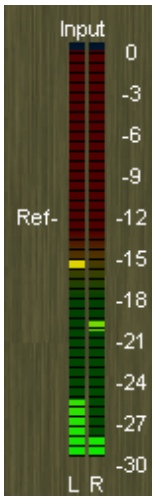
NOTE: TCP port 55890 must be open in any switch or router configuration in order to permit traffic between VP-8IP and GUI.

Setting the Input Audio Levels

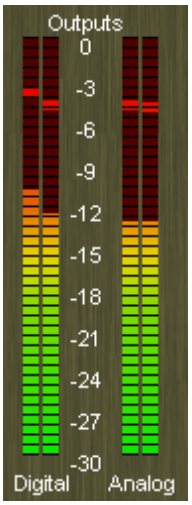
- Connect the Analog or Digital audio inputs as appropriate for your installation.
- Start up the VP-8IP Guru application and configure a device if you haven't done so already and then connect to the unit.
- Click on the *Calibrate Input/Output* Menu tab at the top of the GUI and select the input source to match the Input type you connected to the rear panel. If your audio source is already providing an audio feed, you should see meter activity on the Input and Output meters.

While still in the Input menu and observing the Input meters within the GUI or on the front panel, adjust the *Input Gain* control (AES or Analog as required) until the input meters are peaking around the -12dBFS *Ref* marking with normal program levels applied.

NOTE: To ensure proper headroom inside the processor's algorithms frequent peak levels exceeded the -12dBFS *Ref* marking on the Input meters are strongly discouraged.



Input Meters



Output Meters

Setting the Output Audio Levels

Now it's time to make the audio output connections as appropriate for your installation and adjust audio output levels. We'll use the VP-8IP's FM Mode for this example, although the procedure is similar for all six of the VP-8IP's processing modes.

If using the Analog outputs pre-emphasized, navigate to the *Calibrate Input/Output* screen and select the check box for *Analog Out Mode*: *L/R*. Then adjust the *Analog Output* Gain control to set the correct output level.

If using the Analog outputs de-emphasized navigate to the *Calibrate Input/Output* screen and select the check box for *Analog Out Mode*: *L/R de-emph* (de-emphasized). Then adjust the *Analog Output* Gain control to set the correct output level.

If using the AES Digital Outputs, connect it to the equipment being fed by the VP-8IP, navigate to the *Calibrate Input/Output* screen and adjust the *Digital Output* level as appropriate. The Digital Output may be operated either pre-emphasized or de-emphasized.

If using the Composite outputs associated with the rear panel BNC connectors, connect the VP-8IP to the exciter's MPX or baseband input connector (it's usually a BNC connector like on the back of VP-8IP). Then navigate to the *Calibrate Input/Output* screen, select the check box for *Analog MPX*, and adjust the *TX1* and *TX2* controls to set the desired modulation levels.

A note for the techies: When the *Analog MPX* output mode is selected the balanced Analog Left/Right outputs are also transmitting composite stereo, not balanced Left/Right audio as you might expect. If you have an exciter that provides a balanced composite input (most do, even though the composite input is a BNC connector) the best signal to noise performance will be obtained by using the VP-8IP's balanced outputs. Give us a call if you'd like to interface to your exciter using this method and we'll walk you through what's involved.

Note that the control for setting the Diversity Delay (normally encountered in HD installations) is not present in Guru.

Taking Presets

The VP-8IP arrives from the factory set to operate in its FM Mode and with that mode's "A Quick Start" preset selected. This preset provides a well balanced sound and is usually a good starting point for a variety of program formats.

To select another preset (when the front panel is unlocked) simply turn the rotary encoder until the desired preset name is displayed (the display will flash, as well as scrolling the preset name). Then press and release the encoder wheel. The display will briefly show TAKEN and then begin showing the new preset name. If the name is longer than eight characters the display will scroll the name.

It's good to remember that our preset names don't necessarily mean that the preset can ONLY be utilized for a format with a similar name. Please feel free to experiment and try each preset we provide to see how it sounds on your station and in your market!

We frequently add new presets to our Wheatstone Processing website for customers to download and play with. Presets can easily be installed into our VP-8IP using the Preset management features of either Guru or Pro GUI (more about Pro GUI later on…).

Tuning Audio Processing – an Introduction

During the many years we've spent designing, building, installing and tuning audio processors in all kinds of broadcast facilities there are a few things that have consistently been proven true. Since this seems like the perfect place to do it we'd like to share some of them with you now:

No one at a radio station listens more critically to it than after the audio processing is changed.

It can take many weeks or more to narrow down the perfect sound for a station - even more for a station in a major market.

Radio markets are moving targets – you make a change, "they" make a change, you make another change, "they" make another change...

Expectations of sound improvement when changing audio processors will almost always surpass reality.

No one in the station's audience hears what "radio people" hear.

Changing to a more modern audio processor can make the station sound worse by revealing previously masked problems in the station's audio chain.

There is almost never enough loudness, bass, or high end.

There is no free lunch – you must always pick something good to give up in order to get something that you want even more.

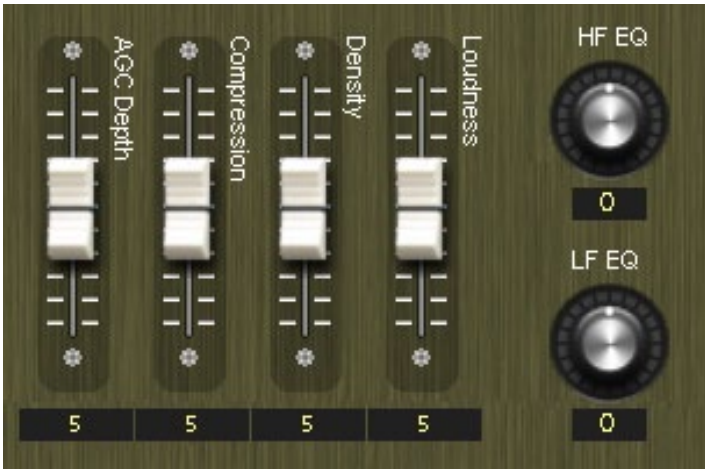
The most important advice that we can share about setting up audio processing, particularly new audio processing, is this:

Make *small* adjustments and listen over a long enough period of time that you get to hear what your current settings sound like on <u>*most*</u> program material (you'll never have time to hear <u>*all*</u>).

Resist the temptation to try to fix the processing for the one song that sounds bad. Instead, try to determine <u>why</u> that cut sounds bad. If it's the source material, fix it. If not, *then* you can work on the processing. Too many get these processes reversed and spend a lot of time tuning and tuning and still get nowhere.

The tweaking controls in Guru are located on the *Adjust Sound* tab of the GUI. On this tab are four vertical sliders for "processing" and two knobs for "equalization." The image below shows how the *Adjust Sound* tweakers appear....

Guru's six tweakers were specifically designed for manipulating Factory presets, and each of the six controls has eleven positions. If someone wanted to know how many different sound settings (preset iterations) that makes, it's easy math. Simply multiply eleven times eleven, and do it six times (once for each control). The resulting number represents how many different sound textures that could be made with one Factory preset – the number is 1,771.561.



Guru's Six *Adjust Sound* Controls

If we take into account just the FM Mode presets we have to multiply the above result by the 21 factory presets to arrive at the new total number of possible sound variations. That turns out to be 37,202.781 (almost 40 million) different sound textures created from just those eleven FM Factory presets and six controls.

Six Controls! Why Not Just One?

Over the years there have been audio processors equipped with a "one-knob" less-more adjustment as a "simplified" user interface. After selecting a Factory preset the less-more control would be turned up or down to see if a desired sound could be found.

When asked about it, most customers told us that one knob usually didn't allow enough control. They wanted *more*, but didn't want it to be complicated or force them to know technical terms like attack time or gate threshold. They just wanted to take a preset and then crank on a couple of tweakers (knobs) that could get them the sound they wanted. We took their advice and set about to design a better way...

There are only six tweakers associated with adjusting the sound of VP-8IP Factory presets. Four of them are assigned to tuning the processing "texture" while the two remaining ones tweak "equalization."

As mentioned before, each tweaker has eleven positions numbered 0 through 10. When a Factory preset is recalled all six tweakers reset to their midpoint positions, which is "5" for the texture tweakers and "0" for the EQ controls. Each tweaker then allows adjustments up and down from a Factory preset's starting point.

We've made the adjustment range of each tweaker provide plenty of range around our Factory preset values and as we've previously seen, with just six tweakers millions of possible sound textures can be created. As always, you can safely get right back to where you started out simply by recalling the original Factory preset!

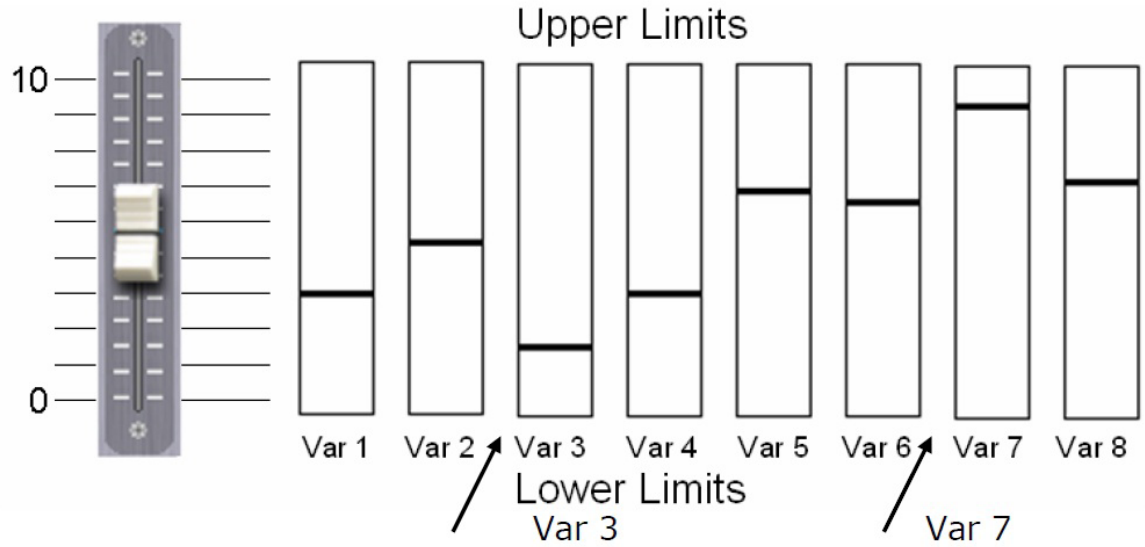
How Do the Tweakers Work?

Each Guru tweaker moves a group of controls that would normally be associated with Pro GUI. Note that the Guru controls are not new controls or even ones that are hidden behind a secret back door. Rather, they are the *same* controls that already exist in the Pro GUI version of the graphical user interface. The difference is that a Guru control can move more than one control at once while Pro GUI can't.

What Causes a Guru Control to Stop Before Reaching 0 or 10?

Because Guru is adjusting parameters that would normally be associated with Pro GUI, it's possible for one of the parameters being adjusted to encounter its normal end stop while being adjusted by a Guru tweaker.

Since we're not allowed to go past a control's natural end stop Guru must do something to let the user know that an end stop he can't see has been reached. We manage this by preventing the Guru tweaker from going any further in whatever direction is causing a parameter to reach one of its own natural limits – the same limit that Pro GUI would bump into if *it* were being used. The following graphic illustrates this.



See the vertical bars representing Var 3 and Var 7? Now imagine that a single Guru tweaker is simultaneously adjusting eight of Pro GUI's controls, Var 1 through Var 8. Remember that these controls are *not* seen by a Guru user but they *do* exist in Pro GUI. Remember also that each Var has its own pre-determined min/max limits.

Imagine now that the Guru tweaker is being moved towards 10. Notice also that the parameter assigned to Var 7 is going to reach its upper limit before the others. It might also be obvious that Var 7 might reach its limit before the Guru tweaker reaches 10… in fact the Guru tweaker might only reach 8 before Var 7 maxes out. Whenever an internal value reaches its limit the Guru control is *prevented* from going any further in *that* direction.

Now let's look at the opposite scenario where the Guru tweaker is being pulled down and Var 3, which is already nearer its lower limit, will reach that limit before any others. Guru's tweaker might only be able to get down to a setting of 3 before being prevented from going any lower.

It is important to remember this concept – If a Guru control won't go any further in one direction it's because one of the values it is modifying has reached a normal predefined limit. Each Guru tweaker can do the work of many controls, and each has enough range to provide a lot of variability from Factory presets, even when the adjustment range seems to be inexplicably restricted by control limits that are not seen by a Guru user.

The Six Tweakers

AGC Depth

Some users prefer deeper AGC action than our Factory presets provide while others prefer less. The *AGC Depth* tweaker allows this parameter to be adjusted up or down from our Factory settings.

Higher settings will tend to hold fading program levels up for longer periods of time. Likewise, lower settings leave less gain available to pull up the softer passages in programming.

Note: The VP-8IP Guru indicates up to 30dB of AGC/Compressor gain reduction while the front panel is limited to 22dB. Because there is over 48dB of distortionless gain reduction available in the algorithms users need not be overly concerned about deep amounts of AGC activity shown on the Meters.

Tech Tip: Our feed forward algorithms sound exactly the same at 10dB of gain reduction as they do at 30dB. Therefore driving the AGC harder won't make the resulting sound "denser." That task is best left to the *Compression* and *Density* controls…

Compression

While the *AGC Depth* tweaker adjusts how deep the compression is, the *Compression* tweaker adjusts how audible it is. Pushing the tweaker up to higher numbers causes an increase in the *audibility* of compression. This effect may be desirable in some formats in order to create a certain level of "excitement" on the air. Likewise reducing the setting of the *Compression* tweaker reduces the audibility of compression.

Those seeking that old 60's retro sound will find the *Compression* tweaker quite useful.

Tech Tip: The *Compression* tweaker primarily alters the difference between slower AGC and faster Compressor activity. For very dynamic material higher settings of the *Compression* tweaker can become quite obvious. The reverse is also true – very compressed source material will be affected less by the *Compression* tweaker.

Density

The *Density* tweaker operates like a power control. Higher settings increase short-term density and the average audio level which makes the audio not only louder, but also fuller. Some formats desire a denser sound in order to create a certain emotional feel for their on-air presentation.

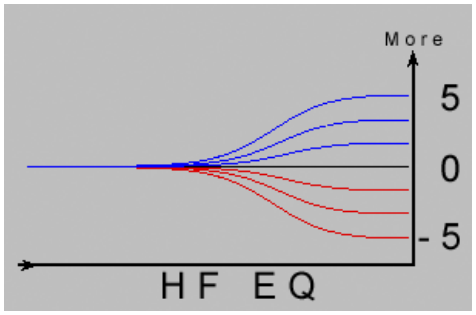
Lowering the setting of the *Density* tweaker reduces the short-term density and makes the audio sound more relaxed. A lower *Density* setting will also be less loud to the ear even though the peak levels have not changed. Loudness can be made up by using the *Loudness* tweaker described next.

Loudness

Operation of the *Loudness* tweaker will be very obvious… higher settings will be louder while lower settings will be less loud. At the upper extremes of this tweaker's settings and with some combinations of other Guru settings the loudness versus distortion "ouch" factor might be reached before the tweaker reaches its 10 setting. If it's loud enough (or too loud) on the air and is also riding the hairy edge of distortion – or is already over it – simply reduce the setting of this tweaker by a number or two.

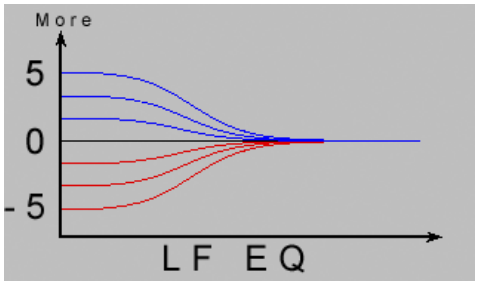
HF EQ

This control modifies the balance of high frequencies to mids and lows. The control assumes a setting of "5" when a factory preset is taken and the more the control is turned above five, the brighter the audio will sound. Conversely, the more the control is turned towards zero, the less bright the audio will sound. The control will reset back to "5" if the factory preset is retaken.



LF EQ

This control modifies the balance of low frequencies to mids and highs. The control assumes a setting of "5" when a factory preset is taken and the more the control is turned above five, the more low end the audio will have. Conversely, the more the control is turned towards zero, the less low end the audio will have. The control will reset back to "5" if the factory preset is retaken.



Guru Caveats

While Guru can't compete with Pro GUI in terms of the ability to perform deep customization of our Factory presets, it can help customers with limited audio processing experience quickly achieve their desired sound without needing to learn the interactions of literally *hundreds* of controls.

What follows is an overview of what to keep in mind about how Guru interacts with both the audio processor and the Pro version of the GUI.

1. **Guru can only modify virgin *Factory* presets**.

   Why? Guru relies on known starting points within the Factory presets before it can make modifications. The only presets that can be modified by Guru are Factory presets.

2. **Once a Factory preset has been modified by Pro GUI it cannot be modified using Guru.**

   Why? If a Factory preset has been modified by Pro GUI it is no longer a "Factory" preset and therefore it is not allowed to be modified using Guru.

3. **Pro GUI can customize presets created by Guru although the reverse is not true.**

   Why? Pro GUI has full control of all parameters regardless of how, where, and by whom they were created. Guru does not have this capability.

4. **Guru is not permitted to recall a factory preset, modify it, save it, recall it, and then modify it again.**

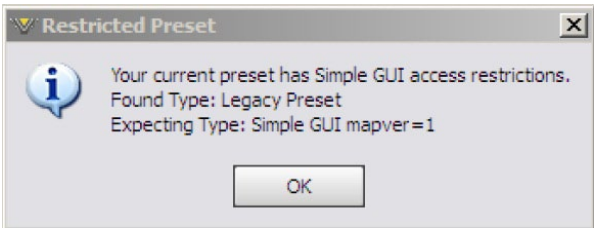
   Why? Such an exercise would likely not prove useful. If more control than Guru is required, then Pro GUI is the tool to use.

Understanding the Differences Between Factory, Legacy, and Guru Presets

- *Factory* presets are those created by the Factory using specialized tools.
- *Legacy* presets are presets that are created by a customer using the full control program, Pro GUI, or created and saved by Guru.

The biggest difference between the two types of presets is that Guru is permitted to modify *only* Factory presets.

If while using Guru you receive the "Restricted Preset" dialog, shown on the right, *and* you know for sure that the preset being adjusted is a virgin Factory preset, simply recall the preset again using Guru and the error should disappear.



The Other Two Pages of Guru...

The ability to make simplified adjustments to Factory presets in order to put the VP-8IP in service is nice, but a way to configure settings such as the Input Source and Output Levels are also needed. That's where the other two pages of Guru come in. To find all the controls in Guru simply press one of the three navigation buttons at the top of the GUI's navigation bar.

Up to now we've been discussing the *Adjust Sound* page, as that's the one that most customers will want to play with first. But there are two other pages of Guru that need some explanation too, the ones that are reached using the *Calibrate Input/Output* and *Configuration* buttons. Below is the *Calibrate Input/Output* page for the FM Mode.

VP-8IP Guru *Calibrate Input/Output* Page

In the FM Mode there are specialized controls available for setting up the Stereo Generator, SCA (for subcarriers like RDS), and the output configuration required for the particular application.

VP-8IP Guru *Calibrate Input/Output* Page

Along with Stereo Generator controls there are others for selecting the Analog or Digital input, adjusting the gains for the Analog and Digital inputs, setting the *High Pass Filter* cutoff frequency, and selecting the audio source for the headphone output.

Note that for convenience the headphone volume control appears on *all* pages of Guru and the headphone source selector is present on *both* the *Calibrate Input/Output* and *Configuration* pages.

Finally we come to the third and final page of Guru, the *Configuration* page. This is the one that, among other things, allows the VP-8IP to be switched to one of its six specialized operating modes. This page also hosts the links to the *Preset Scheduler* and *Device* configuration dialogs.

VP-8IP Guru *Configuration* Page

Device configuration was covered earlier in this chapter. Preset, or Event, Scheduling is covered in the next chapter, which describes the Pro version of the GUI.

Additional Notes on VP-8IP Network to GUI Connectivity

The VP-8IP uses both TCP port 55890 and the first available UDP port in the range of 60001 to 60010 during normal communications sessions with the remote GUI software.

The TCP protocol is used for controlling the VP-8IP and reporting its control settings back to the GUI because that protocol works to repair *all* transmission errors that might occur on the network. This ensures that the VP-8IP's controls will *always* do exactly what you tell them to do and that the control settings you see on the GUI are in 100% agreement with what the VP-8IP thinks they are.

Metering from the VP-8IP back to the GUI is a different story however. For this we chose the UDP protocol, a simple data transmission model without any hand-shaking to guarantee data reliability. UDP is used for metering because of its low overhead at both ends of the link and because we don't mind if an occasional meter data packet (or a bunch of meter packets) gets lost. Metering data is updated frequently so a few missed metering packets have an inconsequential effect on the visual appearance and no impact whatsoever on the sound.

What this means to an end user is that the VP-8IP expects to be able to use TCP port 55890 and UDP ports in the range of 60000 to 60010 on the portion of the network connecting the VP-8IP and the host PC running Guru.

The most common problem we see is UDP packets being blocked on the host network. When this happens Guru won't be able to receive metering data from the VP-8IP. In this case you'll notice that you're connected (GUI status is *Online*) and will even be able to control the VP-8IP. But you won't see any meters working.

If this happens, navigate to Guru's Configuration tab and check the box labeled *Metering Data over TCP/IP*. Using TCP for metering will increase network traffic because of the handshaking required and metering might, depending on network congestion, appear a bit jerkier than if UDP were being used.

Don't forget to check the Windows Firewall settings if the VP-8IP GUI and hardware can't connect!

Unlocking the Front Panel

There is no factory default passcode for the VP-8IP – it leaves our factory in an unlocked condition.

To unlock the front panel after it has been locked with a passcode follow the procedure outline here.

Use the front panel rotary knob to scroll through the front panel menu until the word **Access >** is displayed. Press and release the front panel knob to see the **PW:0000** prompt appear, with the first digit (the first **0**) flashing. Turn the knob to select the first digit of the passcode and press the rotary knob or the illuminated "W" button to accept this digit.

Enter the remaining three digits of the passcode in a similar manner, dialing the number then pressing to accept it. After pressing to accept the fourth digit the display will show **Unlocked** if you were successful.

Locking the panel is done by scrolling to **Access >** and then pressing and releasing the button to see **Lock...** on the display. Press again and the display will show **Lock...N** with **N** flashing. Scroll to change the **N** to a **Y** and press again. The display will briefly show **Locked** before returning to its normal showing of the current preset. Please note that the front panel cannot be locked if you have not stored a passcode.

The passcode can be changed from the front panel. Scroll to **Access >** and press, then scroll to **Edit...** and press. The display will show **Change:N** with the **N** flashing. Scroll to change the **N** to **Y** and press. The display shows **Old:0000** with the first **0** flashing. For each of the four digits in the current passcode, scroll to the correct digit and press. The display now shows **New:0000** with the first **0** flashing. Enter the four digits of the new passcode, pressing the button after each digit. After entering your new passcode the display will briefly show **Confirm** and then **New:0000** with the first **0** flashing. Enter the new passcode one more time. When done the display will briefly show **Okay** before returning to its normal showing of the current preset.

Contacting Wheatstone Technical Support

Our technical support department is available weekdays during the hours of 8:30am to 5:30pm, U.S. East coast time. Our telephone number is 252-638-7000.

Our website can be found at http://wheatstone-processing.com and provides a wealth of information about our products. User manuals, specialized presets, the very latest software, and a plethora of application notes in PDF format are available for download.

It is our goal to design and build the best sounding and most reliable audio processors available. Input and product suggestions received from our customers are taken seriously, so please feel free to email us at techsupport@wheatstone.com with your suggestions. We value your input!

Pro GUI – What It Is and How to Obtain It

The original GUI remote control software (Pro GUI) is widely acclaimed for its unrestricted access to processing controls and the quantity and quality of visual feedback provided to the end user. Unfortunately it can also be a bit overwhelming for some users (our top of the line processor has about 470 individual controls!).

Because of its complexity, requesting Pro GUI is, in effect, proclaiming your abilities as an audio processing expert. Audio processing experts understand the relevant technical terms and are able to perform deep processing adjustments on any brand of audio processor without needing advice from the experts at the factory who designed it.

Visit our website http://wheatstone-processing.com and sign up under *Home* button to receive Pro GUI at no charge. Via e-mail we'll provide you with login credentials, tell you how to download it, and turn you loose on the processor of your choice.

VP-8IP Pro GUI

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VP-8IP Pro GUI

Architecture of the VP-8IP Pro GUI

The VP-8IP Remote Pro GUI (Graphical User Interface) allows you to adjust the processor as well as view in real time how it is reacting to the audio passing through it. The VP-8IP Pro GUI has been very carefully designed to put parameters never more than two mouse clicks away. This is a radical change from competing products where constant navigation through numerous layers of menus and screens is required.

NOTE: To obtain the Pro GUI please see the note on page 3-18.

Network Configuration – VP-8IP Pro GUI

The Pro GUI PC software allows you to control any number of VP-8IP's via a standard Ethernet local area network (LAN). Utilizing TCP and UDP protocols, the Pro GUI can control VP-8IP devices from anywhere with an Internet connection, including those behind firewalls and through VPN tunnels.

Getting Started

The VP-8IP Pro GUI is a Windows™ based program designed to be intuitive and straightforward to use. It is supplied as a self-installing executable program that may be installed to any folder desired on the host computer. Use the Browse opportunity when presented during the installation process to point the installer to put the Pro GUI files in any folder desired.

If a custom program location is not entered, the resulting files will be installed under the default location for all Windows-based Pro GUI's: "\Program Files\ Vorsis\application folder name," where in this instance the application folder name is "VP8IP."

Starting with version 4.x VP-8IP presets are stored in the logged on user's documents folder instead of in the application's installed directory. This change was made necessary by Windows 7 which insists that in order to be able to ever locate them again, "user" files must be stored somewhere other than what would normally be a hidden windows system folder.

After installation, a VP8IP icon will appear in the Programs list and clicking on it will start the Pro GUI program.

To configure the VP-8IP network connection a basic familiarity with Windows navigation techniques is assumed. If terms such as "left-click," "right-click," "double-click," and "drag-and-drop" seem alien, operating the VP-8IP's Windows-based GUI's could be a bit problematic.

Connecting the VP-8IP and the Pro GUI

Using a LAN Connection

The VP-8IP and PC running the VP-8IP Pro GUI program can be straightforwardly connected together over a standard Ethernet Local Area Network (LAN).

The VP-8IP should be installed, powered-up, and verified to be operating normally. A CAT5 Ethernet cable connected to the 100baseT LAN should be inserted into the appropriate socket on the rear of the VP-8IP. Likewise, the Pro GUI should be installed on the desired PC and that PC verified to be working correctly with the LAN.

If it is desired to change the VP-8IP's TCP/IP address from the factory default, the remote Pro GUI can be used to accomplish this.

Configuring the VP-8IP IP Address*

Once the Pro GUI has been installed you must configure it and the VP-8IP so that they can communicate with each other. This requires configuration for both the VP-8IP and the remote Pro GUI so that they agree on the networking parameters.

Before starting the configuration procedure, please locate and carefully make a note of the VP-8IP's 12-digit MAC Address which is located on a label either on the top cover or the rear panel. This address will be in the format 00:50:C2:xx:xx:xx where xx:xx:xx are the digits unique to your VP-8IP. Note that you can also obtain the MAC Address from the VP-8IP front panel.

Note that if the MAC address is not entered carefully and correctly in the following steps, the VP-8IP's IP address will not be changed!

Start the VP-8IP Pro GUI software. Then right click on the control area of the Pro GUI and select Hardware/Assign IP Address. The following window will appear:

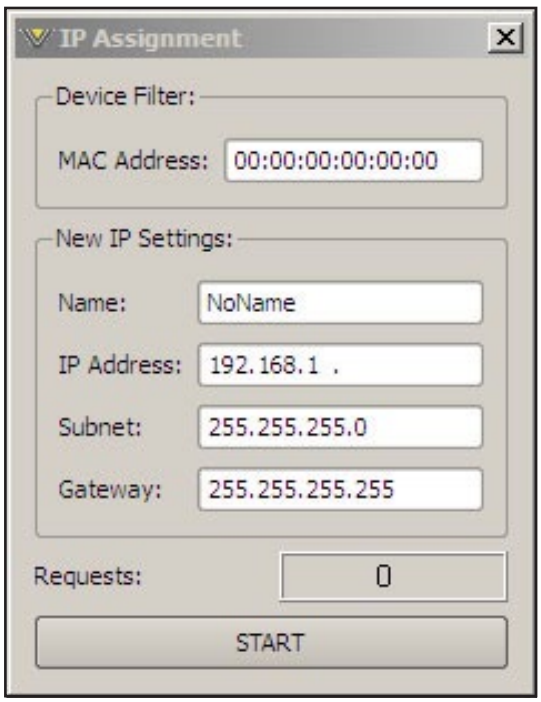


Figure 1

Next, carefully enter your VP-8IP's MAC address in the MAC Address field. The MAC address must be carefully entered because the Pro GUI will send the IP address information to the device that owns the MAC address that was entered in this step. If the MAC address is incorrect, the VP-8IP will never hear it.

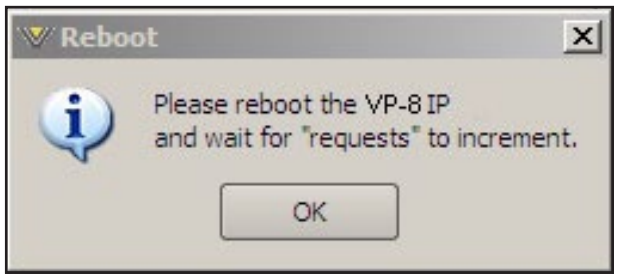
** The VP-8IP does not support DHCP (Dynamic Host Configuration Protocol) and therefore requires a static TCP/IP address on the network. Its presence on the network will not interfere with DHCP addressing of other network connected devices as long as the IP address that is configured for the VP-8IP does not conflict with the address of any other device on the network. Please consult your friendly IT manager if necessary.*

Next, enter a pet name for your VP-8IP, like *WABC VP8IP 2*, etc.

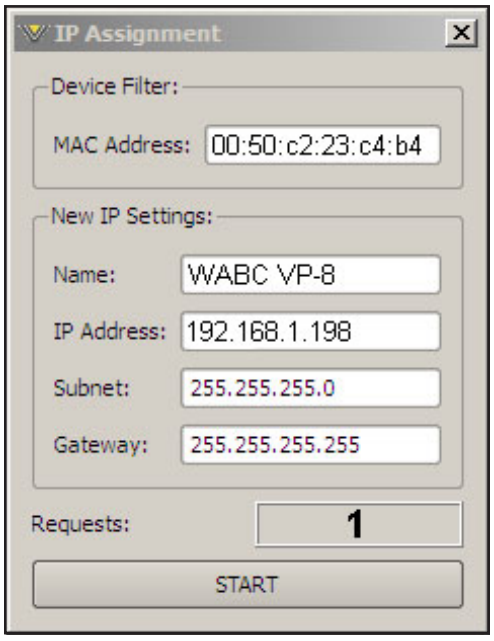
Then, enter the IP address that you wish the VP-8IP to have, noting that this address must be unique if the VP-8IP will be communicating over your network.

Next enter the desired Subnet Mask and Gateway IP addresses. In some cases (but not all) it is sufficient to use the default values as shown on the screen.

Once this is done, click the *Start* button at the bottom of the IP address configuration box. A message similar to the one below should appear:



Click OK, remove power to the VP-8IP for a few seconds, and then reapply it. As the VP-8IP is booting up, the *Requests* number in the IP Address window should increment to something other than zero (please see the bottom of the image at right). When this occurs the VP-8IP has been programmed with its new IP address and is ready for use.

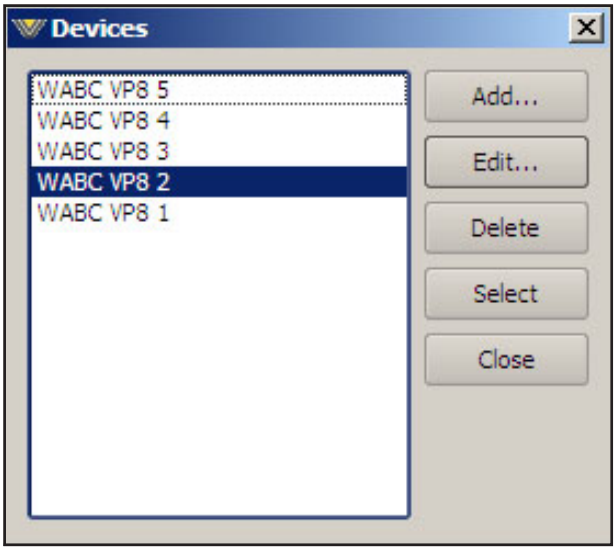


Adding Devices

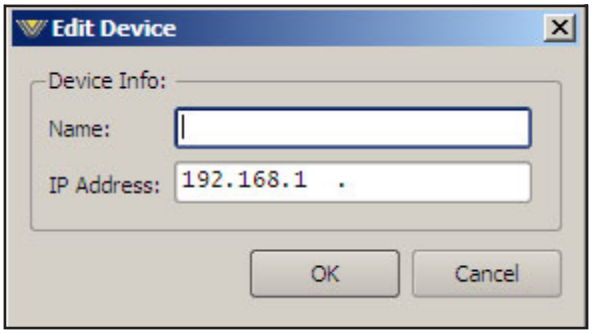
Now that the VP-8IP has its own TCP/IP identity, we must configure the Pro GUI so that it can talk to the VP-8IP. This is done by adding devices to the list of VP-8IP's that the Pro GUI knows about. To do this, locate and click on the *Devices* button that is located along the right side of the Pro GUI.

Devices Button

When this is done, the following window will pop up:



Next, click the *Add* button and the *Edit Device* dialog box will appear. This box is used for adding new devices or modifying the configuration of existing ones.



First, enter the same pet name that you used when you assigned the IP address to the VP-8IP.

Then enter the IP address that you configured the VP-8IP for in the previous steps. Note that it is always best to completely delete any existing data in the IP Address box and enter the complete address from scratch. We apologize for this inconvenience.

After this is done you may click *OK* on the *Edit Device* box. Then, in the *Devices* dialog box, highlight (single left click) the VP-8IP device that you just added and click on *Select*. This step tells the Pro GUI specifically which VP-8IP you want to connect to.

Next, click on the Online/Offline button just to the left of the *Online Status* indicator.

In the Status window you may see the message *Trying* as the Pro GUI is handshaking with the VP-8IP. Once the handshake is complete and communication protocols are locked in the Status message should revert to *Online*.

If for some reason the Pro GUI cannot connect with the VP-8IP, the *Trying* status message will remain and will occasionally blink as the Pro GUI retries the establishment of a connection. Under these circumstances the configuration of the Pro GUI and VP-8IP should be carefully examined to ensure that the destination TCP/IP address is consistent between the two.

Once the Pro GUI has been made aware of the VP-8IP's existence in this manner, it will always appear in the list of VP-8IP *Devices*, and be instantly accessible, and from anywhere.

VP-8IP Pro GUI Status Bar

When connected to a VP-8IP the status bar will indicate *Online* and the *Devices* text will show the name of the VP-8IP that it is currently communicating with. The status bar will also display the name of the preset that is running in the VP-8IP you're connected to.

Connecting Directly Without a LAN

It is also possible to work without a LAN by connecting the VP-8IP and the PC Ethernet ports together using a standard Ethernet cable; either a straight through or crossover cable may be used since the 10/100 Ethernet switch inside the VP-8IP has auto-sensing ports. The controlling PC and the VP-8IP's network settings should be configured to place them both on the same subnet!

PC System Requirements

Constraints for the system are not extraordinary – the PC should be preferably at least 1GHz in speed, running at least Windows 2000 or Windows XP, and the screen resolution should be at least 1024 x 768 pixels in order to get the full benefit of the high resolution graphics utilized. There should be at least 15MB of free space on the hard drive for the Pro GUI software and presets, and the PC's LAN interface (NIC) should be 100baseT (100MHz) capable for best performance.

Wireless Connection

Many users will want to make adjustments to the VP-8IP while in a remote location such as their home, office, or car. This can be accomplished in a variety of ways.

The Pro GUI can be run on a portable computer connected to theVP-8IP's LAN by a wireless (802.11b as a minimum) device, as long as the VP-8IP itself "sees" a 100baseT network. This can be done by connecting the VP-8IP to a readily available Wireless Access Point and then connecting to the VP-8IP through the access point from a wireless interface on a laptop computer.

Be aware that a fairly high quality connection is required in order to have full advantage of the real-time graphics displays on the remote Pro GUI. Low speed connections will still function, but graphics performance, especially when displaying FFT information, may be degraded.

The Really Remote Connection!

One of our customers routinely adjusts his processor from the comfort of his company car. In order to accomplish this he connects it to his station's business LAN router, maps the unit's IP address over to a public IP address via NAT (Network Address Translation), and creates a hole in the firewall for this IP address. This gives the processor Internet connectivity, albeit to a very select few.

Then from the laptop in his car he connects to the Internet using his Verizon wireless card, fires up the Pro GUI, connects to the IP address that is running in the box (NAT takes care of the IP address differences), and *voila*! He can adjust his VP-8IP from anywhere.

We will leave the myriad other ways in which the VP-8IP can be remotely controlled wirelessly up to your imagination and the rapid changes in wireless technology!

VPN

Another way to connect remotely to the VP-8IP is via VPN. A VPN, or Virtual Private Network, is a private data tunnel between computer devices. It behaves just as if the local and remote computers were sitting beside each other – devices connected via VPN act like they are on one big LAN regardless of where they are physically located. VPN tunnels can be configured both with and without data encryption. Setting up a VPN is not difficult, but certainly beyond the scope of this manual. Although we have not specifically tested the VP-8IP with VPN connections there is no technical reason why it would not work.

The VP-8IP and Internet Security Concerns

The VP-8IP does not act as a web device or web server, nor does it support open ports to the Internet. Therefore there is no worry that someone could hack into the VP-8IP and use it as a pathway to the rest of the network to which it is connected.

About DHCP and the VP-8IP

The VP-8IP does not utilize DHCP – Dynamic Host Configuration Protocol (or automatically assigned network addressing) – and needs to be assigned a Static network-unique IP address.

Many DHCP servers assign addresses starting at the bottom of the group 192.168.x.xxx. Therefore choosing an address high in that group, say 192.168.0.200, as a static IP address will likely keep it out of the way of the busiest DHCP addressing. However, it's always wise to check with your local friendly network administrator to determine what static IP address is open and safe to use.

Sometimes the local network warrants the Subnet and Gateway values to be vastly different, but our suggested starting values of 255.255.255.0 for the Subnet and 255.255.255.255 for the Gateway should suffice for all but the most complex situations.

About Ports and the VP-8IP

The VP-8IP uses a variety of TCP and UDP ports in order to communicate with the remote Pro GUI. The actual TCP port used depends on the processor model. However, in each case the processor is the server and the PC is the client. Wheatstone processors can accept up to four simultaneous GUI connections to host PC's.

The VP-8IP uses TCP port 55890 for commands (control adjustments, etc.) between the host PC and the processor. Metering data does not go over TCP.

Once a TCP connection is established the GUI opens a UDP port and asks the VP-8IP to send meter data on that port. The port is typically 60000, but not always. With UDP ports only one application on a PC may open that port at a time, and since we do not know what other applications are running on the PC and we also want to allow more than one GUI to run at a time on any given PC, the GUI will attempt to open port UDP 60000.

If that port cannot be opened, the GUI will then try port 60001, then 60002, etc., up to 60010. Whichever port successfully opens first, that port number will be sent to the VP-8IP and the VP-8IP will begin to stream meter data over that UDP port back to the GUI.

Note: if there are two or more GUIs attached to the VP-8IP then the VP-8IP will stream the meter data twice – once for each of the opened port(s) to each GUI.

If the GUI fails to open any of the 10 UDP ports it will still run and be able to control the VP-8IP via the TCP port, but it will not receive any meter data to display. Therefore if you have a connection that appears to control the processor just fine but there is no metering displayed on the GUI, check to ensure that there is no firewall or other blockage to UDP ports 60000 through 60010.

The system menu provides an option to use the TCP Protocol for metering data. This option can be used when the network path between the VP-8IP and Host PC is blocking UDP traffic.

Using the VP-8IP Pro GUI

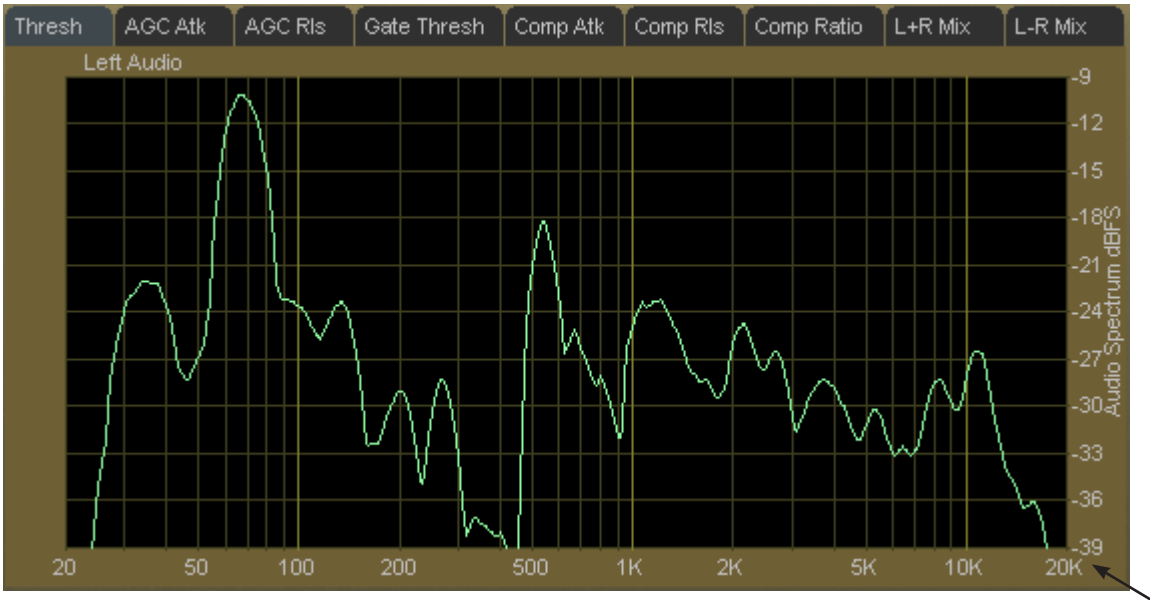
The Pro GUI may be positioned on the host computer's screen by left-clicking and holding the top edge of the Title Bar and dragging the Pro GUI to the desired position. In true Windows fashion, the Pro GUI may be minimized on the desktop (taken off the screen) and/or closed by way of the familiar controls at the extreme top right.

Descriptions of the features and functions of the Pro GUI will follow different discrete regions of the screen; it will begin with the most eye-catching region – the Dynamic Displays – and then we'll explain the Control Area above it, followed by the Side Bar buttons, and lastly the Title Bar.

Dynamic Displays Region

Frequency-Domain Graph

Visually the most important element on the main screen, and in fact dominating the lower half of it, is a large graphical display depicting the standard audio frequency range of 20Hz – 20kHz arranged in logarithmic fashion with gradations at selected frequencies. The vertical axis scale changes according to its context, however the overall vertical range remains at 30dB with gradations every 3dB.



The scale may be changed by dragging it with the mouse.

VP-8IP Audio Spectrum Display

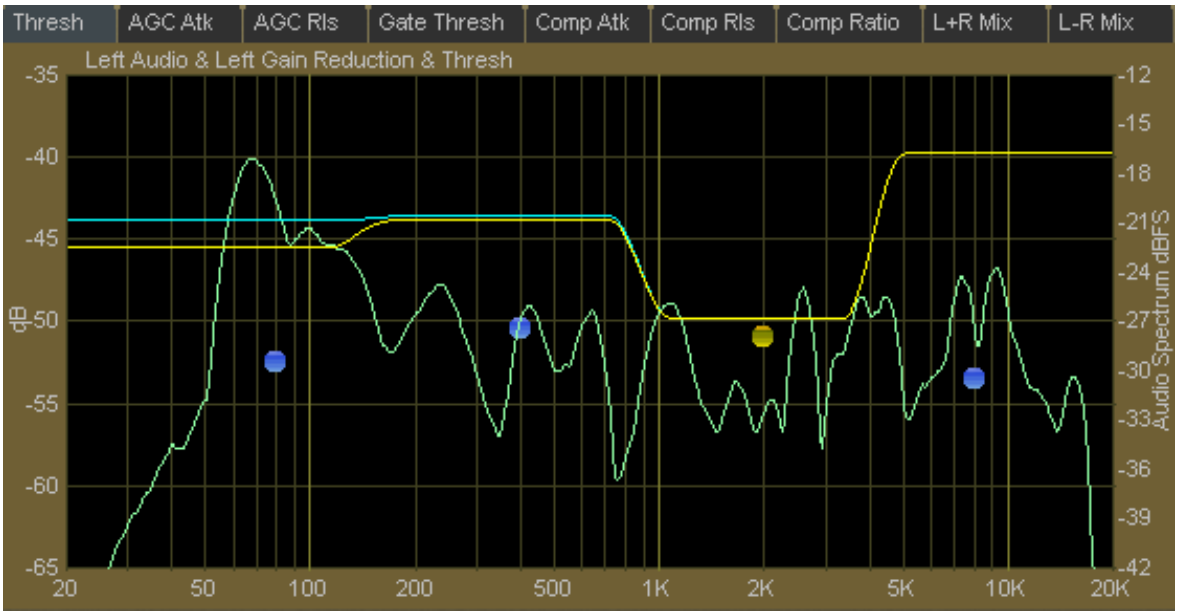
This display appears when *Audio* is checked and *Gain Reduction* and *Controls* are unchecked.

A number of real-time dynamically varying spectra and other frequency-response-versus-time data may be observed simultaneously. A row of large buttons under the graph determines what data is displayed, and there are also buttons to enable/disable the display of equalizer frequency response curves, filter responses, etc.

The display range of the display may be changed by dragging the right-hand scale up or down as desired.

VP-8IP Graphical Interface Operation

All operating parameters of the four-band AGC and the eight-band limiter can be individually adjusted by manipulating the positions of various blue dots on that sections' graphical screen. Each of the blue dots on the screen represents the center frequency of a band. By double clicking a dot (to highlight it to turn it orange) and then dragging it with the mouse, various operating parameters for each band can be customized.



Example Graphical Control Screen

There are other ways to set parameters, too. In fact all of the controls on the graphical screen work the same way regardless of which processing section owns them. Parameters within each of the processing sections can be accessed by clicking on the associated tab at the top of the screen.

By right clicking on the graphical display area with your mouse the following options will appear:

First, a short tutorial....

By default, all adjusters (the blue dots) are Grouped. That is they are coupled to each other so that if you grab one with the mouse (single left click and hold) and then slide the mouse up and down, all blue points move up and down together.

If you double click a blue dot, it turns orange to indicate that it has been selected, after which it may be moved individually to anywhere along the vertical scale.

If you now double click another blue dot somewhere else on the line, it too will turn orange....

Now if you move the mouse to a point *between* the two dots selected in the above steps, right click the mouse and select *UnGroup Adjusters*, all the points in between the first two dots you selected will now also turn orange. At this point all of the orange dots between the first two can be *individually* adjusted up or down.

If you now place the mouse over the same graphical area, right click again and this time select *Group Adjusters* the points between the first two that you highlighted will turn back to blue. If you then grab any blue dot between these points with the mouse (left click and drag) you can move the entire line connecting the two highlighted end dots up and down, with the two end points remaining anchored in place. Cool, eh?

Right clicking on the graphical area and selecting *Group All Adjusters* removes all highlighted dots, turning all orange dots back to blue. Conversely, right clicking and selecting *UnGroup All Adjusters* turns all (remaining) blue points orange, meaning they can now each be individually adjusted.

The *Flatten* function, although next in the list, is best explained last....

Okay, highlight a single dot again...any dot will do. Then right click the graphical area and select *Proportional Drag*. Now left click the line to the left or right of the dot you highlighted and drag it up or down. See how the line behaves? It pivots at the point where you highlighted the dot, and bends at the point where you grabbed it with the mouse. Virtually ANY curve can be created this way.

Now right click again and this time select *Flat Drag*. Then grab the line again... anywhere (left click and drag)... and move it up and down. See how the line pivots at the highlighted point but remains flat above or below it as you raise and lower the line?

So now, after these exercises the line is all messed up, and you want to put it back to where it was or at least make it all flat again without having to move every single point back. But how do you do that? Easy.....

Right click on the graphical area again and select *Flatten*. If there were any points still highlighted orange you'll want to first click *Group All Adjusters* before you select *Flatten* in order to turn them off. When you finally click *Flatten* the line will return back to flat and you may move it up and down again with the mouse, this time as a whole.

To review:

Group Adjusters – Gangs the **blue** dots *between* two highlighted **orange** points into a commonly behaving entity.

UnGroup Adjusters – Disconnects any **blue** dots that were grouped by "Group Adjusters."

Group All Adjusters – Gangs all the dots, turning them **blue**, *regardless* of their previous highlighting.

UnGroup All Adjusters – Ungangs *all* the dots, turning them **orange**, regardless of any previous highlighting.

Flatten – Causes adjacent dots to align with each other into a flat line.

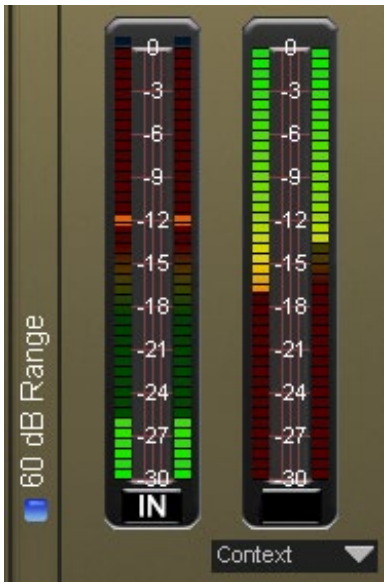
Proportional Drag – Allows the dragging of dots to create slopes.

Flat Drag – Allows the dragging of dots together in a flat orientation.

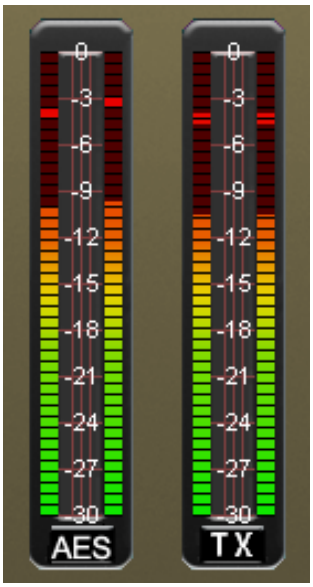
Bargraph Metering

Adjacent to the graph and to the left and right, and sharing a 30dB vertical range (0-30dB of gain reduction, 30dB signal level meter range), are a group of vertical bargraph level indicators for signal levels and gain reduction.

The level meters are peak-over-average types, with peaks riding as a lone dot over a solid bar-graphed average. An Over indication is at the top (but slightly distanced from) the input level bargraphs.



Left of Graph
Left pair - Input Signal Levels
Right pair - Selectable - see below



Right of Graph
Left pair - AES Outputs
Right pair - TX Outputs

Selectable Meter (Left of Graph)

The actual function of this meter is selected in the drop down list below it. You can select one of the self-explanatory functions, or allow the meter function to change depending upon the processing section being viewed in the Control Area Region by selecting *Context* from the list.

AES And TX Output Meters (Right of Graph)

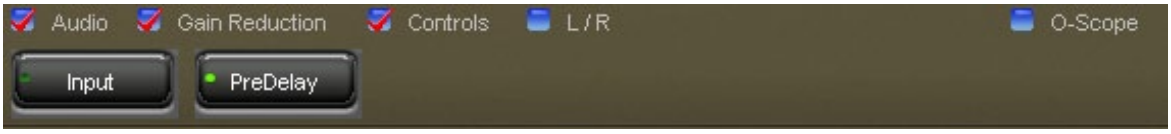
The leftmost pair of the two vertical bargraph meters to the right of the graphical display window shows the output level at the AES digital output while the rightmost pair shows the TX output level.

60dB Range

To the left of the Input meter screen is a check box labeled *60 dB Range*; when checked, it toggles the Input meter from the normal 30dB view range to a 60dB range. The latter range is useful for finding signals during initial set-up, or for observing program material when the signal source is of high dynamic range, such as live classical music.

Display Signal Selection

Beneath the graph area are two large buttons: Input and PreDelay.



By single clicking on a button a green indicator on the button will light to indicate which signal is being viewed on the large graphical display above it.

Input

Pressing this button enables a Fast-Fourier Transform (FFT) based real-time spectral analysis of the selected (Left/Right) input signal. An FFT is a mathematical filtering process which calculates the spectral content of a time domain signal.

The FFT[1] is widely used in test equipment in the audio industry because of its extremely precise measurement capability. Thanks to the extremely efficient DSP chips utilized in the VP-8IP, we are able to offer the same exotic measurement capability in an audio processor. Thus what you see on our graphics displays is as precise as current technology permits.

[1]The Fourier analysis is named after French mathematician and physicist Jean Baptiste Joseph Fourier, who first explained it. For more information on the FFT and some of its capabilities, please consult the "Handbook for Sound Engineers" by Glen Ballou and published by the Howard W. Sams Company (ISBN 0-672-21983-2).

PreDelay

Pressing this button enables the viewing of the output of the FM processing chain prior to the diversity delay.

Just above the two large buttons, and below the Graphical display, is a row of several check boxes. We will describe these in order, starting with the leftmost one.

Audio

When this box is checked and the O-Scope box (on the right) is unchecked, the graphical display will show an FFT of the audio. The displayed range may be modified by left clicking on the scale on the right-hand side of the graphic and sliding it up and down with the mouse. The displayed dynamics range is always 30dB.

- When the Input button is illuminated, the FFT shows the audio spectrum of the Input audio.
- When the PreDelay button is illuminated, the FFT shows the audio spectrum of the output of the processing chain prior to the diversity delay.

Gain Reduction

When this box is checked, the graphical display includes information related to the amount of gain reduction occurring in the AGC and compressor. The Blue line indicates long-term AGC activity while the Yellow line indicates the shorter-term compression.

Controls

When this box is checked, and if there are controls available for the page being viewed, they will be shown as blue (or red if selected) dots within the graphical display window.

L/R (Left/Right)

When this box is checked the right channel of the audio is displayed on the graphical display according to the selection made by the Audio check box. Alternately, when L/R is *not* checked, the left channel is displayed.

O-Scope

When this box is checked *and* the Audio box is also checked, an oscilloscope-like display of the audio waveform is displayed. Which waveform is being displayed at any instant is a function of the selection made under the *Audio* section explained above.

Control Area Region

This is the large enclosed area directly above the graph and meters.

All of the knobs, faders, switches, and check boxes affecting the audio processing itself are within this area. To operate them, either:

- Place the cursor over the desired control, and use the mouse's scroll wheel to increase or decrease the control's position, or

- Click on the control, and slide the mouse to move the control – up or right increases the control's indicated position, down or left decreases it. When using the mouse in this fashion (as opposed to using the scroll wheel), the resolution of the control variation can be increased – made finer – by holding down the keyboard's Ctrl key while adjusting the control as above, or

- In most cases you can double click on the text that indicates the setting value (for example, the window that reads X.XdB under the *Line Gain* knob on the *Input Section*) and a box will pop up where you can enter an exact value and click *OK*.

From left-to-right across the upper portion of the control area are a series of buttons corresponding to logically grouped processing sections in general input-to-output signal flow fashion.

A small green indicator at the left end of each button indicates if any signal processing within that block is active. Left-clicking a button opens the associated control panel to gain access to the controls for that block of signal-processing.

INPUT MENU

Input Settings

This is the first tab on the top far left of the Pro GUI. (Note: the screen that is currently being displayed is indicated by a green background on the selection tab).

Input Screen

The Input screen contains controls for the input source selection: *Input Failsafe*, Analog and AES input gain controls, input L/R gain balance, *Phase Rotator*, and the high pass filters which may be operated in Stereo or Sum/Difference mode.

Input Source

The Input Source has three choices, one is analog and two are digital. The analog choice is Left/Right, and the two digital choices are AES/EBU (AES3) or WheatNet-IP (WNIP). Note that the Automatic input Failover operates between analog and digital choices only (digital to digital is not supported)..

The currently selected input is shown by a red checkmark.

Input Signal Presence

Below the source selection check boxes are a pair of signal presence indicators. The Analog indicator lights if there is a signal present at the analog input that exceeds the threshold of -48dBFS. The AES indicator lights if there is a valid AES signal present at the digital input that is above -48dBFS.

Automatic Input Failover

In the event the currently selected input source fails and the *Automatic Input Failover* check box is checked, an alternate source of audio connected to the other input can automatically be put on the air.

- If the primary audio source was digital (AES3 or WNIP) the analog input will be selected immediately if the input was AES3 and after a 30 second silence sense timeout with audio below -48dBFS if the input was WNIP.
- If the primary audio source was analog, then a silence sense timeout of 30 seconds must elapse with audio below -48dBFS before failover to the digital inputs will occur.

Analog Gain

The Analog audio input gain can be adjusted by clicking on the knob and dragging the mouse or touchpad to the right or left. The *Analog Gain* level can be adjusted over a +12dB/-36dB range.

AES Gain

The Digital audio input gain can be adjusted by clicking on the knob and dragging the mouse or touchpad to the right or left. The *AES Gain* control affects both AES3 and WheatNet-IP and can be adjusted over a +12dB/-36dB range.

L/R Balance

Static level errors in Left/Right channel balance can be corrected by using the *L/R Balance* knob. The Left/Right balance can be adjusted over a +/- 12dB range. This control should only be used as a short term "Band-Aid." The relationship between left and right should be corrected upstream if it is found the audio is out of balance.

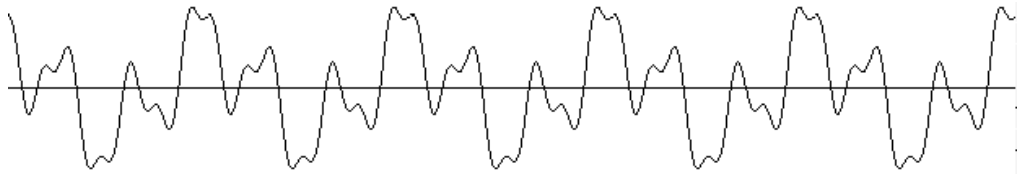
Phase Rotator

Human voice is asymmetrical in nature, which means that it is usually "peakier" in one polarity than the other. The algorithm fixes these asymmetrical peaks by phase shifting the signal harmonics relative to their fundamentals. This reduces the peak to average ratio and enables an increase in apparent loudness with minimal audible detriment. Processing the audio for a symmetrical medium like FM is greatly simplified if the waveforms are symmetrical.

Ordinarily, human voice looks like the waveform below. Notice how the width and amplitude of the upper and lower signal peaks are different? This is asymmetry, and makes the processing of voice waveforms more difficult.

Human voice before *Phase Rotator* algorithm

After passing through the VP-8IP Phase Rotator the same voice appears like the waveform below. Notice how the upper and lower portions of the waveform now have equal distribution?



Human voice after *Phase Rotator* algorithm

No information is lost and no harmonic or intermodulation distortion is created by the Phase Rotator process. Only the signal's harmonic phase relationships are rearranged to make the waveform energy more uniform in polarity.

HPF - High Pass Filter

The *High Pass Filter* is used to remove undesired subsonic energy from the audio signal prior to being processed. By removing this energy, processing is cleaner and modulation energy is not wasted by transmitting sounds that would not be perceptible to a listener.

The *High Pass Filter* also removes subsonic energy that could upset the operation of equipment later in the audio chain, such as the AFC loop in the FM exciter.

The *High Pass Filter* has three operating modes:

Off – No *High Pass Filter* boxes are checked. The High Pass Filter is not enabled.

Stereo – If *Stereo* is checked the High Pass Filter operates on both left and right audio channels equally by separate high pass filters operating at the frequency set by the single frequency tuning control. In this mode the *Diff Freq* slider is not visible.

Sum/Diff – *Stereo* is not checked, opening a second tuning slider to reveal adjustments for the stereo difference channel. Separate check boxes, one for the *Sum Freq* slider and one for the *Diff Freq* slider are visible. Checking a box activates a high pass filter on that channel, sum or difference (L+R or L-R). Separate frequency sliders are available for adjusting the high pass filter corner frequency for each signal path. The mono (L+R) and stereo (L-R) signal paths may have different high pass filter cutoff frequencies.

Why the Sum and Difference Option?

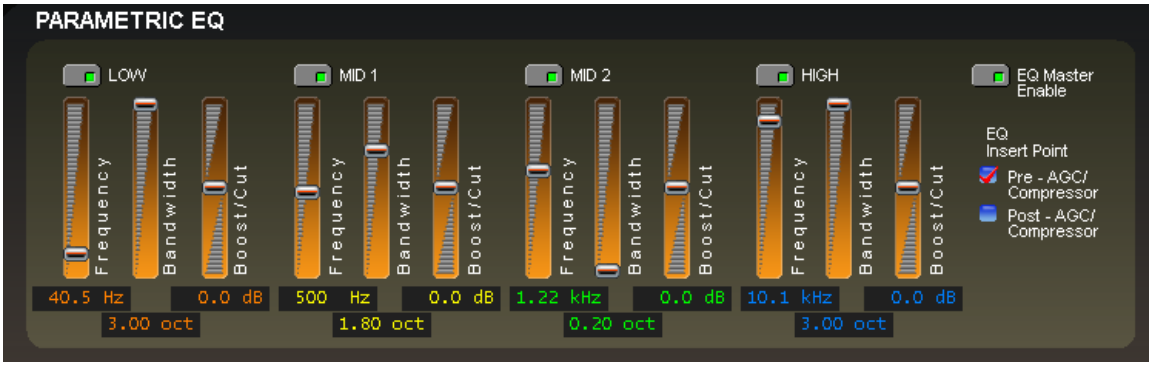
In most program material there is little very low frequency energy in the difference (L-R) signal. In fact, what energy is there is not typically correlated with the program material. Therefore the difference channel can operate with a much higher high pass filter cutoff frequency than the sum channel, and in fact 180Hz is a good tradeoff setting for the majority of program material.

The sum (mono) channel, on the other hand, carries most of the bass energy in the majority of program material and therefore it cannot utilize as high a cutoff frequency as the difference channel without seriously impacting the bass feel of the program.

A good starting setting when using Stereo mode is 30Hz. Good starting settings when using Sum/Diff mode is 30Hz for Sum and 180Hz for Diff.

Both HPF controls operate over a range of 20Hz to 300Hz.

PARAMETRIC EQUALIZER MENU



Parametric Equalizer Screen

This powerful parametric equalizer has four identical and independent bands. Each of the four bands can be switched in or out independently, and is fully adjustable in three ways:

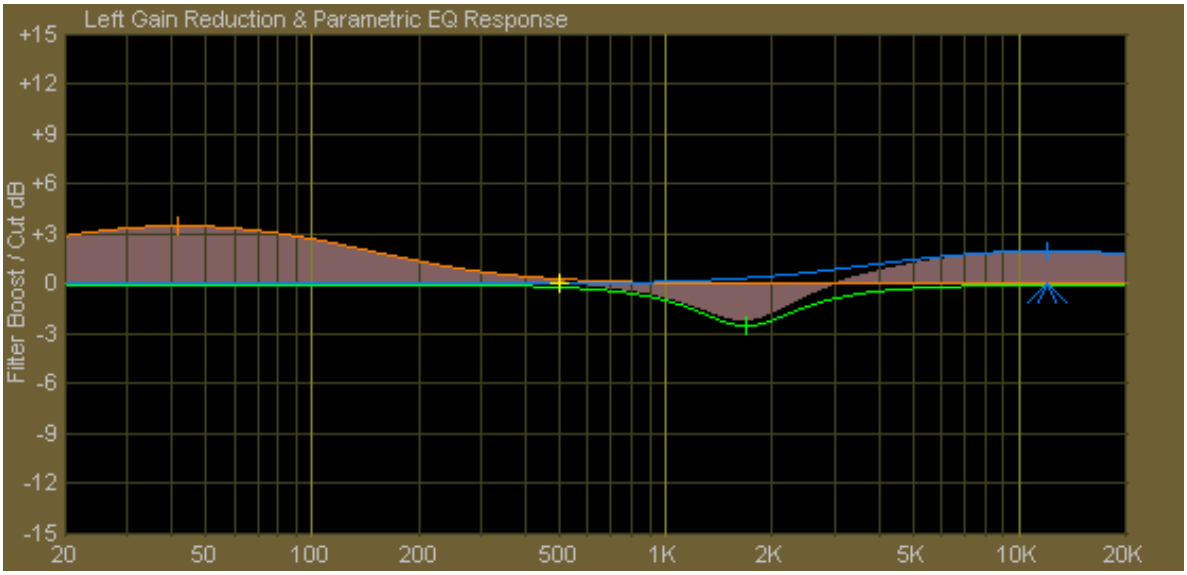
Center Frequency	—	20 Hz to 20 kHz.
Bandwidth	—	0.2 to 3.0 octaves.
Boost/Cut	—	+/- 14.0dB.

The position of the parametric equalizer in the signal chain can be selected to be prior to the multiband AGC/Compressor, or after it. The effect created by inserting the equalizer into each position is different.

When the parametric equalizer is used ahead of the multiband AGC the equalization curves “pre-load” the AGC drive to make it operate in a frequency conscious way. Be aware however that EQ changes made to the audio – even dramatic changes – might be reduced when the audio goes through the multiband compressor, especially when operating with ratios greater than about 4:1.

The controls of the equalizer can be manipulated in two ways. The first is by directly clicking on and dragging the controls in the parametric equalizer control screen. As the controls are manipulated, numerical representations of the control settings appear in the boxes below the band that is being adjusted.

The second method of adjusting the parametric equalizers is by directly manipulating the actual curves on the graphical screen below using the mouse.



Parametric Equalizer Display
(the shaded area depicts band that is "on")

Double clicking on the little crosshair at the top (or bottom if it's below the line) of the curve turns that section on and off.

Clicking on the crosshair and holding the left mouse button down as you drag the mouse moves the curve to any frequency (left or right move) as well as adjusts the curve's height above or below the 0dB reference line (up or down move).

Clicking on the little "tent shaped" symbol underneath, and holding the left mouse button down, and sliding the mouse left or right allows you to change the bandwidth, or Q, of the equalizer section.

Right clicking on the graphic opens a window for manual selection of which curve is to be manipulated with the mouse.

EQ Master Enable

This is the master Equalizer In/Out button. When it is illuminated, the equalizer section is in, even if no other section of the equalizer has been enabled by clicking the *LOW*, *MID 1*, *MID 2*, or *HIGH* buttons.

AUTOMATIC GAIN CONTROL and COMPRESSOR MENU

The VP-8IP AGC and Compressor sections are inextricably linked by DSP algorithms that logically connect them. Therefore they will be covered collectively in one chapter rather than discussed separately.

Although the *AGC/Compressor* section's outputs may be separately enabled or disabled, the Compressor algorithm is always running so that the desired derivative control signals are available to the AGC section.

Four Band AGC/Compressor Screen

The automatic gain control (AGC) operates in four frequency bands and its long-term operation is a special derivative of the type of program-related processing occurring in the Compressor algorithms. The *AGC/Compressor* operates in Sum and Difference mode – the highest signal, whether in the sum or difference channel, determines the amount of processing to be applied to both channels.

AGC Drive

The AGC *Drive* control adjusts the amount of signal into the *AGC/Compressor*, primarily determining the amount of compression that can be achieved. Driving harder (higher numbers) results in a more consistent on air sound, with an increased capability to pull up low passages in program material. Less drive creates a more gentle on air sound and has less capability to bring up low passages. The *Drive* control simultaneously adjusts the sum and difference channels. The range of this control is from -79.0dB to +6.0dB, or OFF when fully down.

Normal *Drive* levels achieve between 12dB and 18dB of gain reduction in band 2.

AGC Output

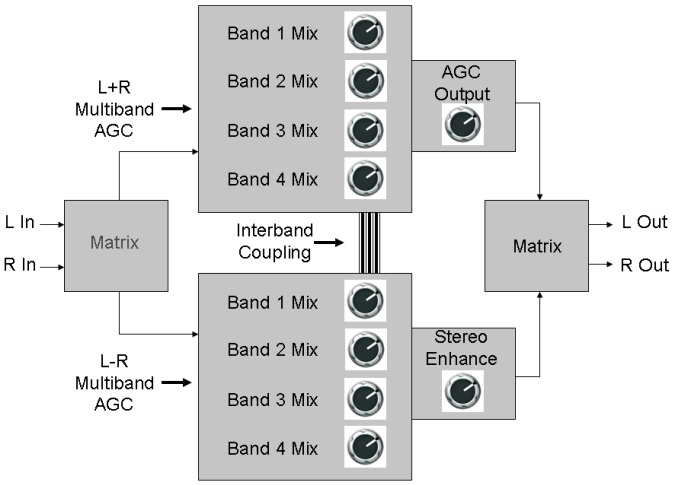
The control adjusts the overall output level of the *AGC/Compressor* into the following stages. It simultaneously adjusts the Sum and Difference channels. The range is from -20.0dB to +48.0dB.

Because the *AGC/Compressor* is a feed-forward design, the loss of signal amplitude created while the input signal undergoes compression must be "made up" afterwards in order to return it to a normal level. This is why the *AGC Output* control has such a wide range.

Stereo Enhancement

The VP-8IP is equipped with a flexible stereo enhancement scheme allowing the air sound to be fully customized. There are two sets of controls involved in the stereo enhancement algorithm; the main Stereo Enhancement control and the five multiband AGC mix controls for the L-R signal.

The main Stereo Enhancement control adjusts the mix level of the AGC's L-R processing after the four individual band mix controls. The individual AGC L-R mix controls adjust the level prior to the main Stereo Enhance control. Why is this needed? At times it can be desirable to modify how the L-R signal sounds to a listener. For instance, female-oriented formats might prefer a somewhat softer sound in order to enhance long term listening so being able to slightly reduce the L-R levels from band 3 and 4 can make the sound more "mellow." The diagram on right shows the basic signal flow of the stereo enhancer algorithm.



AGC Enable/Comp. Enable Buttons

These buttons determine the presence of the AGC section and the Compressor section in the output.

AGC Enable – When illuminated by clicking on it, indicates that the output of the AGC section is in the audio path.

Comp. Enable – When illuminated by clicking on it, indicates that the output of the Compressor section is in the audio path.

Compressor Out / AGC Out – When both AGC and Compressor are switched off, signal is being bypassed around the Compressor and AGC sections with no gain control being performed.

Compressor In / AGC Out – Signal is being routed through the Compressor only, which operates with relatively fast time constants. The sound of the processing will be quite obvious on most program material.

Compressor Out / AGC In – Signal is being routed through the AGC only, which operates with relatively slow time constants. The sound of the processing will be quite invisible on most program material but very short term variations in level will not be corrected.

Compressor In / AGC In – Signal is being routed through the AGC and Compressor, which together operate with different time constants. The sound of the processing will be quite smooth with both short and long term variations in level being corrected (also see *AGC Backoff*).

Gating and AGC Backoff

Gating Threshold

The *Gating Threshold* is the audio level in dBFS, at which the gain reduction release is modified from simple attack and release to a modified release (see explanation for *Gain Freeze* and *Slow Recovery* below). The control has a range from -79.0dB to -20.0dB, or OFF when fully down.

On each of the four bands there is also a Gate Offset trim control allowing offsets of +/-6.0dB to be made to each band referenced to the main *Gating Threshold* setting. This allows precise *Gating Thresholds* to be assigned to each band as appropriate for the program format requirements.

Gating Delay

Gating Delay is how long the gating circuit waits after the signal has fallen below the *Gating Threshold* before it actually modifies the release time according to the *Gating Option* setting. The *Gating Delay* is adjustable from 50.0 milliseconds to 500.0 milliseconds, with values on the order of 50.0 milliseconds to 200.0 milliseconds being useful for typical programming environments.

When *Gating Delay* is adjusted for longer periods of time beyond about 100.0 milliseconds, partial release is afforded when the input signal falls below the actual *Gating Threshold*. This can be useful for maintaining a more natural drop in signal level during program fades.

Gain Freeze

When *Gain Freeze* is selected and the audio falls below the *Gating Threshold*, the gains of the four bands will freeze at their current values and will hold at those values until audio is present that is above the *Gating Threshold*. This prevents a gain increase in the absence of audio which would likely increase background noise.

Note that the time between achieving the *Gating Threshold* and the actual gain *Gain Freeze* is determined by the setting of the *Gating Delay* control.

Slow Recovery

When *Slow Recovery* is selected and the audio input falls below the *Gating Threshold*, rather than the gains of the bands holding as above, they will slowly return towards 0dB. This setting may be useful for formats that contain material with very wide dynamic range or for certain speech-based formats where the *Gain Freeze* setting may not be appropriate.

Note that the time between achieving the *Gating Threshold* and the actual gain transition to *Slow Recovery* is determined by the setting of the *Gating Delay* control.

The controls associated with the Gate algorithm are quite subjective and therefore should be adjusted entirely by the requirements of the program format. There is no right

or wrong setting of the Gate controls unless that setting causes undesired, unexpected, or unwanted behavior of the processor during low level program passages.

InterBand Coupling

The VP-8IP utilizes interband coupling to ensure that the audio spectrum remains well controlled under all programming conditions. What this means is that the gains of bands 1, 2, and 4 can never be greater than the difference between the gain in band 3 and the setting of the *InterBand Coupling* control, which ranges from -30.0dB to 0.0dB.

Example: Suppose that the *InterBand Coupling* control has been set to a setting of -3dB. Under this condition bands 1, 2, and 4 can never have a gain that is *more* than 3dB greater than the gain in band 3.

Another example: Suppose that the *InterBand Coupling* control is set to -5dB. If band 4 now becomes un-gated because its level is sufficient to do so and band 3 is busy enough to be requiring 12dB of gain reduction, band 4 *cannot* have a gain that is more than 5dB *higher* than the gain in band 3, which means that it's gain cannot increase beyond -7dB (12dB gain reduction in band 3 minus the 5dB from the *InterBand Coupling* control setting).

The *InterBand Coupling* control works in *one* direction only and *does not* affect a band's ability to *reduce* its gain when appropriate.

AGC Backoff

The *AGC Backoff* adjusts the processing balance between slower AGC action and faster compression. Think of it as the "density" control.

With the *AGC Backoff* control set at 0, the audio gain control is primarily due to slower AGC action, and therefore short term program dynamics are largely unaffected unless they exceed the AGC gain reduction by at least 1dB. In that case momentary gain reduction by the compressor will place the gain where it needs to be to return control to the AGC.

With the *AGC Backoff* control at the other extreme, or -12dB, gain control is primarily due to fast compression and short term dynamics will be aggressively controlled.

The setting of the *AGC Backoff* control allows the audio compression texture to be modified from a gentle, almost unobtrusive control (AGC Backoff between "0" and about minus 3.0dB), to very obvious compression (AGC Backoff control at "-4.0" or greater negative numbers) creating a sound that is similar to vintage compressors and limiters from the 1960's.

Note that regardless of the setting of the *AGC Backoff* control, the AGC and Compressor stages operate with their algorithms linked mathematically and there is no method for changing the relative thresholds between the AGC and compressors. If for some reason less AGC control signal is desired, it may be achieved by lengthening the AGC attack time to large or very large values.

Crossover Frequencies

The AGC/Compressor is a four band design utilizing 48dB/octave phase aligned filters. Careful filter design eliminates unnatural spectral coloration when deep gain reduction is occurring in one or a few bands.

There are three crossover frequencies to select.

- The Low to Low Mid (*Band 1 > Band 2*) crossover frequency is selected using the red list box. Frequencies between 50Hz and 198Hz may be selected.
- The Low Mid to High Mid (*Band 2 > Band 3*) crossover frequency is selected using the yellow list box. Frequencies between 250Hz and 1.50kHz may be selected.
- The High Mid to High (*Band 3 > Band 4*) crossover frequency is selected using the green list box. Frequencies between 3.00kHz and 8.00kHz may be selected.

Selecting Crossover Frequencies

The selection of crossover frequencies plays an important role in determining how an audio processor sounds with different types of program material and therefore the crossover frequencies should be selected carefully.

The upper and lower ranges of crossover frequency settings are calibrated so that they are most useful with typical program material.

For most music a Low to Low Mid frequency crossover of 120Hz will be quite satisfactory. Likewise, a High Mid to High frequency crossover of about 4kHz works for most material.

For fine tuning the sound with different types of program formats, some general guidelines can be given:

For Rock music a slightly lower Low to Low Mid crossover of about 100Hz can put some extra "gut" into the feel of the music – likewise with Dance or music of other formats that contain punchy, deep bass. There is normally not a *significant* amount of fundamental energy below about 60Hz in most program material.

If excessive brilliance is an issue the High Mid to High crossover can be reduced slightly down to 4kHz or lower.

To achieve the opposite effect, that is, to impart a "shimmer" to the highs, one might instead move the High Mid to High crossover up in frequency to around to 6kHz. There is generally no sensible reason to increase the High Mid to High crossover to settings above 7kHz on FM.

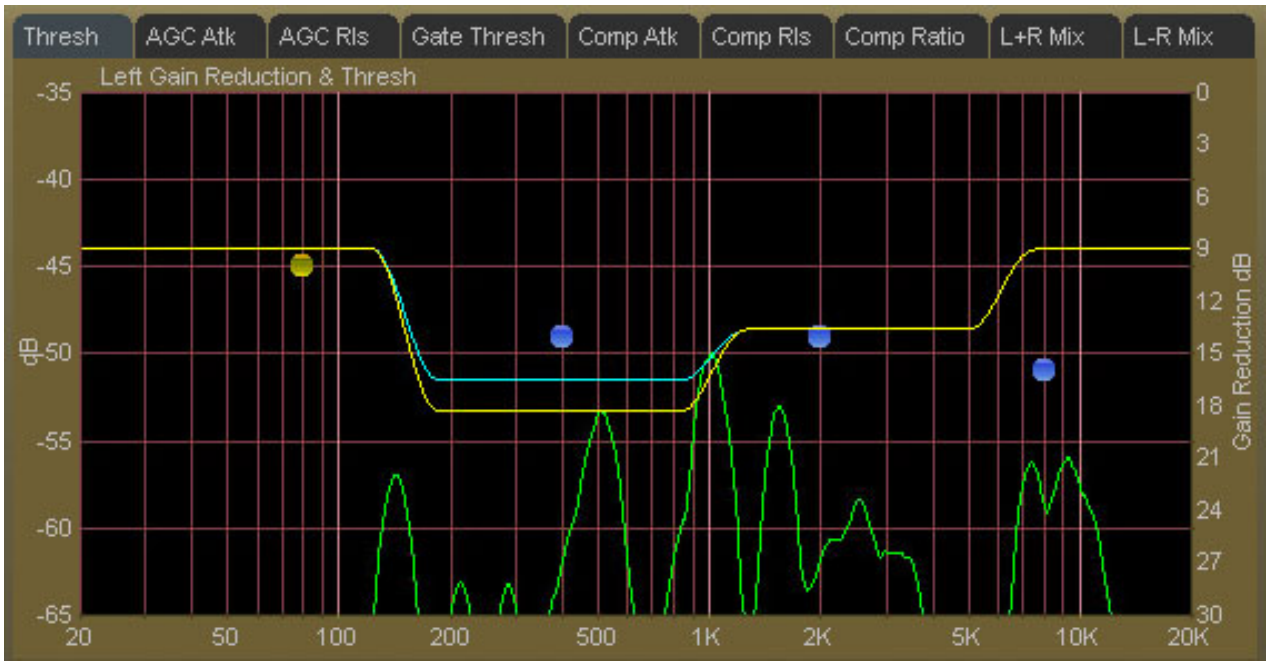
For more middle of the road music and for Classical and Jazz the Low to Low Mid crossover may be set between 120Hz and 150Hz, while the High Mid to High crossover may be set between 3.5Hz and 5.5Hz.

The Low Mid to High Mid crossover may be set around 700-800Hz with such a setting giving good results. Further tailoring of the sound of the midrange may be achieved by skewing the Low Mid to High Mid crossover in one direction or another until the proper sound texture is achieved.

These are very general suggestions and your particular program format may sound better with other settings. Please feel free to experiment!

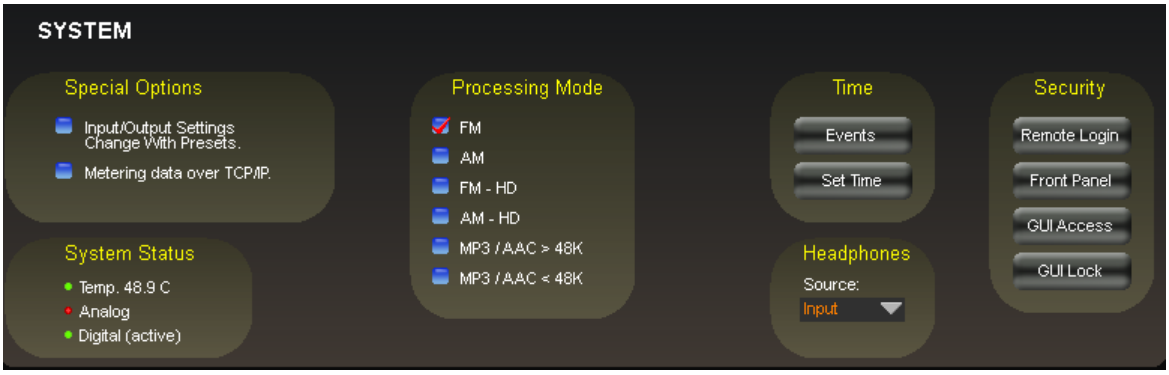
NOTE: Although the AGC and Compressor always operates in sum and difference mode, the crossover frequency settings track both signal chains, assuring a stable stereo image at all times.

AGC/Compressor Gain Reduction



The curve above is broken into four segments representing the instantaneous gain control for each of the four processing bands. The blue line depicts the gain reduction occurring in the slower AGC, while the yellow line depicts gain reduction in the faster Compressor section.

SYSTEM MENU



System Screen

The SYSTEM screen allows system-wide settings to be configured. In the first third of the screen is a very important check box labeled:

Input/Output Settings Change With Presets

When this box is unchecked, recalling presets ignores the settings of the input and output level controls and other system controls associated with those functions. The following are considered to be VP-8IP SYSTEM, or global, parameters, which are saved with presets but not restored when presets are taken, unless the *Input/Output Settings Change With Presets* check box is checked. When this box is checked, the following control settings are recalled whenever a preset is recalled:

From the INPUT Screen:

Audio Source - Analog or Digital

Analog Gain (Analog)

AES Gain (Digital)

Balance (common to both Analog and Digital Inputs)

Phase Rotator

From the FM Output Screen when in FM Mode:

DeEmphasis

PreDelay

Digital Output

From the Stereo Encoder Screen when in FM Mode:

Pilot

Pilot Phase

BS412

SCA 1

SCA 2

TX 1

TX 2

Pilot Only

Analog - L/R, L/R Deemph, and MPX output style choices

baseband192

From the AM Output Screen when in AM Mode:

DeEmphasis

Predelay

TX Tilt EQ

Digital Output

TX 1

TX 2

Analog - L/R, L/R Deemph

From the FM-HD Output Screen when in FM-HD Mode:

Digital Output

Analog Output

From the AM-HD Output Screen when in AM-HD Mode:

Digital Output

Analog Output

From the MP3/ACC>48K Output Screen when in MP3/AAC>48k Mode:

Digital Output

Analog Output

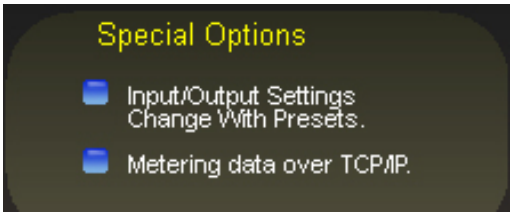
From the MP3/ACC<48K Output Screen when in MP3/AAC<48k Mode:

Digital Output

Analog Output

The option allows the VP-8IP to modify its input and output gain settings whenever presets are recalled. Input and output gain settings are always stored with the presets when processing values are stored, but those settings will be ignored upon preset recall when the *Input/Output Settings Change With Presets* check box is *unchecked*.

Metering Data Over TCP/IP



The VP-8IP utilizes the UDP protocol by default for transmitting metering data to the remote GUI (TCP with its guaranteed packet delivery is *always* used for control).

In some instances UDP packets are blocked by Internet Service Providers, and when this occurs the GUI can successfully connect to the VP-8IP but there is no visible metering data. Unless one can successfully convince the ISP to allow unsolicited UDP traffic on their network (doubtful) the solution is to utilize VPN Tunneling (which encapsulates all packets into TCP) or simply switching the VP-8IP and its GUI to using the TCP protocol for metering.

Switching the VP-8IP to utilize TCP instead of UDP is done through the System menu. In the upper left corner of that screen is a check box *Metering data over TCP/IP*. When the box is checked the TCP protocol is used for transmitting metering data instead of UDP.

Headphone Source

The headphones may be driven by a number of selected points inside the signal processing chain as follows:

Input: Output of the audio input selector, analog or AES.

Ana Inp: Audio appearing on the analog line input.

Dig Inp: Audio appearing on the AES3 digital input.

PreDelay: Output of the FM limiter before the diversity delay.

Ana Out: Final analog output of the processing chain.

NOTE: The VP-8IP headphone routing selector allows you to monitor an unselected audio input *without* having to put it on the air. This can be handy for confirming that an audio source is present without actually switching to it!

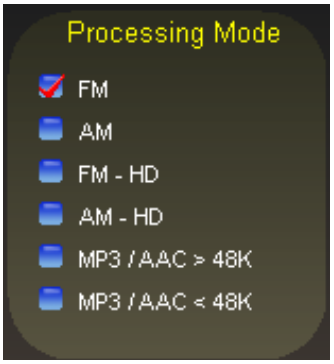
Headphone Control

On the left side of the front panel of the VP-8IP is a quarter inch female TRS jack for headphone use. The volume control for the headphone output is located on the right side of the front panel, and at the lower right of the remote Pro GUI.

To adjust the headphone volume press and hold the black button to the left of the knob while turning the knob clockwise to increase volume and counter-clockwise to reduce it.

Note: The headphone outputs of the VP-8IP are driven by National Semiconductor LM675 power operational amplifiers capable of >3A into any reasonable load. Actual headphone impedance is not a concern; however headphone circuit current limiting is provided at 160mA in order to protect external devices and also to keep you from setting your hair on fire from too high a headphone listening level!

Process Modes



The VP-8IP has six built-in operating modes for FM, AM, FM-HD, AM-HD, MP3/AAC>48k, and MP3/AAC<48k. Changing operating modes is as simple as selecting a new mode from within the Pro GUI; no reboot of the unit is required. For a detailed description of the Process Modes see the next chapter.

Remote Login

When *Remote Login* is clicked, a password prompt box will open as shown below.

Remote Login is for password restricting Pro GUI access to a VP-8IP over Ethernet. When a GUI TCP connection session is initiated, the VP-8IP accepts the connection and then asks the remote Pro GUI for a password.

If the remote Pro GUI cannot provide the correct password, then a dialog box will pop up asking the user to enter the correct password.

Until the login password can be verified the VP-8IP will not send any of its current settings *to* the remote Pro GUI nor accept any changes *from* the connecting Pro GUI.

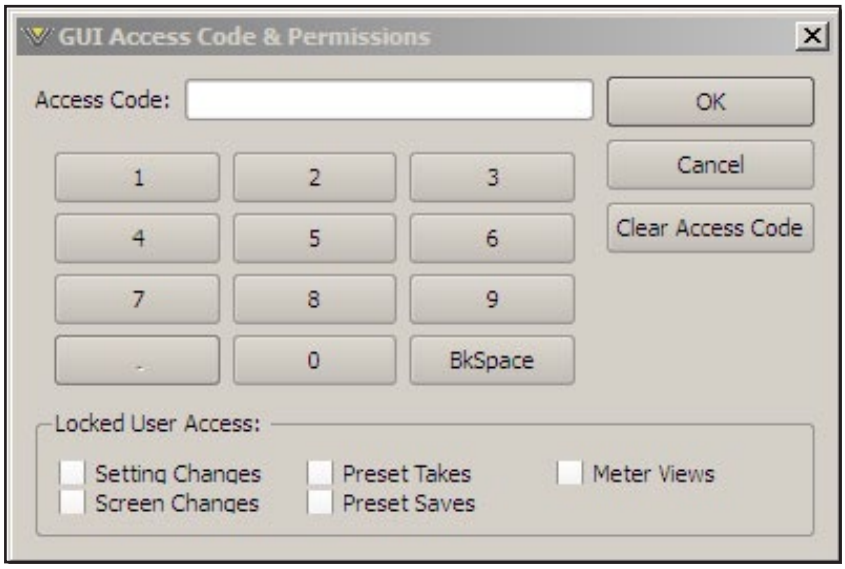
The dialog box allows you to change the login password if one has been set. You will need to know the current password beforehand, and also enter the new one twice for verification. The password can be any combination of characters, numbers, or symbols. If the *Remember This Password* check box is checked, the Pro GUI stores the password on the local PC. If you do not check the check box you will be asked for the password the next time you connect (since the first round of verification will fail). When asked for a password while logging in you can also check the "remember" box to save what you enter. That way you don't need to enter the password every time you login.

Occasionally you may see the word *Pending* temporarily appear in the connection status box of the remote Pro GUI. That means the VP-8IP and Remote Pro GUI are handshaking on the password string that is being sent.

Front Panel

The *Front Panel* button allows the assignment of pass codes to the VP-8IP front panel to control access to it. Front panel access may also be completely disallowed by checking the *Disable Front Panel* choice in the *Front Panel Password* dialog. Note that locking out the front panel does not disable the ability of the encoder knob to alter the headphone volume.

Pro GUI Access



The system allows you to grant access at certain levels of permission in order to control the privileges of those accessing the various features of the VP-8IP. Privileges are granted by the locked user *Access Code*, not by user type. In fact, user types are created *through* the generation of access codes for those user types.

Examples:

- Suppose *Access Code* 0000 is created with *Preset Takes* being the only box checked. This means that a person assigned access code 0000 can *only* take presets.
- Suppose *Access Code* 5678 is created with *Preset Takes* **and** *Setting Changes* both checked. This means that a person with access code 5678 cannot only *take* presets, but can also *make* changes to the settings within them. Note however that because *Preset Saves* was *not* checked when this access code was created, that person may *make* changes to a preset but may not *save* them!

Pro GUI Lock

In the event that security has been enabled by configuring the *GUI Access Code & Permissions* Dialog (above), then selecting Pro GUI Lock will immediately lock the Pro GUI on the host PC, prohibiting access until it has been unlocked with the proper access code.

Unlocking the VP-8IP Pro GUI when the passcode has been lost or forgotten:

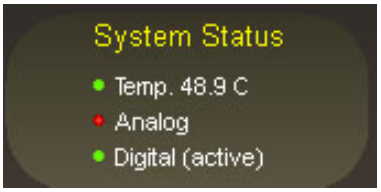
The following instructions must be followed exactly as presented!

1. Re-install the PC Remote Pro GUI, but this time into a NEW folder on your computer. DO NOT start up the Pro GUI yet!!
2. Make sure that the VP-8IP is connected to your network or your computer and that you know its HARDWARE IP address (you can use the command prompt PING command to make sure that your computer can see the VP-8IP). DO NOT PROCEED if you do not have network connectivity to the VP-8IP!

3. Start the new remote Pro GUI on your computer but DO NOT click on the *Connect* button at the top of the Pro GUI next to the *Status* display. Make VERY sure that you are **not** starting the old GUI instead of the new one!
4. Once the Pro GUI is open click on the *Devices* button on the right hand side of the Pro GUI.
5. In the *Devices* dialog that opens click the *Add* button.
6. The *Edit Device* window will open.
7. Enter a NEW name for your VP-8IP but use the same hardware IP address that your unit currently has.
8. Click *OK* to close the *Edit Device* dialog.
9. In the Devices dialog highlight your new device and click *Select*. The *Devices* box will close.
10. Now click on the *Online/Offline* button left of the *Status* window at the top of the Pro GUI.
11. Once the Pro GUI is online click on the SYSTEM tab. The System screen will open.
12. Click on *GUI Access* button (**DO NOT CLICK ON GUI LOCK!!!**). The *GUI Access Code & Permissions* box will open.
13. Carefully click only the *Clear Access Code* button. Then click *OK*.
14. The *GUI Access Code & Permissions* box will close.

The GUI Access Code is now removed.

Status Indicators



Temp – When the *Temp* indicator is green, the operating temperature inside the VP-8IP is satisfactory. When it is red, the temperature is too high. The displayed value is the current operating temperature in degrees Celsius of the VP-8IP motherboard. Normal temperatures should be in the range of 25ºC to 55ºC, and will vary somewhat depending on ambient temperatures near the VP-8IP.

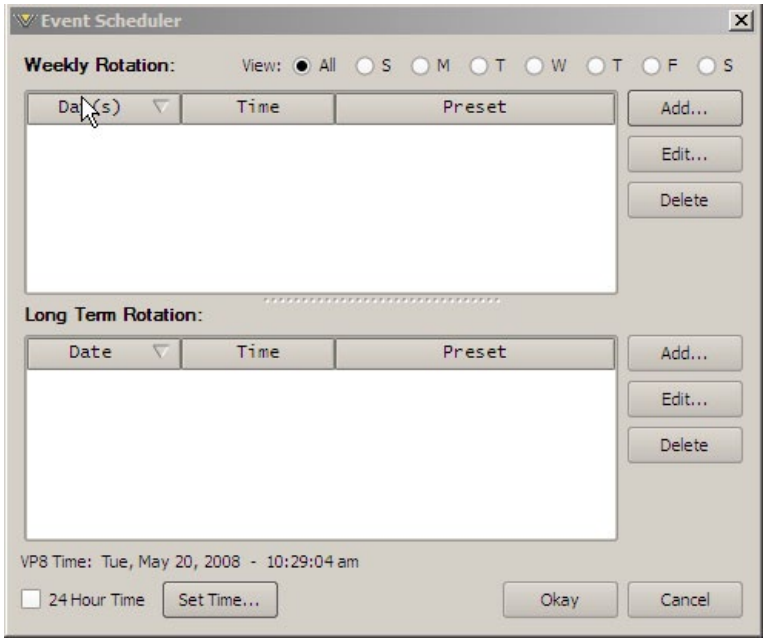
Analog – When green, this indicator shows that audio is present on the analog line inputs and that it is above -48dBFS. When the indicator is red it means that audio is missing or too low to be detected.

Digital – When green, this indicator shows that valid data is present on the AES3 digital input, and that it is above -48dBFS. When the indicator is red it means that valid digital audio data is not presently being detected, audio data could be present but is corrupted in some way, or that audio is present but is too low.

Events

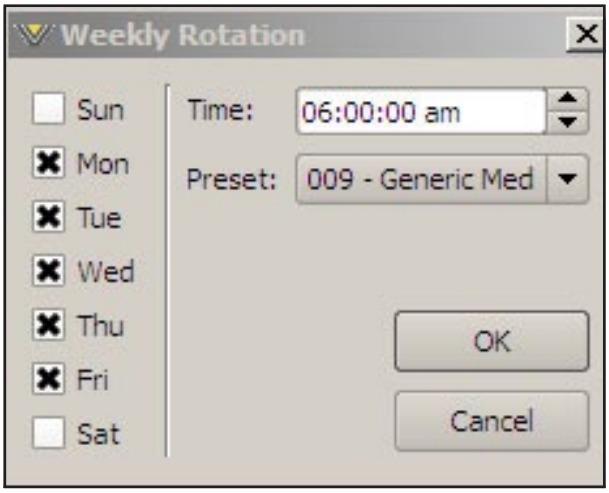
The VP-8IP contains a comprehensive scheduling utility that allows presets to be automatically changed on desired dates and times. This includes a *Long Term Rotation* utility to allows one time preset changes to be scheduled for any time in the future. To create, edit, or view entries in the *Event Scheduler* click the *Events* button on the System Screen.

The following *Event Scheduler* screen will open and the events list will be empty when opened for the first time:



Creating a Weekly Rotation Event

1. Click the *Add* button in the *Weekly Rotation* area of the *Event Scheduler* window. The *Weekly Rotation* window will open as shown below.



2. On the left column, select the days that the preset should be selected. Any combination of check boxes may be chosen.

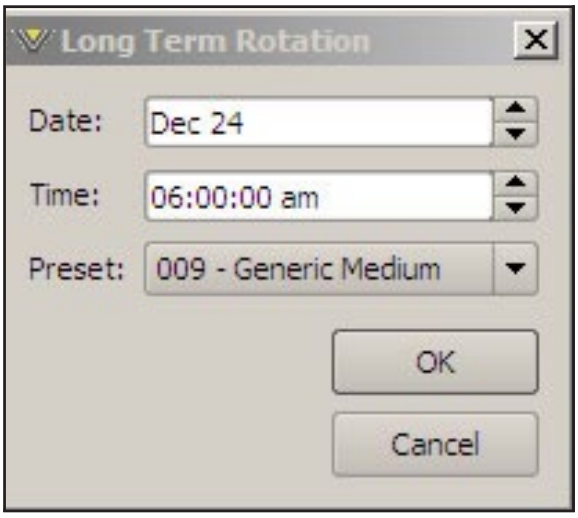
3. Next select the time that the preset should change.
4. Next choose the preset that should be selected at the time programmed in the previous step. When finished editing, click *OK*.

As many preset change events as desired may be programmed in the *Event Scheduler*.

If a one-time preset change is required, such as might be necessary for a special holiday or other event, the *Long Term Rotation* routine should be used.

Creating a Long Term Rotation Event

Adding a new event in the *Long Term Rotation* scheduler works in a similar manner. The *Long Term Rotation* entry window appears as follows:

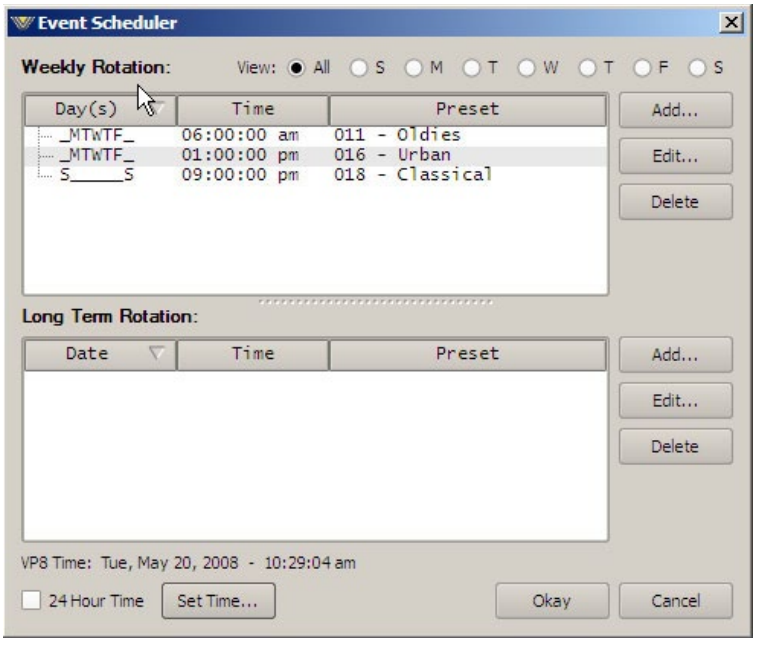


Deleting a Scheduled Event

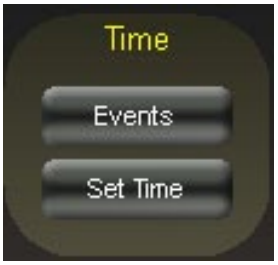
Deleting a scheduled preset change is as simple as adding one. Open the *Event Scheduler* window by navigating to the System menu and clicking *Events*. The *Event Scheduler* window will open.

To delete an item in either the *Weekly Rotation* or *Long Term Rotation* schedules simply highlight it in the list and then click the appropriate *Delete* button on the right side of the Event Scheduler window. The highlighted scheduled event will be removed from the list.

Note that there is no confirmation prior to the actual deletion of the event!

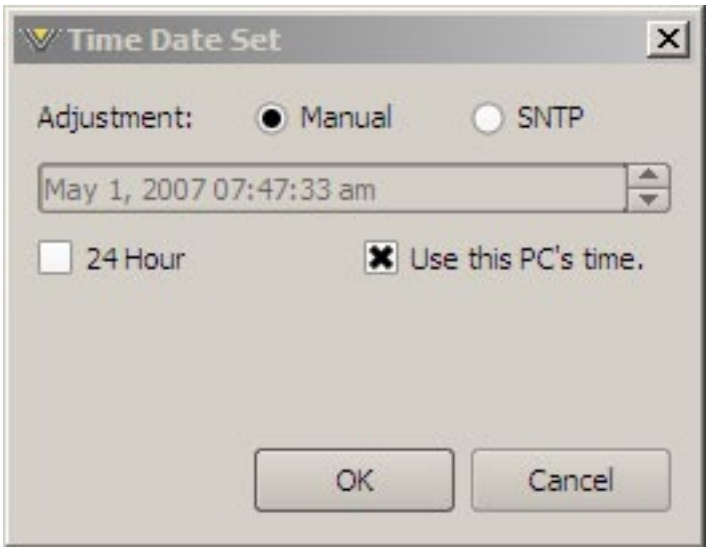


Set Time



Allows the system time of the VP-8IP onboard computer to be set or changed. The VP-8IP system time can be synchronized to external time servers (SNTP, or Simple Network Time Protocol) if desired.

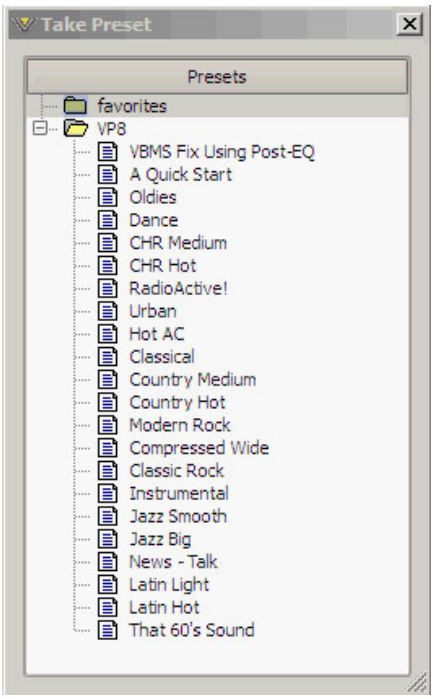
Alternately, time can be synchronized to the PC that is running the Windows-based GUI. In this instance time updates will occur upon successful connection to the GUI on the remote PC.



Side Bar Region

To the right of the control area is a vertically disposed row of buttons:

Preset



When left-clicked, a dialog box appears, showing folders that contain presets, typically favorites, and factory presets, this last containing factory presets already in place in the unit. Opening a folder (if not already open) lists the presets in order. Double-clicking on a preset brings it immediately into use in the VP-8IP. The box stays open until deliberately closed, allowing differing presets to be double-clicked upon readily in succession. This is a very direct means of comparing presets. ***It is important to remember to save the present settings as a preset before invoking others, or they will be lost.***

Note also that you can only select presets that are already loaded in the VP-8IP from this dialog box.

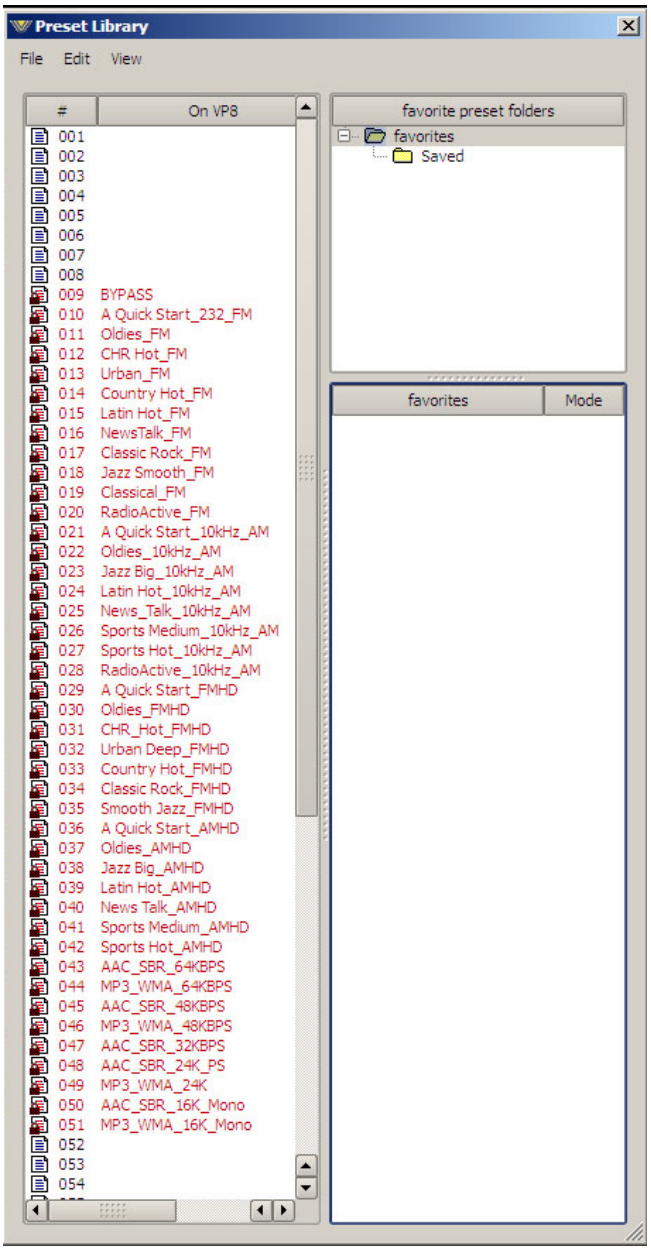
Save

A small dialog box appears, which prompts for a name under which to save the present settings of the VP-8IP. The preset will be saved into the next available empty slot. Alternatively, by nudging the preset number (*Preset #*) up or down, it is possible to save the present settings in place of an existing preset's settings. Needless to say, this erases, as in loses, as in *destroys*, the overwritten preset's contents. Factory presets are protected from being inadvertently overwritten.

Library

A three-panel dialog box appears. The large panel on the left has two columns, being the preset number and the VP-8IP preset contents.

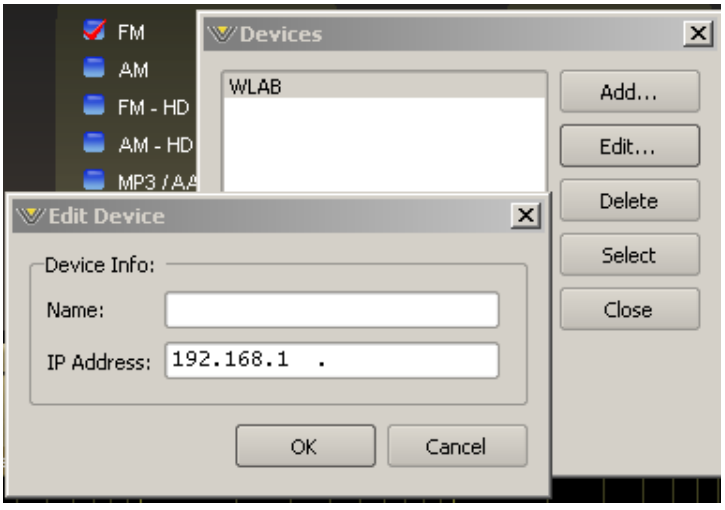
The upper and lower right hand panels give access to the Pro GUI's presets library; the contents of folders in the upper panel are shown in the lower panel; drag-and-drop allows movement between the library folders and the VP-8IP and its mirror archive in the Pro GUI.



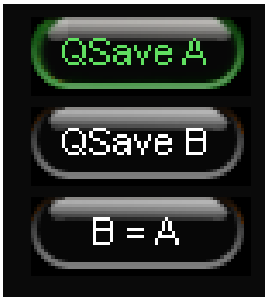
Devices

A list of VP-8IP devices with which the Pro GUI has had cause to become familiar is shown in the *Devices* dialog box. Highlighting the desired VP-8IP's name and hitting *Select* causes the Pro GUI to attempt to connect to it if you are Online.

In the event no VP-8IP's are listed, or one is adding a further device, or a new one for the first time, *Add* brings up the *Edit Device* dialog box, which asks for the new prospect's name and IP address.



Quick Save (QSave)



The VP-8IP has a unique feature that allows instant comparison of adjustments in progress to a known reference, such as another preset, or some midpoint while adjustments are being made. We call this feature **QSave** or **Quick Save**.

In the upper right side of the VP-8IP Pro GUI you'll find three buttons like those shown on the right.

QSave A and *QSave B* represent two temporary buffers that can hold all current VP-8IP processing settings as long as power is applied to the unit.

While *QSave A* is highlighted green any adjustments that you make to controls on the VP-8IP are being saved to its temporary buffer *A*.

QSave B operates similarly to *QSave A*. Its being selected is shown by its green highlighting, and it operates on buffer *B*.

One way to use this feature is to compare the sound of a factory preset to changes that you've made to that preset without having to first save your preset as a user preset. To do this you would:

- Recall the factory or user preset that you wish to adjust.
- Ensure that *QSave A* is highlighted. If it is not, press its button to highlight it.
- Press the *B=A* button. This will copy the contents of QSave buffer *A* to QSave buffer *B*. Now the contents of both buffers are identical.
- Change some settings on the VP-8IP. These settings will automatically be stored in the *A* buffer.
- Compare your changed settings to the recalled factory preset by pressing the *QSave B* button.
- Compare those settings back to the factory preset by pressing *QSave A*.
- When you are happy with your changes you can commit them to a user preset using the *Save* dialog explained previously.

Likewise, *QSave A* and *QSave B* can be used to compare the sounds of two different sets of user settings. To do this you would:

- Load the preset that you want to change, then make changes to it and press *QSave A* to save those settings to buffer *A*.
- Make additional changes as desired and then press *QSave B* to save those additional changes to buffer *B*.
- Now you can compare the two sets of settings by toggling back and forth between *QSave A* and *QSave B*.
- When you are pleased with one set of settings and need more buffers for further tweaking, you can use the *A=B/B=A* button to make the two buffers the same and have one of them to use to start comparing from again.

Title Bar Region

Along the top edge of the VP-8IP Pro GUI screen (in line with the *VP-8IP* product label to the left, and the Windows About, Minimize, and Exit icons to the right) are indicators and controls for the management of devices and presets. The VP-8IP Pro GUI is capable of controlling multiple VP-8IP units (*Devices*) and managing the *Presets* within them. Whether a connection is made (*Status*), and which device and which preset are presently under command are indicated. Double-clicking these indications will gain access to respective management screens.

Status

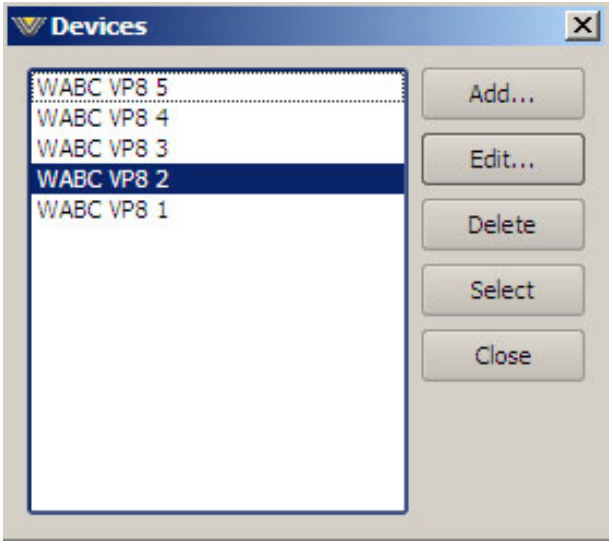
This indicates: *Online*, meaning the GUI is in communication with and directly reflects a VP-8IP; *Offline*, meaning the GUI and VP-8IP are disconnected; and *Trying*, meaning the GUI is attempting to find the VP-8IP on the LAN and connect to it. An adjacent button toggles between *Online* and *Offline*.

Devices

The name of the VP-8IP to which the Pro GUI is connected (or with which it is attempting to connect) shows here in amber.

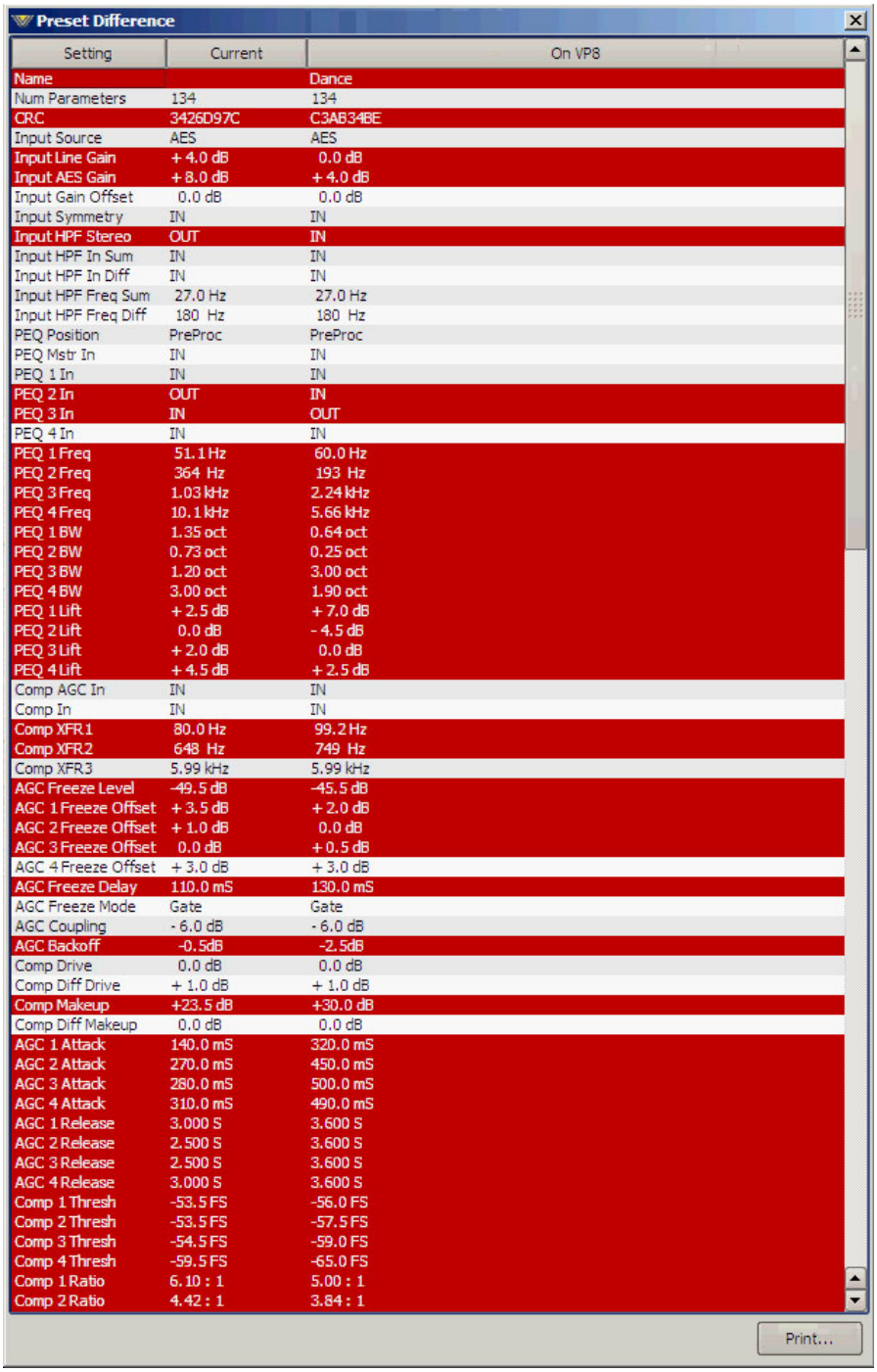
Double-clicking on the name area brings up the *Devices* dialog box containing a list of VP-8IP devices with which the GUI has had cause to become familiar. Highlighting the desired VP-8IP's name and hitting *Select* causes the GUI to attempt to connect to it if you are *Online*.

In the event no VP-8IP units are listed, or one is adding a further device, or one for the first time, *Add* brings up the *Edit Device* dialog box, which asks for the new prospect's name and IP address.



Presets

The name of the last invoked preset is indicated here. Green indicates that the VP-8IP's settings for this preset agree with those in the Pro GUI's mirror archive; red indicates a discrepancy. Double-clicking on the name brings up a *Preset Difference* – parameter-by-parameter, setting-by-setting – comparison list between the current VP-8IP settings and the recalled preset.



Notes on Online and Offline Working

Most often operationally it will be required that the Pro GUI act directly and instantaneously upon the VP-8IP selected under *Devices*; changes on the main control screen take immediate effect on the signal processing and any selected presets are immediately rendered active. This is *Online*.

Offline, however, is more of a supervisory and system management mode and is for those who feel the need to set up complex audio processors without listening to them. Changes can be made to preset settings and configurations, and manipulation of the presets selection list can be made without affecting signal processing within the VP-8IP itself, which can continue to be processing away obliviously. (Preset list manipulation can be undertaken *Online*, too).

Importantly, any adjustments made on the Pro GUI *Offline* will not take effect until the VP-8IP is rendered *Online* again.

Invoking a preset – by double-clicking a preset from the box brought up by clicking the *Preset* button – does one of two things, depending on whether the Pro GUI is in *Online* or *Offline* mode:

- *Online* – The double-clicked preset is immediately sent to the VP-8IP and becomes active.
- *Offline* – The double-clicked preset brings that preset onto the main control and display screen for perusa/editing. Any data entry/control which involves activity within the main graph will still operate upon the preset, but not on the VP-8IP's audio.

Accessing Menu Options

Right clicking anywhere on the VP-8IP Control Panel will open a pop up menu tree with access to *File*, *Hardware*, and *Presets* choices. These choices lead to sub-menus and dialog boxes that may also be accessed by clicking on other dedicated buttons on the main VP-8IP control panel. As with many Windows programs, there are multiple ways to access menu trees – go ahead and explore!

File Menu Items

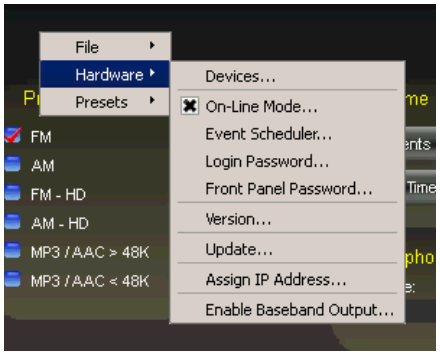
The *File* menu tree may be accessed by right clicking anywhere on the main VP-8IP Control Panel.

About... – Brings up the About box to indicate the Pro GUI version.

Choose Skin... – Brings up the *Choose Skin* dialog box with the option to change the look of the Pro GUI's skin. As new skins are released from Wheatstone they will be available for download from the Wheatstone Processing website.

Exit – Exits the Pro GUI program.

Hardware Menu Items



The *Hardware* menu tree may be accessed by right clicking anywhere on the main VP-8IP Control Panel. Please note that many of these functions require you to be connected (Online) to a VP-8IP. Sub menu choices include:

Devices... – Opens the Devices dialog box. Allows the creation, editing, selection, and deleting of processors connected to your system.

On-Line Mode... –Toggles between ONLINE and OFFLINE modes.

Event Scheduler... – Opens the *Event Scheduler* dialog box for editing Events.

Login Password... – Opens the *Passwords* dialog box for editing login passwords.

Front Panel Password... – Opens the *Passwords* dialog box for editing front panel access passwords.

Version... – Displays the current software versions running in your VP-8IP hardware.

Update... – Opens the *Choose a file to download* dialog box. Only files of type **.vbn* are visible. See below.

Assign IP Address... – Opens the IP Assignment dialog box to change the IP settings of a connected VP-8IP.

Enable Baseband Output... – VP-8IP ships with this feature enabled, so this option should be grayed out.

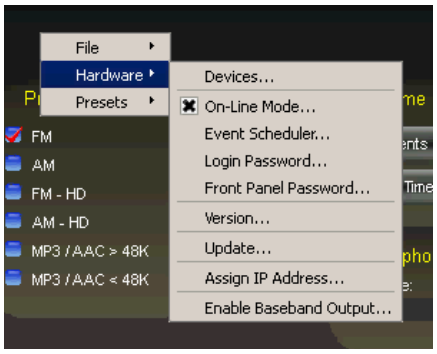
Future versions of the VP-8IP software may be released to implement new features or correct known problems. The Software Update menu choice opens a dialog box and prompts the user to select a file to be uploaded to the VP-8IP. Upon completion, you will be asked to restart the VP-8IP (cycle power).

Only verified updates provided by Wheatstone will work!

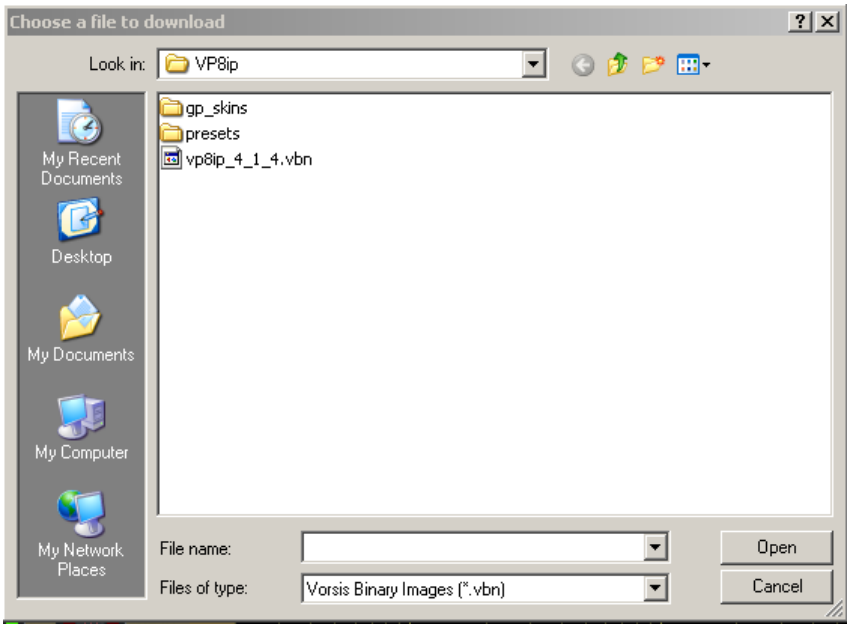
VP-8IP Hardware Update

The VP-8IP hardware is extremely easy to update whenever new firmware becomes available from Wheatstone. There is only one file required for hardware updates, a file with the extension VBN.

The VBN – *VBN* stands for "Vorsis Binary Nugget." This file contains highly encrypted versions of the motherboard application code, FPGA images, and DSP code.

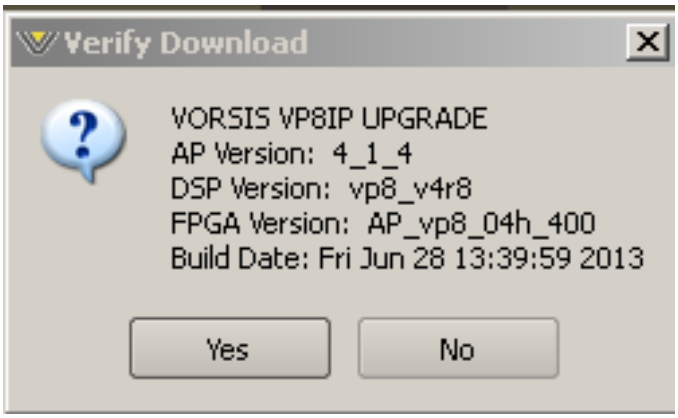


To update the VP-8IP, right click on any portion of the upper control area in any screen of the remote Pro GUI to open the popup menu tree, and select the *Hardware -> Update*... option. The following screen will appear.

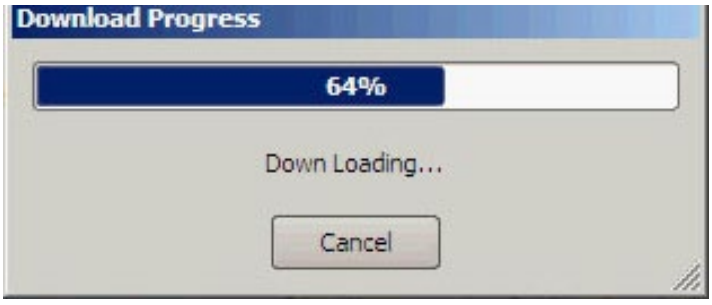


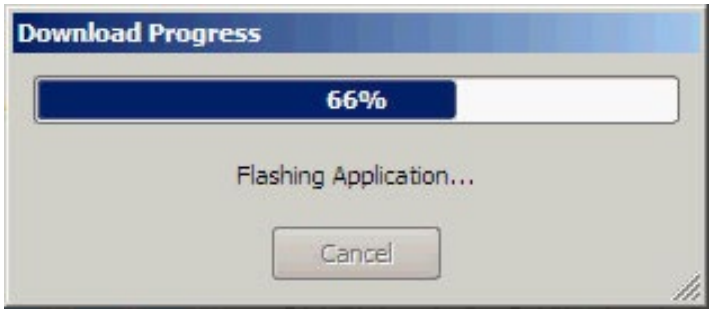
Navigate to the location of the *VBN* file that was downloaded from the Wheatstone Processing website or located on the CDROM that you received from Wheatstone Technical Support. Verify that the version number is correct in that it matches the version number listed in the documentation about the firmware upgrade. If so, select the

Open option in the *Choose a file to download* window. The prompt shown on the right will appear (actual text will vary):

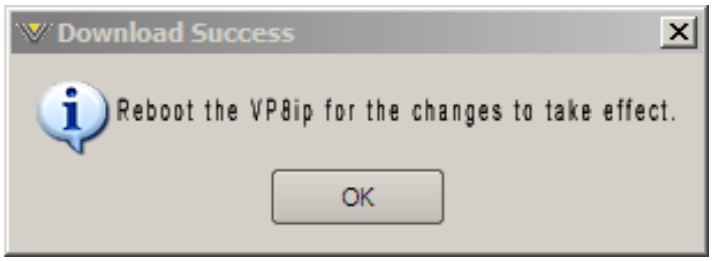


Check once again that the versions match the documentation! Then click *Yes*. Dialog boxes similar to the following will pop up as the download progress occurs:





When the process has finished, you will be prompted to reboot the VP-8IP by the prompt below:



When the VP-8IP reboots the new VBN code will be running in the unit.

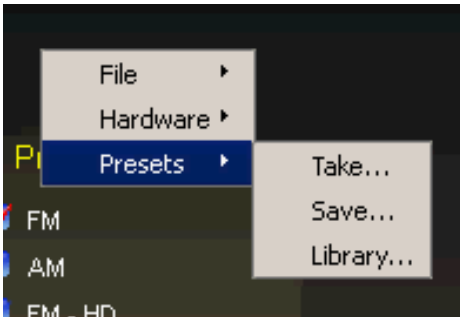
Presets Menu Items

The *Presets* menu tree may be accessed by right clicking anywhere on the main VP-8IP Control Panel.

Take... – Brings up the *Take Preset* dialog box.

Save – Brings up the *Save as Presets* dialog box.

Library... – Brings up the *Preset Library* dialog box.



VP-8IP Process Modes

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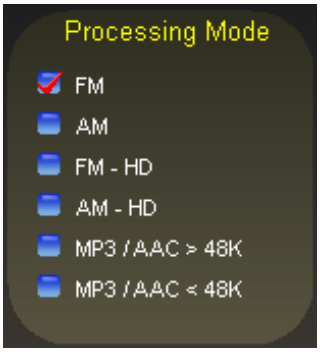
VP-8IP Process Modes

Introduction

The VP-8IP is the industry's first true multi-mode audio processor, with six distinct and separate built-in operating modes. Each of the modes has been specifically designed and then carefully tuned for the intended application. The flexibility of the VP-8IP allows it to be applied to a variety of specialized applications including those that are not specifically spelled out by the names we've given the six operating modes.including those that are not specifically spelled out by the names we've given the six operating modes.

Guru GUI



Pro GUI

Mode Behavior Overview

Processing presets in the VP-8IP are "mode-based." What this means is that presets are always associated with the mode that was running at the time a preset was saved. Said another way, if a user saves a preset while the VP-8IP is in its FM mode, that preset becomes an FM preset. Then if that preset is recalled later when the VP-8IP is in another processing mode the user will be asked if he wants to change processing modes to FM in order to run that preset.

In the System screen of the VP-8IP Guru and Pro GUI's are the mode selection buttons.

FM Mode Summary

In the FM mode standard 50 or 75 microsecond pre-emphasis and peak limiting are performed and the audio is tightly band limited in order to protect the 19kHz stereo reference signal (pilot). This operating mode is also accompanied by a reference grade stereo encoder providing two independent composite multiplex outputs and a pair of subcarrier (SCA) insertion inputs which are digitized at a 192kHz sample rate. While the composite signal is in the digital domain aggressive filtering is applied to constrain the output bandwidth to 53kHz to protect RDS or SCA subcarrier signals that may occupy the spectrum above 53kHz. Baseband192 technology is a standard feature.

AM Mode Summary

In the AM mode NRSC AM pre-emphasis is available within the eight band limiter section and may be applied pre or post multiband limiters. The most commonly used AM and medium wave broadcasting low pass filter frequencies are provided, and all filters are overshoot-free to assure accurate modulation control. Low Frequency Tilt Correction is also independently available for both transmitter outputs to offset inadequacies in the transmitter's low frequency response that can compromise modulation accuracy. Specialized limiting and clipping algorithms permit up to 150% positive modulation for those applications where it is warranted.

FM-HD Mode Summary

In the FM-HD mode special look ahead algorithms developed by Vorsis condition the audio in a manner that makes it most suitable to be coded by the iBiquity FM HD perceptual codec. The audio source for the HD mode look ahead limiter is the output of the four band AGC/Compressor.

AM-HD Mode Summary

This mode has been optimized to operate favorably with the HDC (Hybrid Digital Coding) codec, a proprietary and lossy audio compression scheme developed by iBiquity for specific use by US AM-HD stations. HDC is known to utilize SBR (Spectral Band Replication), and in 2003 it replaced the earlier PAC codec which had been proposed for AM-HD. Very little is known (outside iBiquity) about the codec's inner workings, although some codec experts believe that it may be a customized version of MPEG-4 HE-AAC.

The input to the AM-HD mode look ahead limiter comes from the output of the four band AGC/Compressor. While there are others involved, two important pre-processing steps involve (1) carefully managing the stereo image to ensure that non-stereo information is never presented to the codec, and (2) managing the transient high frequency spectrum to ensure that certain kinds of high frequency energy is also never presented to the codec.

MP3/AAC >48k & MP3/AAC <48k Mode Summary

In the previous "codec" modes the end user knows what codec is being used to perform the audio data compression and therefore has some insight as to its probable behavior with his program material. The two MP3/AAC>48k and MP3/AAC<48k modes assume that the user may not always know what codec(s) are being used during transmission of his audio; however he may at least know what bitrate the codec is utilizing.

Because the behavior of perceptual codecs have certain predictable similarities when operated at various bitrates (as far as how they analyze the incoming audio for opportunities to discard information within it) it becomes a bit easier to design processing algorithms that can make the best use of the codec's ability to mask the material that it must remove.

This last point is important: if a codec is continuously tasked with coding increasingly difficult-to-code material its decision-making process will begin to permit it to discard more and more audible parts of the incoming program material. The task of a codec pre-processor then is to purposely condition the incoming program in a manner that makes the codec's job easier, even if it means making subtle changes to certain aspects of that program material before it ever gets to the codec.

This task of codec preprocessing is arguably best accomplished by utilizing special audio processing techniques that purposely alter the program material prior to the codec in a manner that either removes subtle audio information that the codec does not need to waste its precious few bits coding, or enhances it to ensure that the codec assigns enough bits to obtain good quality. Because there is a fixed bit pool in most codecs, removing unnecessary information prior to the codec leaves "more bits" available to code the most audible portions of the program, which increases the perceived quality for the listener.

Audio Processor Tuning for CODECS

We strongly caution the user to never use the sound at the output of the VP-8IP to determine how to tune it for best behavior with a codec or make judgments about the sound quality. This is because the work that the VP-8IP must do to "fool" the codec into producing a better overall audio quality may cause it to sound odd or unnatural with program material. You should always listen at the codec output in order to make judgments about the sound quality!

CODEC Overview

Though there are many CODECS in use throughout the world in myriad specialized applications, the most popular CODECS that a typical VP-8IP user will currently encounter in the field are:

- AAC (Advanced Audio Coding) has replaced MP3 in many applications due to its higher efficiency and higher audio quality at equivalent bitrates.
- High Efficiency AAC (HE-AAC) is a lossy data compression scheme and an extension of the Low Complexity AAC (AAC LC) codec that was originally optimized for low-bitrate applications such as streaming audio.
- HE-AAC version 1 (HE-AAC v1) uses Spectral Band Replication (SBR) to enhance compression efficiency in the frequency domain.
- HE-AAC version 2 (HE-AAC v2) couples Spectral Band Replication with Parametric Stereo (PS) to further enhance the compression efficiency of stereo signals.
- Microsoft's WMP Professional codec is quite similar in behavior to HE-AAC. The same holds true in general terms for a very general comparison between the MPEG-4 / AAC-LD codec. Note also that the HE-AAC codec is also sometimes referred to as aacPlus and AAC+.

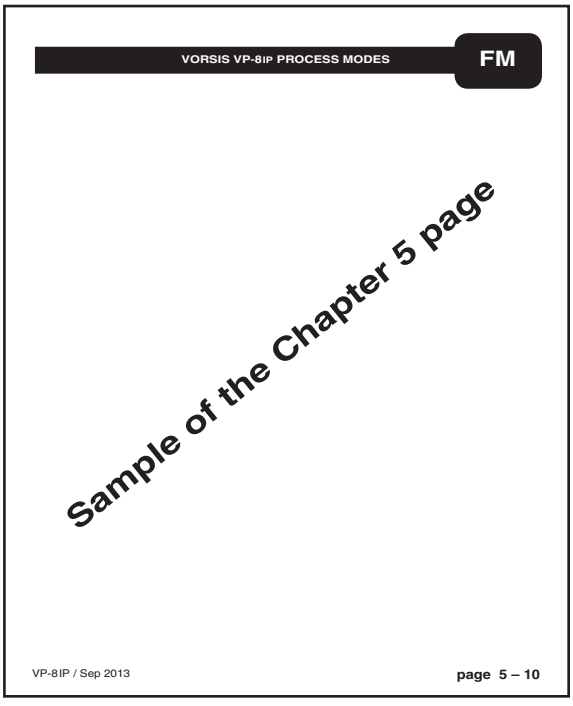
Processing Common to All Modes

In all processing modes the incoming audio is first preprocessed by an all-pass filter (phase scrambler) to remove the natural asymmetry from human voice. A high pass filter then removes non-program related subsonic signals. The signal then passes to a four band full-parametric equalizer followed by an adjustable phase-linear four band crossover. The signal then passes to the four band AGC/Compressor, which is equipped with adjustable inter-band coupling, individual compression ratio controls, and full adjustments for attack, release, band-by-band gating, and output mix levels in the sum and difference domains.

In all processing modes final peak limiting is accomplished by our exclusive eight band peak limiter. Operating modes other than the conventional analog AM and FM modes utilize an oversampled, look ahead peak limiter for final peak control. The AM and FM analog modes utilize (if desired) an oversampled, distortion controlled final clipper which has several available "knees" to enable the sound of the peak limiting to be customized.

The FM analog mode also has a DSP-based stereo coder which generates the composite baseband signal required for analog FM broadcasting. An oversampled composite processor which is used to gain further loudness may be operated in its "hard," "soft," or look ahead modes as desired.

Changing from one VP-8IP operating mode to another is as simple as selecting a new mode from within the System Screen on the PC-based Windows Graphical User Interface – no reboot of the unit is required. While mode changes are taking place the output audio is muted during the process of loading the new mode's DSP code into the DSP farm, preventing undesirable digital noise at the processor's output during mode changes.



NOTE:
The balloon at the upper right on the following pages clearly indicates the processing mode described on that page.

VP-8IP FM Process Mode

The following discussion is unique to the VP-8IP Audio Processor when it is operating in its FM mode.

There are two menu screens that are pertinent to the FM mode, used for adjusting how the processor's final limiting scheme sounds and how it interfaces with the equipment to follow it. These are the Limiters menu and the FM (Stereo Encoder) menu.

LIMITERS MENU - FM MODE

Limiters Screen in FM Mode

Multiband Limiter

The VP-8IP contains an exclusive and fully adjustable eight-band FM final limiter algorithm. This algorithm is especially suited for controlling audio peak energy without generating unnatural artifacts such as "swishing," "squashing," and "phasing" that outdated and simplistic algorithms having fewer bands generate.

The only caveat for using the eight band limiter, at least as far as our opinion is concerned, is that it is probably inappropriate for most program material to drive the limiters continuously to more than 6dB of indicated gain reduction. Instead, we recommend operating the limiters in a manner that allows them to only take care of shorter duration transients that the preceding four band AGC/compressor misses.

For the best musicality with music-based program formats we recommend that the limiters not be operated with extremely short attack and release times. While the limiters are capable of attacking audio peaks in as short as 100 microseconds, we recommend attack times at the mid and high frequencies of around ten milliseconds or perhaps slightly less. Lower frequencies can benefit from longer attack times – perhaps 50 to 80 milliseconds in order to retain good bass punch. Please open several of our factory FM presets and observe the settings in the limiter section to see what we mean.

The eight band limiter section has been designed to be fully adjustable in order to accommodate a wide variety of program material and competitive needs. It also contains, and is tightly integrated with, the proprietary Vorsis Bass Tools to intelligently control bass program energy. In fact the Bass Tools is the reason why we can recommend longer limiter attack times at low frequencies – the Bass Tools will manage the (intentional) limiter overshoots caused by lengthening the bass attack times in an artistic way.

Multiband Enable

This check box determines whether or not any processing is to be applied by the eight band multiband limiter. When the eight band section is not enabled, audio peak control is handled solely by the main clipper, the look ahead limiter (if selected), and the composite processor as selected within the FM stereo generator screen. Note that switching out the multiband limiter changes the gain structure and therefore the *Final Lim/Clip Drive* control may need to be readjusted slightly to compensate.

Multiband Drive

Determines the amount of input signal applied to the eight band limiter. Less drive will create a less dense sound while still controlling peaks, while higher *Multiband Drive* settings make the sound more aggressive by increasing short term audio density.Very high drive settings, or settings that result in more than 6dB of continuous gain reduction are typically unnecessary with most formats.

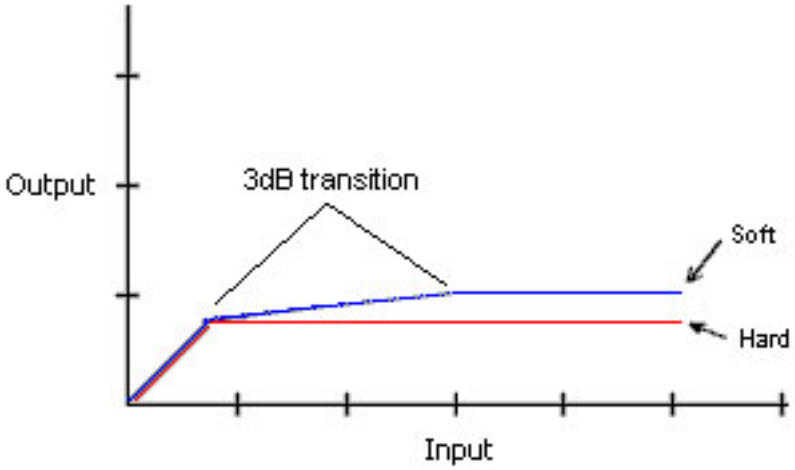
The *Multiband Drive* control adjusts the overall gain into the eight band limiter section, and its relative setting is dependent upon the setting of the *Makeup* control in the four band AGC. For this reason, the *Multiband Drive* control is not labeled in dB but instead is labeled in percent (%). With sane settings of the four band *Makeup* control the eight band limiter *Multiband Drive* control will usually be found between 85% and 100%. The full adjustment range is 0% to 100%.

Higher *Multiband Drive* settings also cause deeper levels of clipping in the main clipper algorithm if the *Final Lim/Clip Drive* is not first reduced before the *Multiband Drive* is advanced. Therefore, if you feel that the *Multiband Drive* control is adding undue distortion, it means that the *Final Lim/Clip Drive* control setting should be reduced by several dB before the *Multiband Drive* is advanced. The *Multiband Drive* and *Final Lim/Clip Drive* are the two main controls for trading off "loudness," "brightness," "density," and "distortion."

Multiband Knee

Each of the eight bands' threshold, attack, and release are adjustable, as is the overall transfer function (knee hardness), which affects how the limiters behave near the threshold of limiting. The graphic below illustrates the difference between the two transfer functions.

In the *Hard* mode, limiting in a band does not occur until the threshold within a band is precisely reached. The sound of this setting is more open and free because no limiting activity occurs until the threshold is achieved. However, once above threshold the audio levels cannot increase.



Comparison of Hard and Soft Limiter Modes

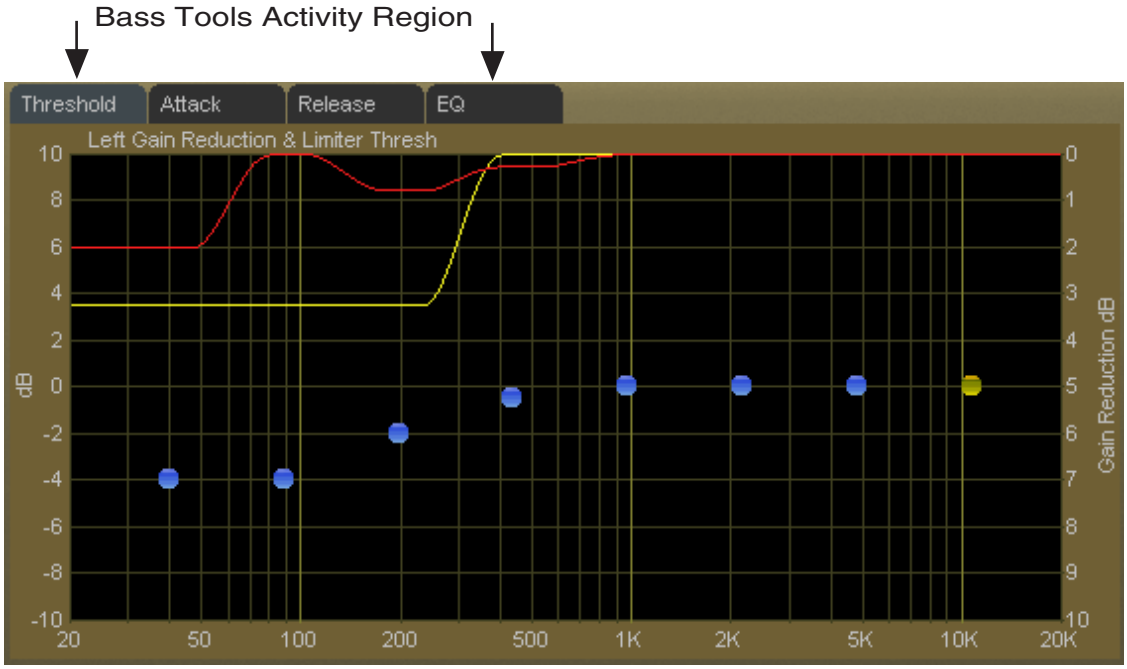
In the *Soft* mode, limiting begins to occur at a level that is 3dB below the actual limiter threshold setting. The transfer function at this point is relatively soft. That is, the audio waveform is not abruptly limited, but instead is allowed to increase over an additional 3dB window until the ultimate limit threshold is reached, whereupon no further amplitude increase is allowed.

Because of its behavior, the sound of the *Soft* function is less obvious and has a "spongy" and "warmer" sound when in operation. It is also less precise as a true limiter until the ultimate limit threshold (the setting of the band threshold) is finally reached.

Bass Tools

Vorsis Bass Tools allows extremely low program frequencies to be processed without causing dynamic gain changes and without generating undesirable intermodulation artifacts in higher frequency signals. Its primary purpose is to allow the thumps of kick drums and other percussion instruments to sound natural in their "percussiveness" without pumping or other undesired artifacts. The algorithm operates in left/right mode treating both channels identically.

Our laboratory research on codec behavior rather surprisingly revealed that it is perfectly acceptable to process very low frequency material using the Bass Tools algorithm provided that the harmonics generated during the process are well controlled spectrally. Since our bass algorithm does not allow significant harmonic levels above the third or fifth (somewhat dependent the Bass Tools *Style* setting) there is no audible downside to using it with coded audio. Furthermore, the ability to utilize the Bass Tools on both the FM and HD sides of the processor allows the sound to be more evenly matched as receiver-based digital to analog crossfades are encountered.



Bass Tools Activity Display - 20Hz up to VBMS Frequency Setting

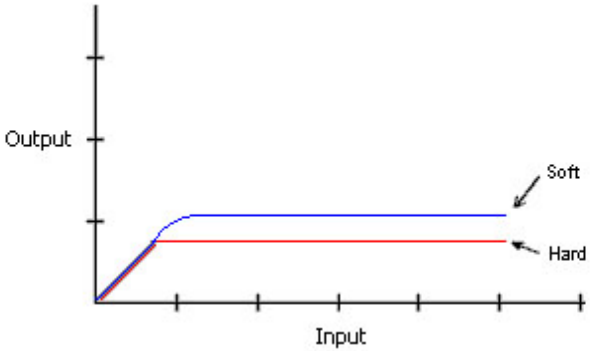
Drive

This control adjusts the amount of Input Drive to the Bass Tools algorithm. Higher drive settings make it work harder and generate higher levels of harmonics. The *Drive* control adjusts the amount of drive to the Bass Tools and is adjustable over a range of -6dB to +12dB. Adjustment is best done by ear with percussive source material.

Style

Two Bass Tools operating styles, plus Off, are available. The *Hard* setting has a transfer function occupying approximately 1dB of transition amplitude. The waveform control is fairly abrupt and generates low order harmonics.

The *Soft* setting operates with a transfer function occupying approximately 2dB of transition amplitude. Therefore the control is less sharp than the hard setting, generating more low order harmonics. While the two settings may appear to be almost the same, the 1dB difference in transfer ratio creates a *significant* difference in how the two Bass Tools styles *sound* on actual program material. When Off is selected, the Bass Tools processing is bypassed.



Comparison of Bass Tools Operating Modes

Frequency

The *Freq* control is simple. All audio below the frequency selected is processed by Bass Tools. Smoother bass can be had with settings above 200Hz, which is recommended.

Output

The Bass Tools *Output* control adjusts the output of the Bass Tools algorithm and can be thought of as a "low bass mix" control. Settings between -6dB and -3.0dB add extra bass to the overall mix, while settings lower than -6.0dB reduce it. Both the Bass Tools *Drive* and the *Output* controls are best adjusted by ear using percussive bass material. The *Output* control is adjustable between -12dB and 0dB.

Pre-Emphasis

Type

Pre-emphasis is selectable as 50 or 75 microseconds, or Flat (OFF).

Location

When pre-emphasis is enabled it may be placed in one of three locations in the signal chain:

Pre-MBLim – In this position, the selected pre-emphasis is placed *before* the eight band limiter/clipper section. In this mode the eight band section may be operated as a frequency conscious limiter tailored to the inverse of the selected pre-emphasis.

Pre-Limiter – In this position the pre-emphasis is *after* the eight band section but *before* the Look Ahead limiter (if enabled) and/or main clipper.

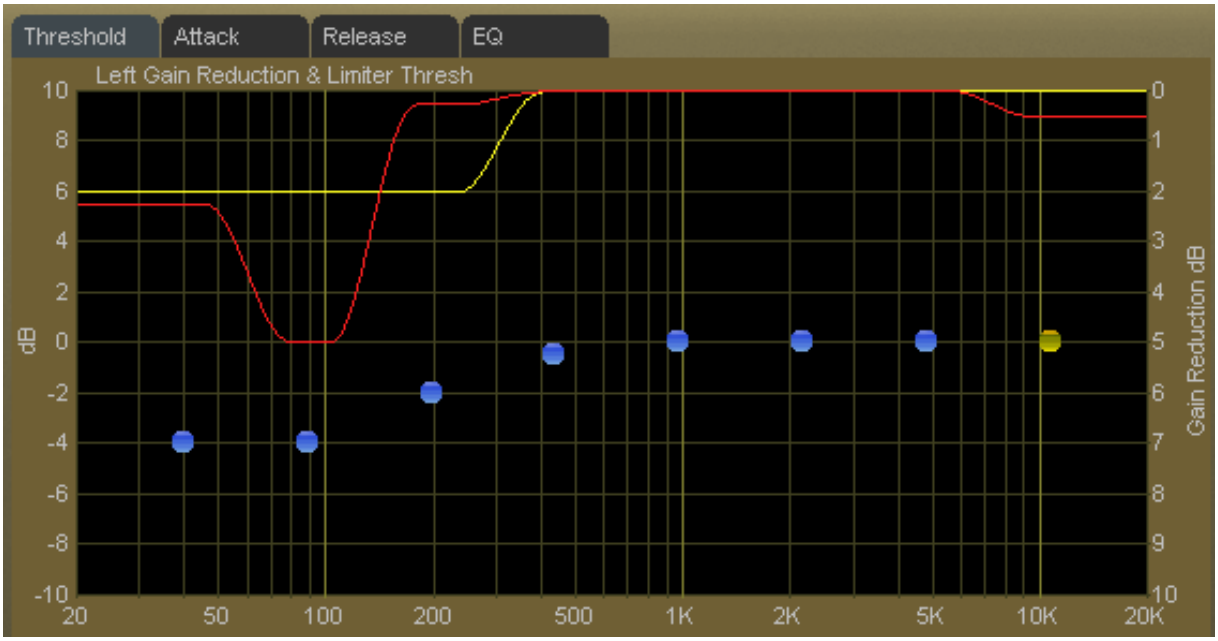
Post-Limiter – In this position the pre-emphasis is *after* the eight band section and Look Ahead limiter and *before* the main clipper.

Entirely different sound textures may be created by placing the pre-emphasis in one location versus the other. There is no right or wrong setting unless that setting does not produce the on air sound desired.

Multiband Limiter Adjustments

Threshold Screen

Clicking on the *Threshold* tab on the graphic opens up the adjustment screen for the limiter thresholds.



Multiband Limiter Threshold Screen

The thresholds are adjustable over a +/- 10dB range and interact with the *Multiband Drive* control. Because the thresholds are adjustable over such a wide range, frequency conscious limiting can be created if needed.

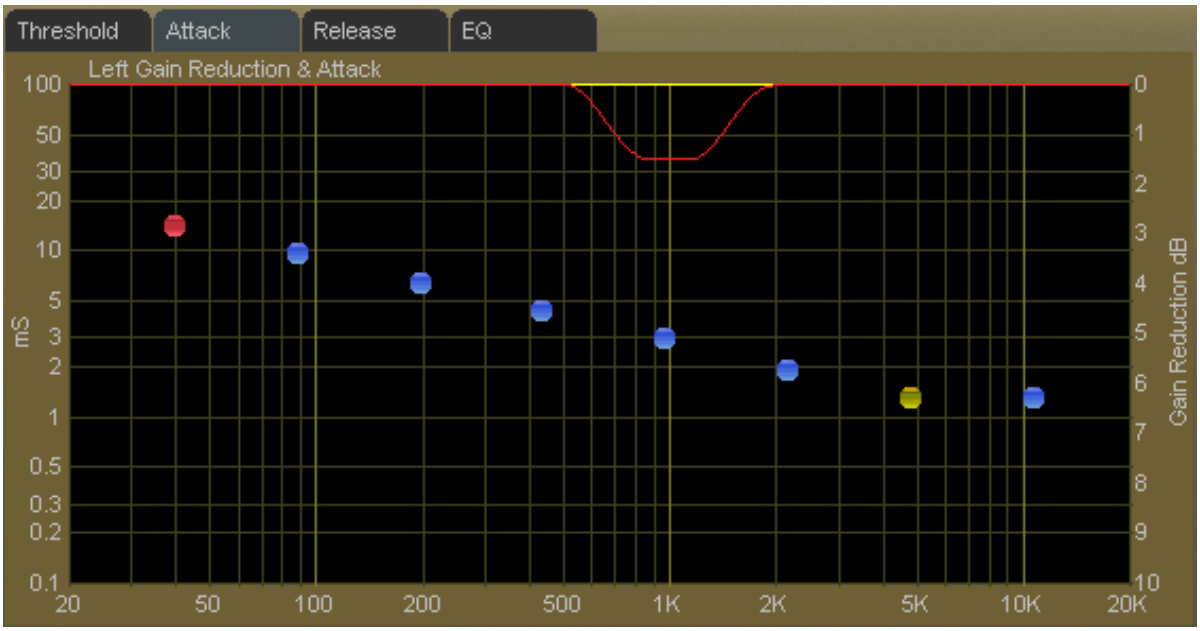
Each of the limiter thresholds can be adjusted separately or in groups according to the manner described in the earlier *VP-8IP Graphical Interface Operation* section. Lower settings (more negative numbers) reduce the threshold, causing limiting to occur at lower audio levels. Conversely, higher thresholds cause limiting to occur at higher audio levels.

Because of the frequency response contouring in FM pre-emphasis it is generally customary to adjust the limiter thresholds according to the limiting sensitivity curve required, creating a multiband frequency conscious limiter of any configuration.

Note: Radically different settings between adjacent bands won't cause strange sounds – there is no swishing in the Vorsis multiband limiter algorithm. Operationally, though, it doesn't make much sense.

Attack Time Screen

Clicking on the *Attack* tab opens up the attack adjustment screen.



Multiband Limiter Attack Screen

The attack time of a limiter is a measure of how fast it responds to an audio peak once it is above threshold. Shorter attack times respond to audio peaks quicker and allow more accurate peak control, but excessively short attack times can quite literally suck the life out of musical material by erasing the short term transients that allow it to sound dynamic.

On the other hand, excessively long attack times will allow audio peaks to escape the limiters largely unmanaged, leaving them to be controlled solely by the final look ahead output limiter (if enabled) or the final clipper.

The attack times for each band are adjustable from 0.1 milliseconds (100 microseconds) to 100 milliseconds. As explained above, faster attack times exert more control over waveform excursions, while slower attack times allow peaks to escape the limiter section with less control.

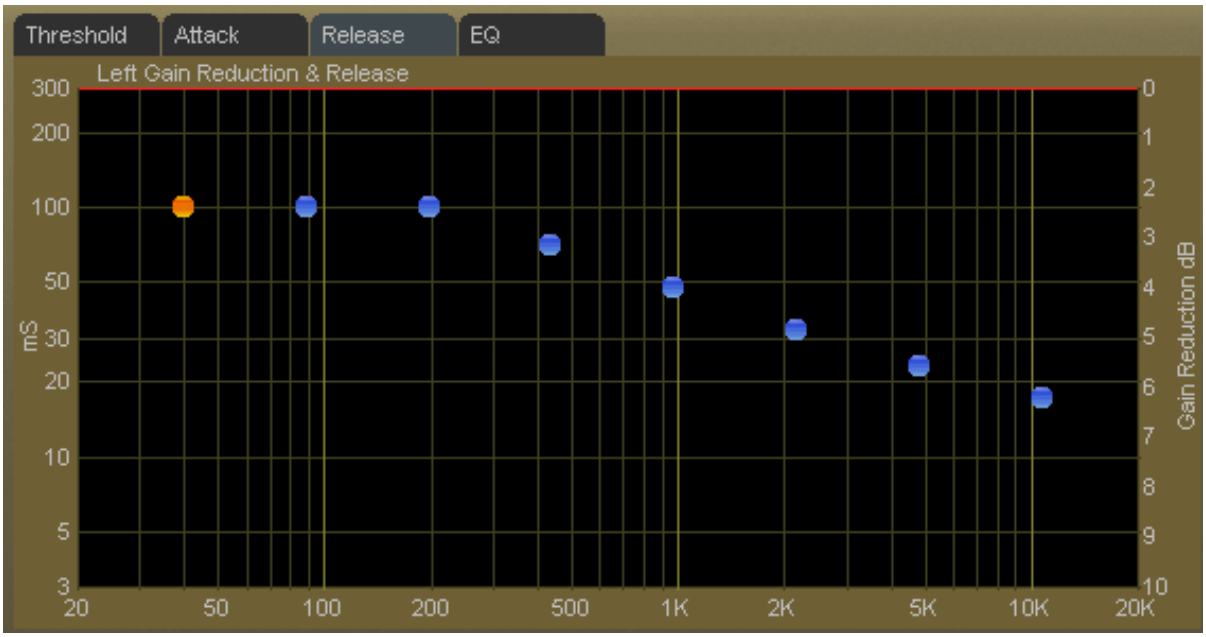
Attack times are generally adjusted to taste, keeping in mind that the slower the attack time the more work the final clipper or limiter must do, while conversely the faster the attack time the fewer short term dynamics the program material will have.

Generally, lower frequencies require slower attack times and higher frequencies require faster attack times for a given amount of control.

The attack times of each band may be individually or collectively modified using the GUI operating instructions provided earlier.

Release Time Screen

Clicking on the *Release* tab opens up the release time adjustment screen.



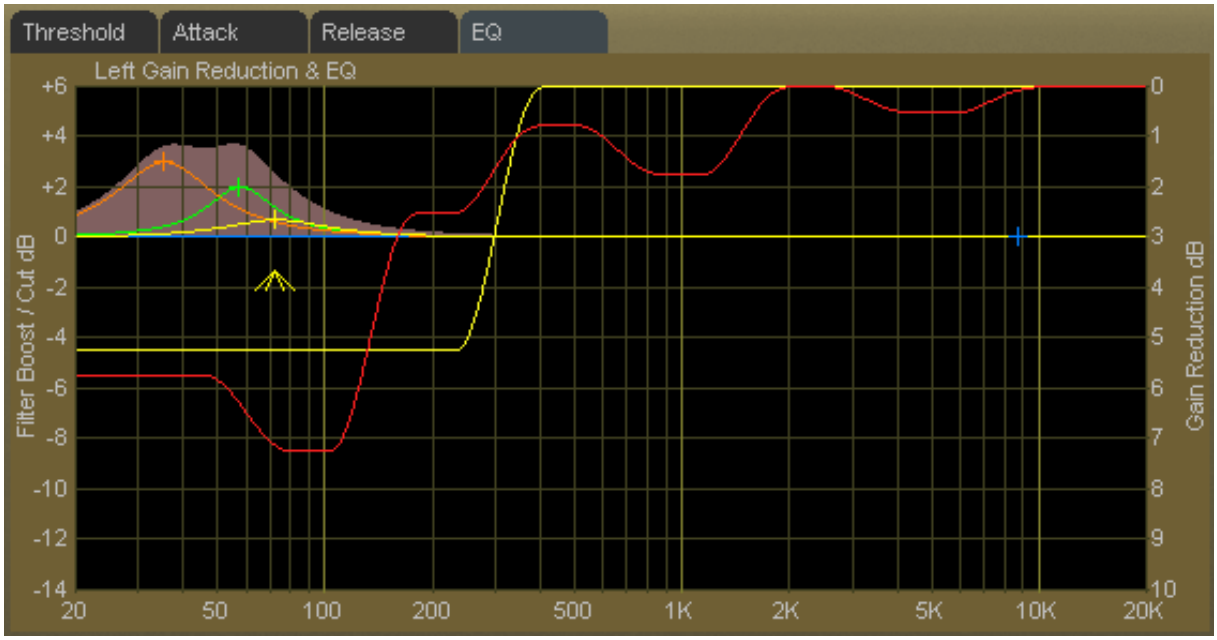
Multiband Limiter Release Time Screen

The release time of a limiter determines how fast it recovers from an audio peak that has caused a momentary decrease in gain. Shorter release times create a more dense sound, and excessively short release times can also increase intermodulation distortion (the mixing of low and high frequency energy – or a muddying of the sound).

The release times for each band are adjustable from 300 milliseconds down to 3 milliseconds. Faster release allows a more rapid recovery after a limiting event, while slower release times do the opposite. Release times should be adjusted to taste, keeping in mind that the slower the release time the more dynamic the sound, but the more likely that audible ducking may be heard.

Final Equalization Screen

Clicking on the *EQ* tab opens up the final equalization screen.



Multiband Limiter Equalization (EQ) Screen

Under certain conditions it may be desired to *slightly* alter the sound exiting the final eight band limiter. A *small* amount of equalization performed after limiting can open up the sound or provide a *slight* enhancement to its overall feel, and minor equalization tweaks can be accomplished after the limiters without radically disturbing the peak levels emerging from the eight band algorithm.

Only very small amounts of boost equalization are recommended once final limiting has been accomplished in the eight band section. In fact, the adjustment range has been limited to +3dB/-12dB in order to curtail the temptation to perform aggressive equalization where it is the most inappropriate, and even this range is quite large considering the effect that can be created.

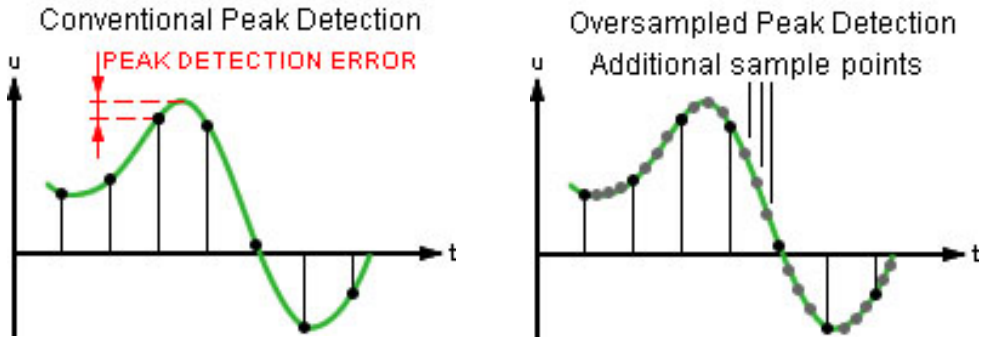
Small adjustments in EQ boost – on the order of 1dB or 2dB at most – are sufficient to alter the overall tonality of the eight band section.

Equalization boosts must be approached with caution because they can add significant energy that must be controlled later by either the look ahead limiter or final clippers.

On the other hand, equalization *cuts* can be made as desired because they do not *add* energy.

FM Look Ahead Limiter

If desired, the FM output may be peak controlled by a special high-speed, feed forward limiter with a 0.5 millisecond (500 microseconds) look ahead time. The control path (not the audio path) is highly oversampled in order to achieve extremely precise peak control. The primary goal of oversampling within the control sidechain is better peak control through finer definition of the audio waveform to be analyzed for limiting. Therefore, oversampling of the audio signal itself is not only unnecessary, but wastes DSP resources. The graphic below illustrates why control signal oversampling is effective.



By sampling the audio waveform with finer resolution we can detect peak excursions that would be missed otherwise. In fact, if *both* the audio *and* control signals were oversampled audio peaks could still be missed because their relative time alignment would be identical!

FM Look Ahead Limiter Controls

The graphic on the right shows the controls available for the look ahead limiter. The following text describes each control according to function.

Look Ahead Limiter Enable

When this check box is checked the look ahead limiter is enabled. Note that the look ahead limiter can be used simultaneously with the main clipper and precedes it. This feature allows for a very powerful limiter/clipper behavior tuning capability.

L/R Linked

When this box is checked the limiter's left and right channel *control* signals are made common. This prevents unnatural stereo image shifts when one channel needs to reduce the signal in order to control peak excursions.

Lim/Clip (Limiter/Clipper) Drive

This controls the amount of drive to the following distortion-controlled clipper and/or look ahead limiter, as chosen. The gain range of this control is +12dB/-18dB. The resolution of this control is 0.01dB, fine enough for the most exacting drive level adjustments.

Attack

Adjusts the attack time of the look ahead limiter. An attack time setting of 0.2mS creates a look ahead limiter with a 0.3mS negative attack time (look ahead) because the total look ahead delay is 0.5mS. Therefore a setting of 0.5mS creates a limiter with zero attack time (again, because the total look ahead delay is 0.5mS). All attack time settings above 0.5mS create a conventional limiter with the control calibration skewed a negative 0.5 millisecond because of the fixed look ahead time. The full range is 0.2mS to 100mS.

Release

Controls the primary (fast) release time of the look ahead limiter.

Higher settings create longer release times which cause the peak limiter to recover more slowly from a limiting occurrence; release times that are set too slow may allow undesirable signal ducking to occur as well as unnaturally lower the average program level.

Conversely, fast release times increase signal density, and release times that are too fast result in increased intermodulation distortion which can "muddy" the sound. See Delayed Release below.

The overall range of the control is from 33 milliseconds to 330 milliseconds.

Delayed Release

Controls the secondary (slow) release of the look ahead limiter, and is adjustable from 100 milliseconds to 1 second.

When Delayed Release is engaged via the *Enable* check box, the limiter will release the first 3dB of gain reduction at the (faster) Release setting, and the remainder of the Release at the Delayed Release timing. This results in a drastic reduction in intermodulation distortion (see Release, above) and allows more limiting depth without incurring a distortion penalty.

Clip Style (Main Clipper)

Three final clipper styles are offered in the Main Clipper to enable the end user to craft his target on-air sound. All three settings offer good peak control because the main clipper is then followed by a sophisticated overshoot calculator that removes peaks that are, due to their frequency and energy content, anticipated to cause signal peak overshoots in the following low pass filters.

The *average* amount of drive to the Main Clipper is determined by the setting of both the *Final Lim/Clip Drive* control and the amount of activity occurring in the eight band limiter section. This is because increased limiting depth increases energy, which in turn can increase the duty cycle of any clipping that is occurring in the Main Clipper.

Hard – the *Hard* style has an infinite ratio with no increase in peak levels above threshold. This clipper adds brightness and density to the program.

Firm – The *Firm* style has a transfer function that allows progressively higher ratios over a 0.5dB transition region. Its operation is somewhat softer sounding than the Hard setting, but is a bit less forgiving in drive level and can therefore sound grainy on certain program material.

Round – The *Round* style has a transfer function that allows progressively higher ratios over a 1.10dB transition region. This style contributes roundness to the audio without sounding harsh but can be the most difficult mode to tune for its sweet spot. However, it can be quite powerful for certain formats when tuned correctly.

In both *Firm* and *Round* modes the ear is the best guide for finding the best settings.

Diversity Delay

The VP-8IP contains an internal digital delay to compensate for the corresponding delay in the analog side of the HD Radio codec. The delay section is capable of providing up to ten seconds of compensating delay in steps of 100 microseconds.

The *Coarse* delay is set first to align the analog and HD signals as closely as possible, and then the Fine delay control is used to trim the delay to precisely compensate for that in the HD channel. Generally it is a good idea to set the *Fine* delay to midrange before adjusting *Coarse* delay so that the *Fine* delay can then be adjusted in small increments up and down until perfect synchronization with the HD signal is achieved. Resolution of the Coarse delay is 50 milliseconds.

The *Fine* control covers a range of 50 milliseconds in 0.1 millisecond (100 microsecond) steps.

Manually Entering Delay Value

When the Fine control does not set the increment exactly to the value desired, double clicking on the delay value text box will open it for manual entry of a delay time. Note that the text box expects the manual input to be in milliseconds. Therefore, a required diversity delay setting of 8.3488 seconds would simply be entered as "8348.8."

You may also hold down the *Shift* key on your keyboard while adjusting a delay slider to affect faster adjustment rates, or hold down the *Ctrl* key to achieve finer control.

When the Delay controls are both set to their minimum settings, *OFF* appears in the delay window to signal that there is no delay at all in the circuit.

FM STEREO ENCODER MENU

FM Stereo Encoder Screen

The FM stereo encoder page contains the controls required to configure the VP-8IP's internal DSP-based precision stereo generator.

Multipath Limiter

The Stereo Encoder (Stereo Generator) contains a special algorithm called the Stereo Width Limiter. This algorithm operates in the sum/difference domain and automatically controls the amount of difference channel (L-R) amplitude as a function of the instantaneous sum channel (L+R) amplitude.

The purpose of this algorithm becomes clear when one realizes that a large amount of L-R is known to exacerbate FM multipath in some stereo receivers. We said exacerbate because a high level of L-R does not *cause* multipath. Rather, the behavior of a stereo receiver as it blends between stereo and mono becomes much *more obvious* when there is a large amount of L-R in the decoded audio, because of the acoustical summation of L+R and L-R energy in the listening environment. When the L-R signal "goes away" during a receiver blend the perceived audio level drops, and the larger the amount of L-R present when such a blend occurs, the more noticeable the multipath event will be to the listener.

When stereo enhancement is utilized in the program chain (or in the audio processor) L-R energy is naturally increased, as this is the *purpose* of stereo enhancement…to exaggerate the stereo image width by *increasing* the level of L-R energy. When taken to the extreme the extra L-R energy can cause unnatural sounding audio on some program material (the hole in the middle effect), exaggeration of reverb in the difference channel, odd behavior in most stereo receivers, or all of the above.

Another thing to remember is that in FM Stereo broadcasting each dB of L-R increase results in a like decrease in the L+R, and it is the L+R signal that is solely responsible for mono loudness. This is important to keep in mind when a fair percentage of the listening audience might be listening on mono receivers!

In order to make stereo enhancement play nice with the majority of stereo receivers in real-world listening environments with all types of program material, it is preferable to have some sort of controlling mechanism in place to manage the amount of L-R energy present in the transmitted signal as a function of program material. This is precisely what the Multipath Limiter does.

There are eleven operational choices available under the *Stereo Width Limiter* control settings: 10% to 100% in 10% percent steps, and an Off setting. Off is the setting immediately *above* the setting for 100%.

When in the Off position the Miltipath Limiter is completely out of the circuit. When the control is at the 100% position the L-R may be permitted to achieve 100% of the instantaneous L+R audio level (not recommended for most program formats). At lesser percentages the amount of L-R energy is constrained to that percentage of the L+R level at that instant. A very good starting point for this control is 70% to 80%, which constrains the L-R to 2dB to 3dB below the L+R.

With the Miltipath Limiter at this setting it will almost never touch normal stereo program material – even when stereo enhancement is used.

This is in contrast to controls in other audio processors that have options for "Stereo," "-3dB," "-6dB," and "Mono" settings within their stereo generator control menu. In those products, setting the control to -3dB causes a static 3dB reduction in stereo separation regardless of the incoming program material! The Wheatstone approach is more intelligent in how it works; it does not *decrease* stereo separation on program material that does not *need* it.

Operating the BS-412 MPX Power Controller

Near the center of the FM Stereo Encoder screen of the VP-8IP's is the control associated with the BS-412 MPX Power Controller. The control is adjustable in 0.05dB steps from +9.0 to -2.0dB.

The 0.05dB resolution of the control is to allow small adjustments in the MPX Power's reference level to compensate for unusual things a user might do to create his on air sound.

Turning the BS-412 Controller On

IMPORTANT! *The MPX Power Controller is required in certain European countries and its sole purpose is to reduce loudness and program density. If you are not required to use the BS-412 Controller it should be left in the OFF position because turning it on can cause up to a 5dB loudness loss when the control is set to its "0dB" position.*

When the BS-412 control is moved away from its OFF position the MPX Power controller is engaged and the algorithm begins measuring and controlling the processor's total MPX power according to the ITU-R BS.412-7 standard. Because the measurement is essentially an average value integrated over a very long time the MPX Power controller will *not* make quick adjustments to the MPX Power when first engaged.

As the controller integrates the MPX energy the drive to the clippers will be modified until the measured MPX power satisfies the reference level as set by the Stereo Encoder menu's BS-412 control. The control's 0dB setting conforms to the current ITU-R BS.412-7 Multiplex Power standard.

Turning the BS-412 Controller Off

If the BS-412 Controller has been on for more than a few minutes and the BS-412 control is set back to the OFF position it will take some time for the MPX Power to *return* to an unregulated state. This is because the BS-412 controller is not turned OFF by bypassing it (which could cause abrupt changes in output level!), but is instead defeated by raising its MPX Power detection threshold well above *any* conceivable or achievable audio density.

Once again … If your station is not required to comply with a Multiplex Power standard where it is licensed to operate the BS-412 control should remain in its OFF position!

MPX Lim/Clip (Limiter/Clipper) Drive

The *MPX Lim/Clip Drive* control adjusts the drive level to whichever composite processor is selected by the *Multiplex Process* check box explained later. It has a range from -6.0dB to +6.0dB.

Digital Output

Adjusts the *peak* audio signal level present at the AES/EBU digital output. The resulting peak output levels are simultaneously indicated in dBFS by the front panel and remote GUI bargraph meters. This control can be set to OFF or adjusted over the range of -35.90dBFS to 0.00dBFS in 0.05dB increments.

NOTE: This control does not adjust the level of the digital baseband192 output. See TX1/TX2 level controls.

De-Emphasis

When the *De-Emphasis* check box is selected it applies complementary de-emphasis according to any pre-emphasis setting that has been made in the Limiters screen's *Emphasis* list box.

PreDelay

Checking this box routes the AES input audio directly to the FM processing circuitry, effectively bypassing the delay.

Test Oscillator

The built in test oscillator is capable of generating sine wave test signals from 50Hz to 80kHz at modulation levels from 0 to 100%. The oscillator is ON only when the *Test Oscillator* check box is selected.

There is included a special frequency, *Bessel*, which is a 31.250kHz test signal that can be used when the first Bessel null is being utilized for modulation monitor calibration.

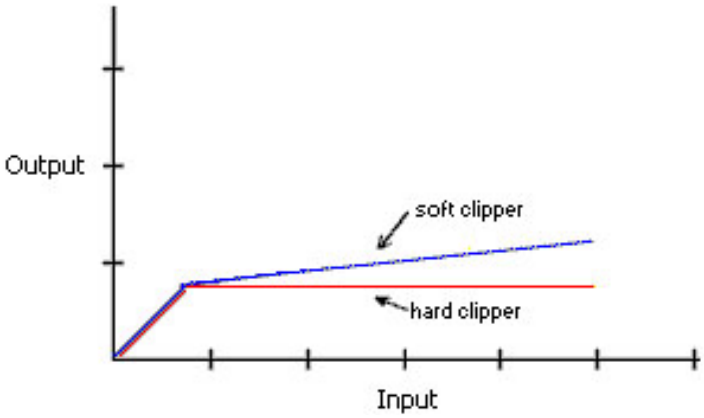
Multiplex Process

The VP-8IP provides three separate ways to process the composite stereo waveform:

Look Ahead Limiter – Uses a 0.5 millisecond look ahead and extremely high sample rates to precisely control composite waveform peaks on a cycle by cycle basis. It works similarly to the oversampled look ahead peak limiter found in the Limiters screen, but at a far higher sample rate.

Soft Clipper – Uses a rounded transfer function to gently control composite waveforms without the harsh distortion and higher order harmonics that hard clipping creates.

The Soft clipper has gain due to its 3dB transfer function, and therefore setting the Drive control to -3dB puts the Soft clipper right at the threshold of clipping with most program material, with settings higher than -3dB driving the clipper harder. Note that we do not mathematically remove the higher order harmonics in this clipper like we do in the VBMS section because such harmonics are *required* in order to make the waveform have a tightly defined peak amplitude.



Comparison of Hard and Soft Clippers

Hard Clipper – Uses a high ratio transfer function of 0.05dB to precisely control peaks. This clipper has a brighter sound than the look ahead limiter and soft clipper because it creates high order harmonic artifacts. As above, we do not mathematically remove the higher order odd harmonics in this clipper algorithm like we do in the VBMS section because such harmonics are *required* in order to make the waveform have a tightly defined peak amplitude.

Stereo Encoder

The VP-8IP contains a mathematically perfect stereo encoder whose performance is limited only by the high speed output Digital to Analog converter (DAC) and its reconstruction filter. Stereo separation is typically >50dB between 20Hz and 15kHz.

Analog Output MPX Mode

Analog Output L/R Mode

Pilot Level

The *Pilot Level* control adjusts the 19kHz stereo pilot amplitude and is adjustable from 0 (Off) to 20% of the main composite signal amplitude in 0.1% steps.

Pilot Phase

Corrections to the relative phase (timing) between the 19kHz stereo pilot and 38kHz subcarrier may be made in 0.5 degree steps up to +/- 22.5 degrees. The ability to vary this phase offset allows the VP-8IP to compensate for small time delays in the transmission system that would otherwise disturb the phase relationships between the 19kHz stereo pilot and the 38kHz stereo subcarrier and therefore cause degradation of stereo separation.

Mode

The stereo encoder can be operated in *Stereo* or *Mono* by selecting the desired operating mode with this switch. When in mono, the Stereo Pilot is turned off.

SCA 1

Inputs applied to the SCA connectors are digitized at 192kHz. Prior to being digitized the signals are high-pass filtered at 1kHz to reduce hum and then brick wall low-pass filtered at 94kHz. This control can be set to OFF or to levels from -79.95dB to +10.0dB in 0.05dB steps.

SCA 2

This control has the same functionality and range as *SCA 1* and serves the second SCA input connector on the VP-8IP rear panel.

Analog Output

When the Stereo Encoder is operated in one of the two L/R modes a single *Analog Output* control appears which adjusts both the left and right channel output amplitudes simultaneously. This control can be set to OFF or to levels from -79.95dB to 0.00dB in 0.05dB steps.

TX 1

In MPX mode, controls the output amplitude of the output signal appearing on the BNC connector for TX 1. This control can be set to OFF or to levels from -79.95dB to 6.00dB in 0.05dB steps. This control also adjusts the left channel of the baseband192 signal when that feature is enabled.

TX 2

In MPX mode, controls the output amplitude of the output signal appearing on the BNC connector for TX 2. This control can be set to OFF or to levels from -79.95dB to 6.00dB in 0.05dB steps. This control also adjusts the right channel of the baseband192 signal when that feature is enabled.

Pilot Only

When checked, converts the TX 2 output to a line-level 19kHz sine wave source.

Analog Output and MPX Modes

Please pay careful attention to the following analog output description as it is unlike that found in <u>any</u> other broadcast audio processor!

L/R – In this mode the twin output level controls seen in MPX mode (see below) disappear and are replaced by a single output level control governing both left and right channels. In this mode the male XLR outputs as well as the female BNC connectors assigned to TX 1 and TX 2 *both* carry line level audio. The XLR outputs are electronically balanced and the BNC connectors are single-ended unbalanced with a level 6dB below that on the balanced outputs. If pre-emphasis has been selected in the FM Limiter screen then both sets of outputs are pre-emphasized also.

L/R deemph – In this mode the twin output level controls seen in MPX mode (see below) are replaced by a single output level control governing both left and right channels. In this mode the male XLR outputs and the female BNC connectors assigned to TX 1 and TX 2 *both* carry line level audio with the XLR outputs being electronically balanced and the BNC connectors single-ended unbalanced. If pre-emphasis has been selected in the FM Limiter screen then both sets of outputs are de-emphasized according to the pre-emphasis time constant setting.

MPX – When selected, there are *two* output level controls present, one for TX 1 and one for TX 2. While in this mode the male XLR outputs *and* the female BNC connectors assigned to TX 1 and TX 2 carry <u>wideband, line level composite stereo audio</u> and, as in the L/R case above, the XLR outputs are electronically balanced. These balanced outputs can be used to gain a signal to noise advantage with exciters having a balanced composite input (most modern ones do).

The adjustment range of the analog outputs, regardless of the operating mode, is -80dB to 0dB (6dB in MPX mode) in 0.05dB steps, covering virtually every conceivable operating need in the field.

NOTE: A 0dB setting of the output level controls results in a +22dBu PEAK output level at the *balanced* outputs!

VP-8IP AM Process Mode

The following discussion is unique to the VP-8IP Audio Processor when it is operating in its *AM* mode.

There are two menu screens pertinent to the *AM* mode; they are used for adjusting how the processor's final limiting scheme sounds as well as how it interfaces with external transmission equipment. These are the *Limiters* and *AM* (Transmission) screens, the latter being visible and selectable via the *AM* tab only when the VP-8IP is in its *AM* mode.

Audio processing while in *AM* Mode takes place in the Sum and Difference domain, which means the VP-8IP is fully compatible with legacy analog AM stereo systems that require separately limited Sum and Difference audio. For L/R Stereo applications the outputs are derived from a mathematically perfect L+R/L-R to L/R matrix. This also means that the VP-8IP has perfect mono compatibility because in Mono mode the L-R channel is simply discarded.

Note: In ALL VP-8IP AM presets the default settings in the *AM Transmission* Screen are as follows:

Bandwidth	10K NRSC
Output Mode	Mono
Positive Modulation	124%
Digital Output Level	-6dBFS
Digital De-emphasis	Off (Unchecked)
Digital Pre-Delay	Off (Unchecked)
Analog 1 Level	-14dB (= +10dBu Peak)
Analog 1 Phase	+
Analog 1 LF Tilt F	20.0Hz
Analog 1 LF Tilt Q	1.06
Analog 1 LF Tilt L	0.0dB
Analog 2 Level	-14dB (= +10dBu Peak)
Analog 2 Phase	+
Analog 2 LF Tilt F	20.0Hz
Analog 2 LF Tilt Q	1.06
Analog 2 LF Tilt L	0.0dB

LIMITERS MENU - AM MODE

Limiters Screen in AM Mode

Multiband Limiter

The VP-8IP contains an exclusive and fully adjustable eight band AM final limiter algorithm. This algorithm is especially suited for controlling peak energy without generating unnatural swishing and "phasey" artifacts that outdated and simplistic algorithms with fewer bands generate.

There is no caveat for using the eight band limiter for AM other than it is probably inappropriate to drive the limiters continuously beyond 10-12dB of indicated gain reduction. We recommend operating the limiters in a manner that allows them to do three things: build RMS energy to maximize loudness, shape the HF spectrum to make it more suitable for the AM medium, and control longer duration transients that the preceding four band AGC/compressor misses.

For best musicality we also recommend that the limiters not be operated with extremely short attack and release times. Please open several of our Factory AM presets and observe the settings in the eight band limiter section to see what we mean.

The eight band limiter's operating parameters are fully adjustable in order to accommodate a wide variety of program material and competitive needs. It also contains the embedded Bass Tools which is tightly integrated within the limiter structure to intelligently control bass program energy.

Multiband Enable

This check box determines whether or not any processing is applied by the eight band multiband limiter. When the eight band section is not enabled, audio peak control is handled solely by the main clipper (which may or not be desirable!).

Multiband Drive

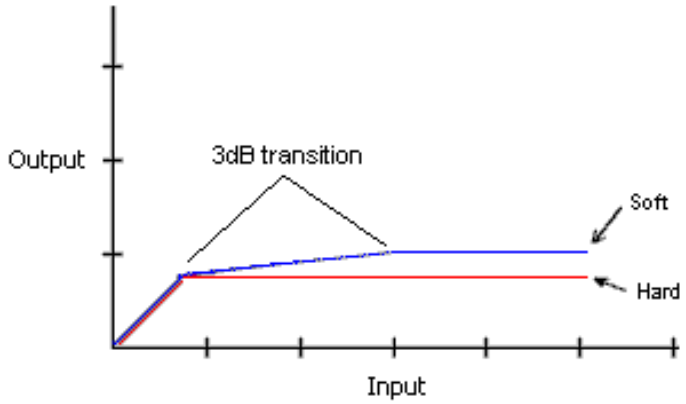
This control determines the amount of input signal applied to the eight band limiter and therefore how hard it is being driven. Less *Multiband Drive* will create a more gentle sound while still controlling peaks, while higher settings make the sound more aggressive by increasing short term audio density. Very high *Multiband Drive* settings that result in more than 12-15dB of continuous gain reduction are typically unnecessary with most AM formats.

The *Multiband Drive* control setting is also somewhat dependent upon the setting of the *Makeup* control in the four band AGC. For this reason, the control is not labeled in dB Drive, but instead is labeled in percent (%). With sane settings of the four band AGC's *Makeup* control the correct eight band limiter *Multiband Drive* control setting will usually be between 85% and 100%. Full adjustment range is 0% to 100%.

Higher *Multiband Drive* settings cause deeper gain reduction in the eight band limiter (as well as higher levels of clipping in the main clipper algorithm if the *Lim/Clip Drive* is not first reduced before the *Multiband Drive* is advanced). Therefore, the *Multiband Drive* control is one of a pair of controls most useful for trading off "loudness," "density," and "distortion."

Multiband Knee

Each of the eight bands' threshold, attack, and release are adjustable, as is the overall transfer function (knee hardness). The knee shape determines how the limiters behave at or near their thresholds of limiting. The graphic below illustrates the difference between the two transfer functions.



Comparison of Hard and Soft Limiter Modes

In the *Hard* mode, limiting does not occur until the threshold within a band is precisely reached. The sound of this setting is more open and free because no limiting activity occurs *until* the threshold is achieved.

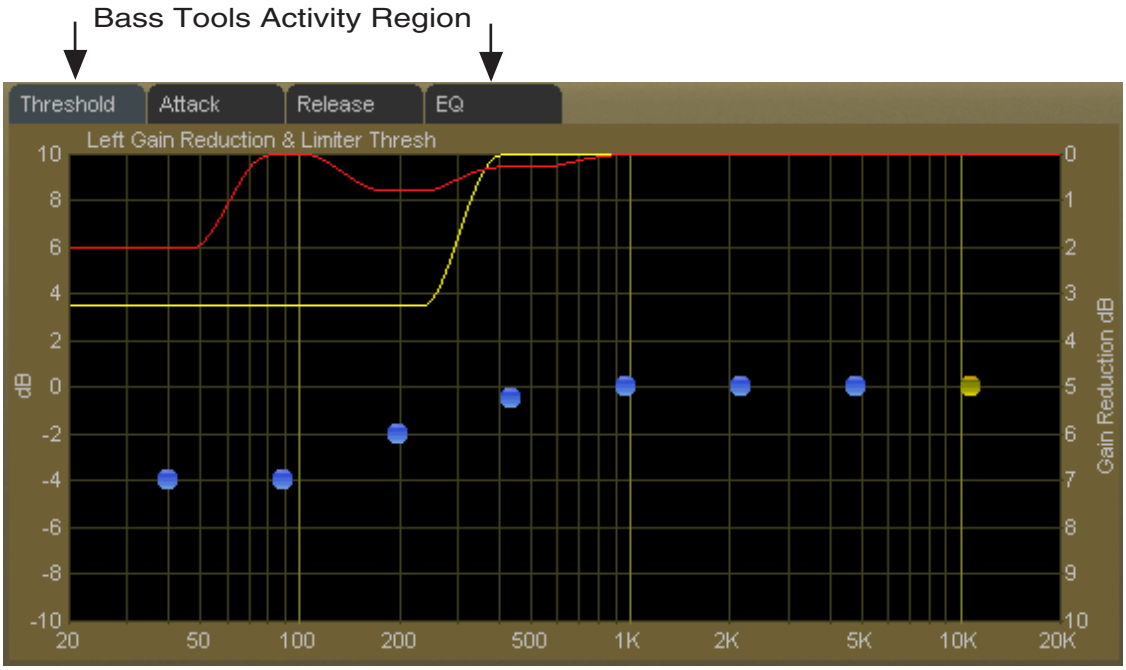
In the *Soft* mode, limiting starts to occur at a level that is 3dB below the actual limiter threshold setting. The transfer function at this point is relatively soft. That is, the audio waveform is not abruptly limited, but instead is allowed to gradually increase over a 3dB window until the ultimate limit threshold is reached, whereupon no further amplitude increase is allowed.

Because of its behavior, the sound of the *Soft* function is more obvious and has a "spongy" sound when in operation. It is also less precise as a true limiter until the ultimate limit threshold (the setting of the band threshold) is finally reached.

Bass Tools

Vorsis Bass Tools allows extremely low program frequencies to be processed without causing dynamic gain changes and without generating undesirable intermodulation artifacts in higher frequency signals. Its primary purpose is to allow the thumps of kick drums and other percussion instruments to sound natural in their "percussiveness" without pumping or other undesired artifacts. The algorithm operates in left/right mode treating both channels identically.

Our laboratory research on codec behavior rather surprisingly revealed that it is perfectly acceptable to process very low frequency material using the Bass Tools algorithm provided that the harmonics generated during the process are well controlled spectrally. Since our bass algorithm does not allow significant harmonic levels above the third or fifth (somewhat dependent the Bass Tools *Style* setting) there is no audible downside to using it with coded audio. Furthermore, the ability to utilize the Bass Tools on both the FM and HD sides of the processor allows the sound to be more evenly matched as receiver-based digital to analog crossfades are encountered.



Bass Tools Activity Display - 20Hz up to VBMS Frequency Setting

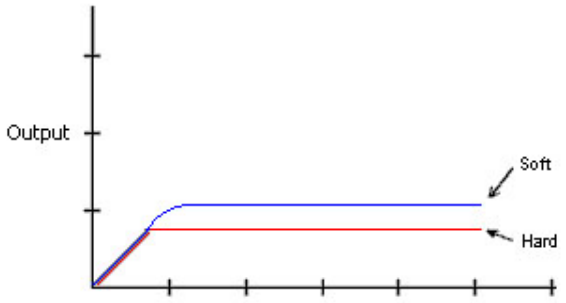
Drive

This control adjusts the amount of Input Drive to the Bass Tools algorithm. Higher drive settings make it work harder and generate higher levels of harmonics. The *Drive* control adjusts the amount of drive to the Bass Tools and is adjustable over a range of -6dB to +12dB. Adjustment is best done by ear with percussive source material.

Style

Two Bass Tools operating styles, plus Off, are available. The *Hard* setting has a transfer function occupying approximately 1dB of transition amplitude. The waveform control is fairly abrupt and generates low order harmonics.

The *Soft* setting operates with a transfer function occupying approximately 2dB of transition amplitude. Therefore the control is less sharp than the hard setting, generating more low order harmonics. While the two settings may appear to be almost the same, the 1dB difference in transfer ratio creates a *significant* difference in how the two Bass Tools styles *sound* on actual program material. When Off is selected, the Bass Tools processing is bypassed.



Comparison of Bass Tools Operating Modes

Frequency

The *Freq* control is simple. All audio below the frequency selected is processed by Bass Tools. Smoother bass can be had with settings above 200Hz, which is recommended.

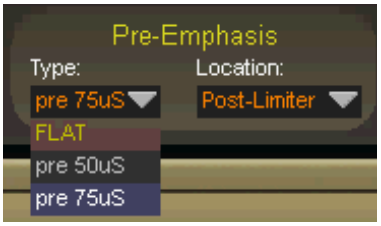
Output

The Bass Tools *Output* control adjusts the output of the Bass Tools algorithm and can be thought of as a "low bass mix" control. Settings between -6dB and -3.0dB add extra bass to the overall mix, while settings lower than -6.0dB reduce it. Both the Bass Tools *Drive* and the *Output* controls are best adjusted by ear using percussive bass material. The *Output* control is adjustable between -12dB and 0dB.

Pre-Emphasis

Type

Pre-emphasis is selectable as 50 or 75 microseconds, or Flat (OFF).



Location

When pre-emphasis is enabled it may be placed in one of three locations in the signal chain:

Pre-MBLim – In this position, the selected pre-emphasis is placed *before* the eight band limiter/clipper section. In this mode the eight band section may be operated as a frequency conscious limiter tailored to the inverse of the selected pre-emphasis.

Pre-Limiter – In this position the pre-emphasis is *after* the eight band section but *before* the Look Ahead limiter (if enabled) and/or main clipper.

Post-Limiter – In this position the pre-emphasis is *after* the eight band section and Look Ahead limiter and *before* the main clipper.

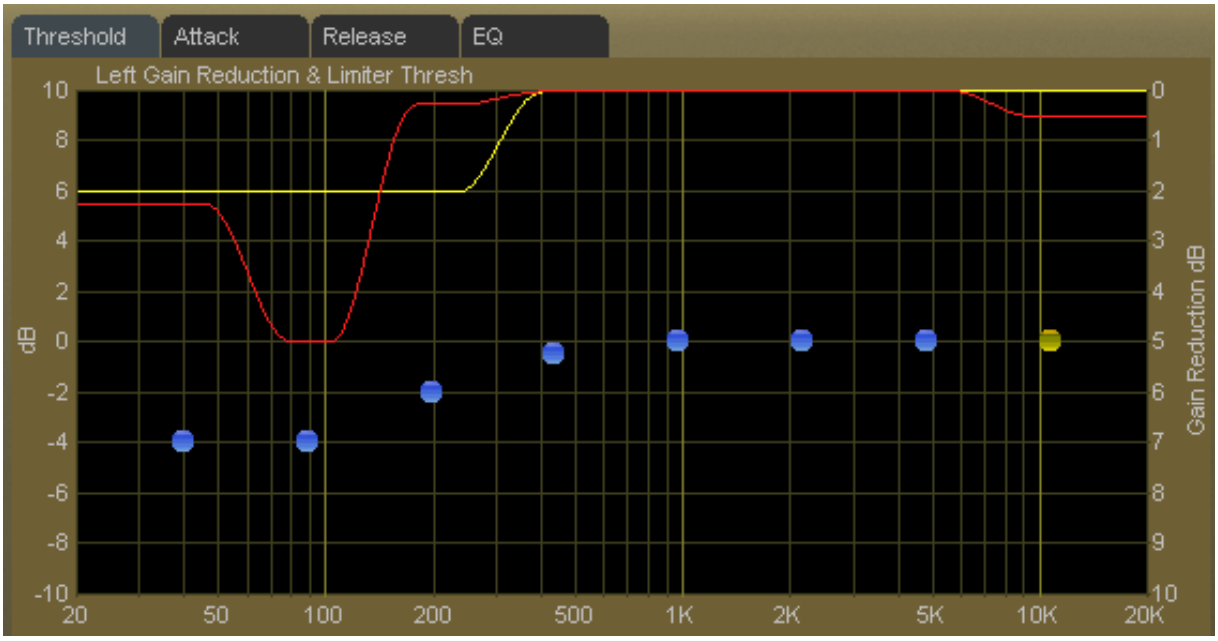
Entirely different sound textures may be created by placing the pre-emphasis in one location versus the other. There is no right or wrong setting unless that setting does not produce the on air sound desired.

Note that in our AM presets we use a combination of the NRSC pre-emphasis in the AM Limiter screen and parametric equalization prior to the four band AGC to create satisfactory subjective brightness and intelligibility on an average AM receiver.

Multiband Limiter Adjustments

Threshold Screen

Clicking on the *Threshold* tab on the graphic opens up the adjustment screen for the limiter thresholds for each of the bands.



Multiband Limiter Threshold Screen

The thresholds are adjustable over a +/-10dB range. Because the thresholds are adjustable over a wide range, frequency conscious limiting can be created if needed.

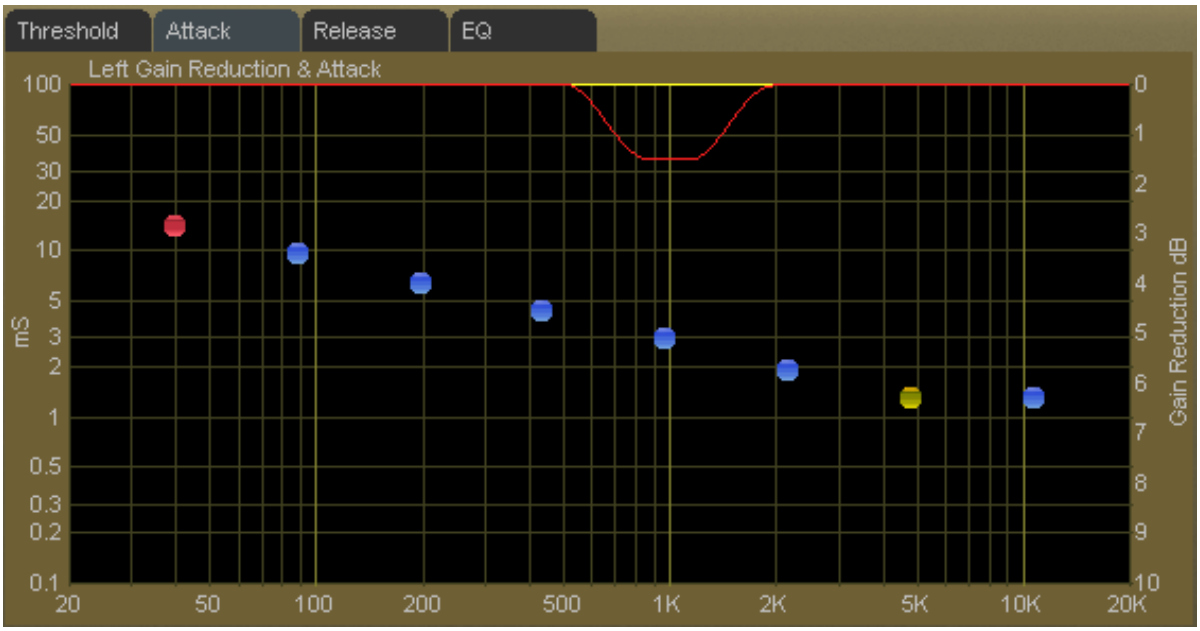
Each of the limiter thresholds can be adjusted separately or in groups according to the manner described in the earlier *VP-8IP Graphical Interface Operation* section. Lower settings (more negative numbers) reduce the threshold, causing limiting to occur at lower audio levels. Conversely, higher thresholds cause limiting to occur at higher audio levels.

Because of the frequency response contouring in AM pre-emphasis it is generally customary to adjust the limiter thresholds in somewhat of a sloping fashion according to the limiting sensitivity curve required, which can create a multiband frequency conscious limiter of any configuration.

Note: Radically different settings between adjacent bands won't cause strange sounds – there is no swishing in the Vorsis multiband limiter algorithm. Operationally, though, it doesn't make much sense, except perhaps at the lower frequencies, where radically slower release times may be required to tame unnatural bass energy buildup when deep limiting is being used.

Attack Time Screen

Clicking on the *Attack* tab opens up the attack adjustment screen.



Multiband Limiter Attack Screen

The attack time of a limiter is a measure of how fast it responds to an audio peak once the peak exceeds the limiter threshold. Shorter attack times cause the limiter to respond to audio peaks quicker, which affords more accurate peak control; excessively short attack times, while creating "perfect" peak control, can literally suck the life out of musical material.

At the other extreme, excessively long attack times can allow audio peaks to escape the limiters largely unmanaged, leaving them to be controlled solely by the final look ahead output limiter (if enabled) or the final clipper.

The attack times for each band are adjustable from 0.1 milliseconds (100 microseconds) to 100 milliseconds. As explained above, faster attack times exert more control over audio peaks, while slower attack times allow peaks to escape the limiter section with less control.

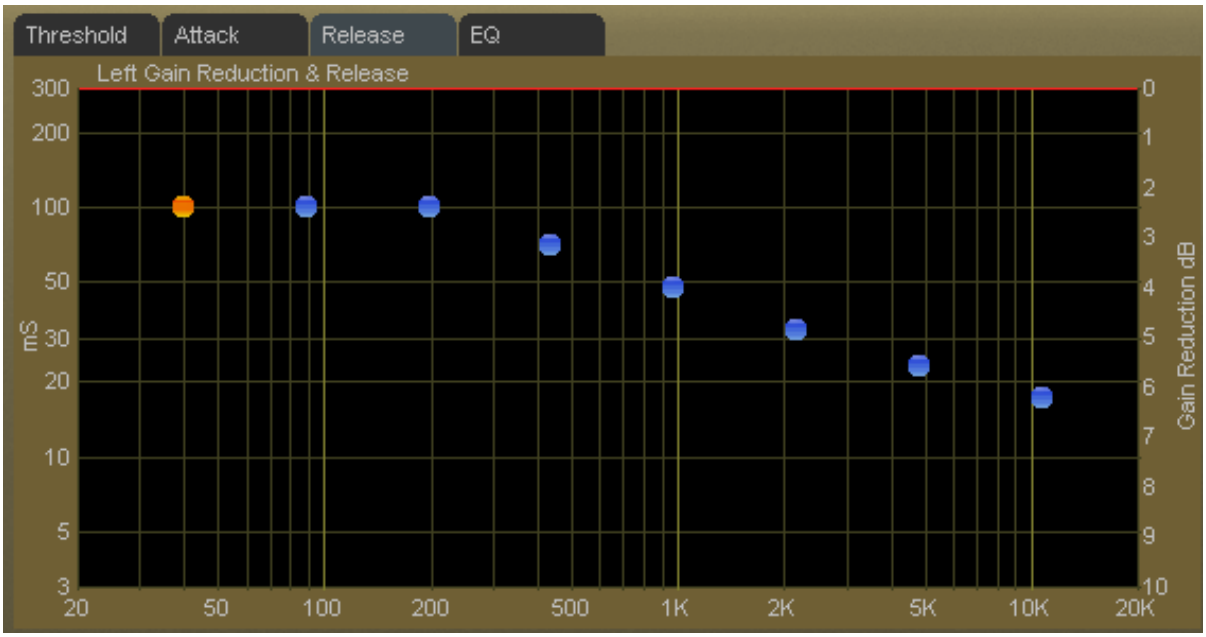
Attack times are generally adjusted to taste, keeping in mind that the slower the attack time the more work the final clipper or limiter must do, while conversely the faster the attack time the fewer short term dynamics the program material will have.

Generally, lower frequencies require slower attack times and higher frequencies require faster attack times for a given amount of control.

The attack times of each band may be individually or collectively modified using the GUI operating instructions provided earlier.

Release Time Screen

Clicking on the *Release* tab opens up the release time adjustment screen.



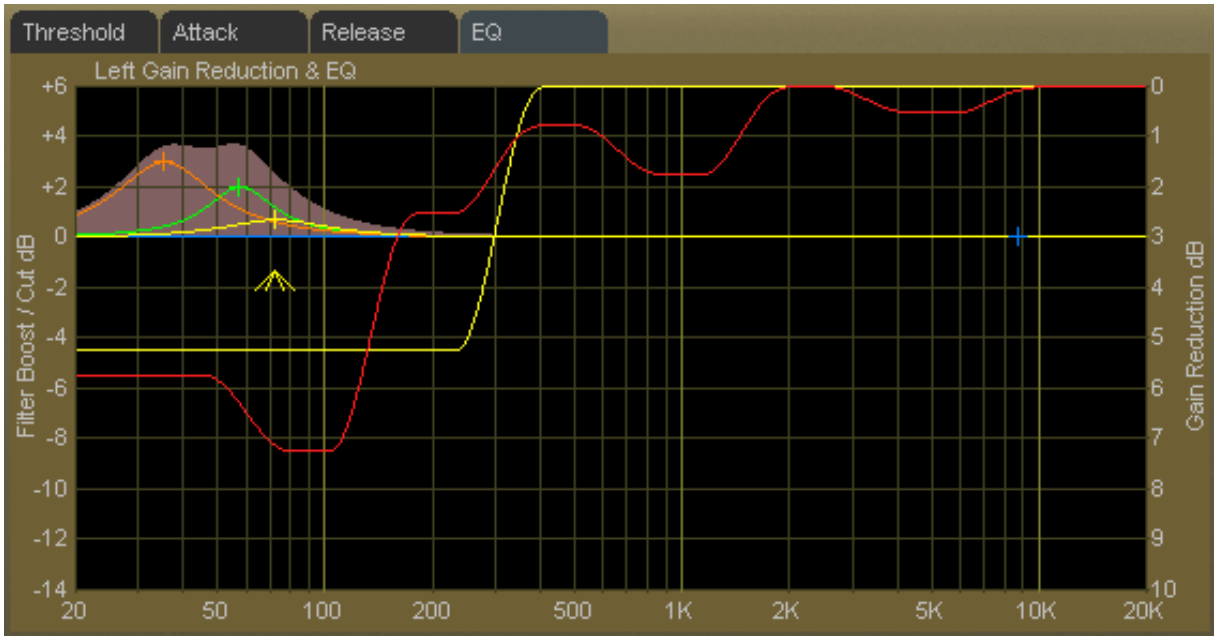
Multiband Limiter Release Time Screen

The release time of a limiter determines how fast it recovers from an audio peak that has caused a momentary decrease in gain. Shorter release times create a denser sound, and excessively short release times might increase intermodulation distortion (the mixing of low and high frequency energy – or a muddying of the sound). The multiband limiters all have Delayed Release timing built in to minimize unintentional intermodulation distortion.

The release times for each band are adjustable from 300 milliseconds down to 3 milliseconds. Faster release allows a more rapid recovery after a limiting event, while slower release times do the opposite. Release times should be adjusted to taste, keeping in mind that the slower the release time the more dynamic the sound, but the more likely that audible ducking may be heard.

Final Equalization Screen

Clicking on the *EQ* tab opens up the final equalization screen.



Multiband Limiter Equalization (EQ) Screen

Under certain conditions it may be desired to *slightly* alter the sound exiting the final eight band limiter. A small amount of equalization performed after limiting can open up the sound or provide a slight enhancement to its overall feel, and minor equalization tweaks can be accomplished after the limiters without radically disturbing the peak levels emerging from the eight band algorithm.

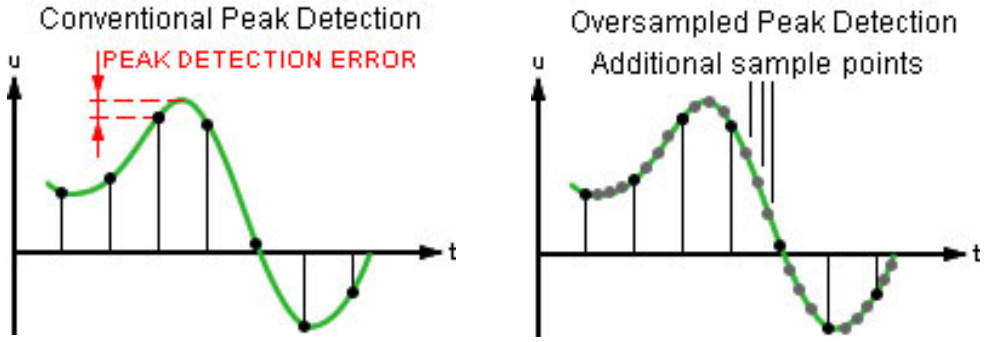
Only very small amounts of boost equalization are recommended once final limiting has been accomplished in the eight band section. In fact, the adjustment range has been limited to +3dB/-12dB in order to curtail the temptation to perform aggressive equalization where it is the most inappropriate, and even this range is quite large considering the effect that can be created.

Small adjustments in EQ boost – on the order of 1dB or 2dB at most – are sufficient to alter the overall tonality of the eight band section.

Equalization boosts must be approached with caution because they can add significant energy that must be controlled later by either the look ahead limiter or final clippers. On the other hand, equalization cuts can be made as desired because they do not add energy.

AM Look Ahead Limiter

If desired, the AM output may be peak controlled by a highly oversampled feed forward limiter with a 0.5 millisecond look ahead time. The control path (not the audio path) is highly oversampled in order to achieve extremely precise peak control. The primary goal of oversampling the control sidechain is better peak control through finer definition of the audio waveform. Therefore, oversampling of the audio signal itself is not only unnecessary, but wastes DSP resources. The graphic below illustrates why control signal oversampling is more effective.



By sampling the audio waveform with finer resolution we can detect peak excursions that would be missed otherwise. In fact, if both the audio and control signals were oversampled peaks would still be missed because their relative time alignment would be identical!

AM Look Ahead Limiter Controls

The graphic below shows the controls available for the look ahead limiter. The following text describes each control according to function.

Look Ahead Limiter Enable

When the *Look Ahead Limiter Enable* is checked the look ahead limiter is enabled. Likewise, when this check box is not checked, final peak limiting is via the distortion cancelled clipper only.

The look ahead limiter and distortion cancelled clipper *may* be used simultaneously if desired, though precisely how to adjust everything when both are operating simultaneously becomes "artistically-driven." Adjustment can get tedious and how to do it is even harder to explain in writing. Therefore we feel it is not only beyond the scope of this manual, but also falls into the realm of "if you really want to do this please call us for advice."

L/R Linked

When this box is checked the look ahead limiter's left and right channel control signals are common. This prevents unnatural stereo image shifts when one channel needs to reduce the signal in order to control peak excursions.

Final Lim/Clip (Limiter/Clipper) Drive

This controls the amount of drive to the distortion-controlled clipper and/or look ahead limiter, as chosen. The gain range of this control is +12dB/-18dB. The resolution of this control is 0.01dB, fine enough for the most exacting clipper drive adjustments.

Attack

Adjusts the attack time of the look ahead limiter. Because the look ahead is 0.5mS, an attack time setting of 0.2mS creates a limiter that attacks 0.3mS before the peak arrives at the limiter's output port. Likewise, a setting of 0.5mS creates a limiter with zero attack time (because the look ahead is 0.5mS). All settings above 0.5mS create a conventional limiter with the control calibration skewed a negative 0.5 millisecond because of the look ahead time. The full range is 0.2mS to 100mS.

Release

Controls the primary (fast) release time of the look ahead limiter.

Slower settings (longer release times) cause the peak limiter to recover more slowly from limiting; release times set too slow may allow undesirable signal ducking to occur.

Conversely, fast release times increase signal density, and release times that are too fast result in increased intermodulation distortion which can "muddy" the sound.

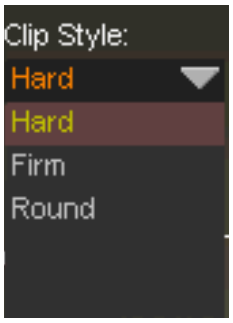
The overall range of the control is from 33 milliseconds to 330 milliseconds.

Delayed Release

Controls the secondary (slow) release of the look ahead limiter, and is adjustable from 100 milliseconds to 1 second. The *Delayed Release* control has no effect unless the *Enable* check box is checked (see below).

When Delayed Release is engaged via the *Enable* check box, the limiter will release the first 3dB of gain reduction at the main Release setting, with the remainder of the release occurring at the Delayed Release setting. Delayed Release affords a drastic reduction in intermodulation distortion and generally allows more limiting depth without incurring an intermodulation distortion penalty.

Clip Style (Main Clipper)



Three final clipper styles are offered in the Main Clipper to enable the end user to create the sound desired. All settings offer good peak control because the main clipper is then followed by a sophisticated overshoot calculator that removes peaks that are, due to their frequency and energy content, anticipated to cause signal peak overshoots in the following low pass filter.

The *average* amount of drive to the Main Clipper is determined by the setting of both the *Final Lim/Clip Drive* control and the amount of activity occurring in the eight band limiter section. This is because increased limiting depth increases energy, which in turn can increase the duration of clipping occurring in the Main Clipper.

Note: Older transmitters may be "happier" using something other that the *Hard* clip style. Because the clipping is less abrupt in *Firm* and *Round* these styles stress the transmitter less, and this is particularly true for the passive components in the transmitter's output network and the antenna tuning unit or phasing system.

Hard – The *Hard* style has an infinite ratio with no increase in peak levels above threshold. This is the most "forgiving" of the clip styles as far as clipper drive levels are concerned and adds brightness and density to the program.

Firm – The *Firm* style has a transfer function that has progressively higher ratios over a 0.5dB transition region. Its operation is somewhat softer sounding, not in loudness but in texture, than the Hard setting. The waveforms exiting this clipper style contain less high-order harmonics and may be easier to handle for older transmitters or antenna systems.

Round – The *Round* style has a transfer function that has progressively higher ratios over a 1.0dB transition region. Its operation is even softer sounding (not in loudness but in texture) than the Firm setting, and the waveforms exiting this clipper style contain even less high-order harmonics than the *Firm* style and may be even easier to handle for older plate modulated transmitters or narrow band antenna systems.

This style contributes roundness to the audio without sounding harsh and can be quite powerful for certain AM formats when tuned correctly.

We offer the *Firm* and *Round* modes for two reasons: (1) they create a different on air sound, a sound that may be suitable for a particular application, and (2), they may be easier on the transmission system and its passive components than the *Hard* clipper style.

Diversity Delay

The VP-8IP contains an internal digital delay to compensate for the corresponding delay in the HD Radio codec. The delay section is capable of providing up to ten seconds of compensating delay in steps of 100 microseconds.

The *Coarse* delay is set first to align the analog and HD signals as closely as possible, and then the *Fine* delay control is used to trim the delay to precisely compensate for that in the HD channel. Generally it is a good idea to set the *Fine* delay to midrange before adjusting *Coarse* delay. That way the delay can be adjusted in small increments up and down until perfect synchronization with the HD signal is achieved. Resolution of the Coarse delay is 50 milliseconds.

The *Fine* control covers a range of 50 milliseconds in 0.1 millisecond (100 microsecond) steps.

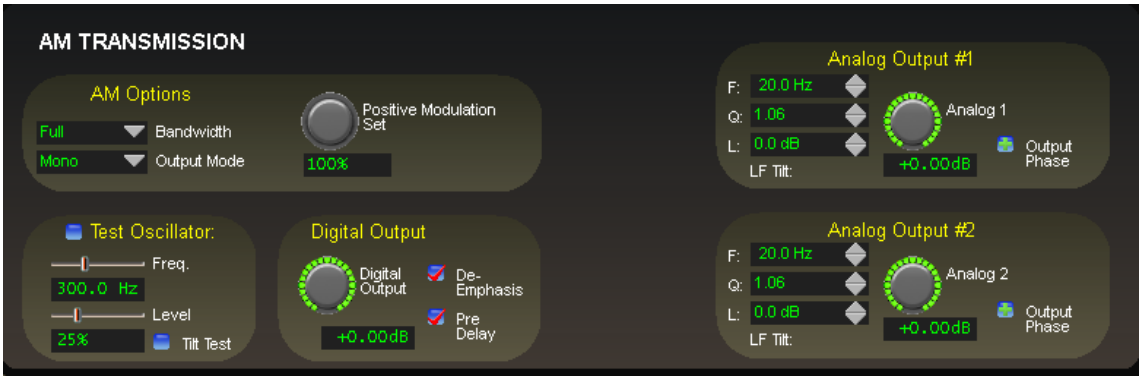
When the Delay controls are both set to their minimum settings, *OFF* appears in the delay window to signal that there is no delay being added.

Manually Entering a Delay Value

You can double click on the delay value window to directly enter a numeric delay value in milliseconds. This is useful when you know roughly how much delay is required or wish to try various values quickly. You may also hold down the Shift key on your

keyboard while adjusting a delay slider to affect faster adjustment rates, or hold down the Ctrl key to achieve finer control. If entering a delay time manually keep in mind that the text box expects the manual input to be in milliseconds, so a delay of 8.3488 seconds would be entered as “8348.8.”

AM TRANSMISSION MENU



AM Transmission Screen

This screen contains all of the controls necessary to select the correct output operating modes of the VP-8IP, as well as the controls for transmitter equalization. It also contains the output control for the Digital AES3 output of the analog side of the processing chain.

Test Oscillator

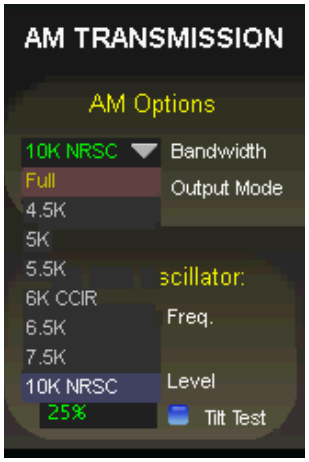
The built in test oscillator is capable of generating sine wave test signals from 25Hz to 12kHz at modulation levels from 0 to 100%. The oscillator is On only when the *Test Oscillator* check box is selected.

Tilt Test

When *Tilt Test* is checked a clipped sine wave at 50% amplitude is produced that can be used to properly set the low frequency tilt correction for older plate modulated transmitters that require it.

Bandwidth

Offers eight Low Pass Filter selections typically required for AM and Medium Wave applications with the eight selections being:



Full: AM Bandwidth is unrestricted out to 20 kHz.

4.5K: Restricts the output spectrum to 4.5 kHz.

5K: Restricts the output spectrum to 5 kHz.

5.5K: Restricts the output spectrum to 5.5 kHz.

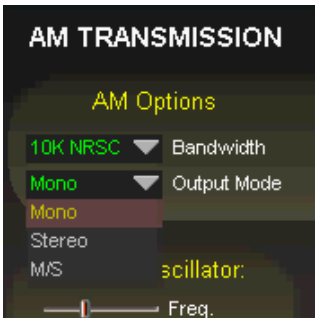
6K CCIR: Restricts the output spectrum to 6 kHz according the CCIR specifications.

6.5K: Restricts the output spectrum to 6.5 kHz.

7.5K: Restricts the output spectrum to 7.5 kHz.

10K NRSC: Restricts the output spectrum to 10 kHz as required by the NRSC-1 specification.

Output Mode



Allows the selection of one of the three possible output modes:

Mono: In *Mono* mode two independent transmitter outputs are provided, each with its own adjustable LF Tilt Equalization and Output Polarity control.

Stereo: In *Stereo* mode there is one output control that adjusts the left and right output channels simultaneously. There is also one LF Tilt Equalizer available that affects both channels.

Stereo M/S: In *Stereo M/S* mode the two outputs are dedicated to the Sum and Difference (L+R and L-R respectively) outputs of the limiter. In *Stereo M/S* mode the "left" output carries the L+R signal while the "right" carries the L-R signal.

One LF Tilt Equalizer affects both Sum and Difference channels (see Note #1, below).

Note #1 – L-R Level While in Stereo M/S Mode

If it is desired to alter the L-R channel's audio level when operating the VP-8IP's outputs in *Stereo M/S* mode, it can be accomplished by adjusting the individual band's *L-R Mix* controls located within the graphical area. A +/-6dB adjustment range has been provided.

Positive Modulation

This control adjusts the asymmetry of the VP-8IP's final clipper, which adjusts the amount of positive modulation compared to a fixed negative value. The VP-8IP's AM mode is capable of providing dense modulation up to the 150% limit of its *Positive Modulation* (Asymmetry) control, and peak control in both negative and positive directions is quite good at all settings.

Caveat #1: *It needs to be mentioned that unless the transmitter is either (1) a grossly over designed plate modulated rig, or (2) is of recent design and has one of the newer modulation schemes and the power supply and the RF power headroom to match, positive modulation above 100% should be approached with caution. Trying to force an inadequate AM transmitter to modulate above 100% in the positive direction may cause adjacent channel interference at one extreme and possibly expensive transmitter damage at the other.*

Remember also that the Positive Modulation *control affects all output channels simultaneously, regardless of the output configuration!*

Caveat #2: *The VP-8IP's analog output amplifiers have a maximum peak output level capability of +24dBu. This means that if you wish to operate with the* Positive Modulation *control cranked open all the way to 150% then the peak transmitter input level required for 100% modulation must be no more than +20.4dBu. Luckily most AM transmitters require approximately +10dBu for 100% modulation which leaves plenty of headroom.*

Peak Polarity

If you observe that the negative modulation increases as you advance the *Positive Modulation* control above 100% the output polarity between the VP-8IP and the transmitter

is mismatched! You can change the polarity by either reversing the analog audio pair in the cable between the VP-8IP and the transmitter, or more simply, flip the audio phase by clicking the *Output Phase* button next to the *Output Analog #1* or *Output Analog #2* output level control (if in a *Stereo* mode).

Output Level Setting

Many AM transmitters with analog inputs require an audio input level of approximately +10dBu to achieve 100% modulation. An output level control setting of 0dB on the VP-8IP results in a +24dBu PEAK output level at its balanced output!

Analog Output #1

Controls the output amplitude of the analog output signal appearing on the left channel XLR on the rear panel. This control can be set to OFF or to levels from -79.95dB (or -59.95dBu) to 0.00dB (or +24.0dBu) in 0.05dB steps.

The output level control should be adjusted to provide the desired negative modulation percentage. Positive modulation (asymmetry) may be set using the *Positive Modulation* control.

Analog Output #2

Visible only in *Mono* mode, it controls the output amplitude of the analog output signal appearing on the right channel XLR connector on the rear panel. This control can be set to OFF or to levels from -79.95dB (or -59.95dBu) to 0.00dB (or +24.0dBu) in 0.05dB steps.

The output level control should be adjusted to provide the desired negative modulation percentage. Positive modulation (asymmetry) may then be set using the *Positive Modulation* control.

LF Tilt Correct

Low Frequency Tilt Correction is used to correct (either fully or partially) for the low frequency response inadequacies of primarily older style AM transmitters. In such transmitters, inter-stage transformers and/or coupling capacitors limit the frequency response at low frequencies due to their high-pass, DC-blocking nature.

Because high waveform fidelity is required for good peak control, and because transmitters with such inadequacies do not have frequency response extending down to near DC, the clipped waveforms generated by the audio processor will not pass through such transmitter without undergoing low frequency losses. These losses cause the waveform to tilt, which can cause the top of the clipped waveform to exceed 100% modulation at low frequencies. In order to prevent over modulation on such waveforms the audio level into the transmitter must be reduced, and this results in lost loudness and coverage.

LF Tilt Correction is designed to *pre-distort the shape* of the waveform generated by the processor in order to make it "complimentary" to what the transmitter will do to the waveform later. In essence, LF Tilt Correction is a specialized, extremely low frequency

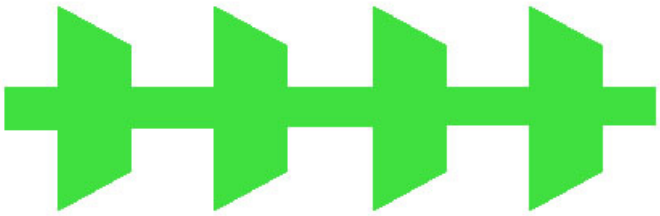
equalizer that tilts the waveform in the direction opposite to how the transmitter will tilt it later, and the result is flatter tops of the clipped waveforms, and therefore more accurate modulation of the transmitter.

Note that the LF equalization generated by the tilt equalizer is NOT intended to boost low frequency energy in program material to make it more audible to the ear. Rather, LF Tilt Correction occurs in the subsonic audio region, typically between 10Hz and 50Hz.

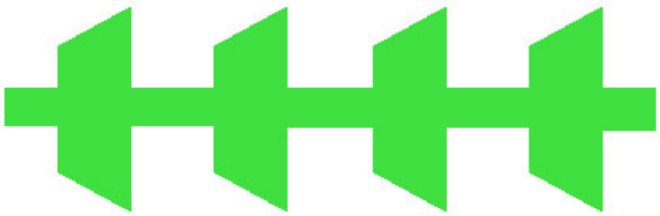
Adjusting the Tilt Corrector

Adjusting the AM Tilt Corrector is a simple process. The first step is to determine if the transmitter will require Tilt Correction in order to modulate well at low frequencies and the following procedure will determine this:

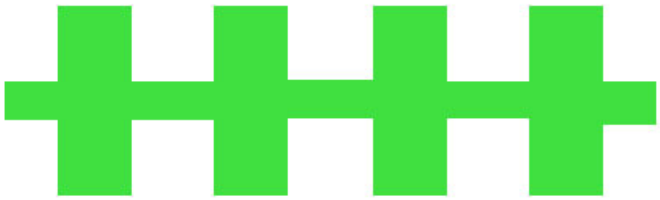
1. Connect the VP-8IP to the audio input of the transmitter to be tested.
2. Connect the transmitter to either a dummy load of rated power or to the antenna system.
3. Turn on the VP-8IP's Test Oscillator. Adjust it to a frequency of 75Hz. Adjust the *Test Oscillator Level* control and the *Analog 1* or *Analog 2* output (as appropriate) until approximately 80% modulation is indicated on the station's modulation monitor. Ensure that the Tilt Correct *L* (level) controls are set at 0dB.
4. Turn on the VP-8IP Test Oscillator *Tilt Test* by clicking its check box. Using an oscilloscope connected to a low level RF sample of the transmitter output observe the resulting RF waveform. It should appear like waveform A or waveform B below, or somewhere in between the two extremes, depending on the response flatness of the transmitter's low frequency performance.



A - Excessive Low Frequency Tilt



B - Excessive Low Frequency Tilt Correction!



C - Proper Tilt Correction OR Good Low Frequency Performance

5. If the response of the transmitter does not resemble Figure C, alternately adjust the Tilt Corrector's *F* (Frequency), *Q* (Bandwidth), and *L* (Level) controls in order to achieve the flattest top and bottom of the waveform. The ideal case would be to have the waveform look exactly like Figure C above, but sometimes perfect correction is not possible, particularly on older plate modulated transmitters with inter-stage coupling transformers* (see below).

Please note, it is entirely possible to overcorrect for tilt. If the transmitter response starts looking like Figure B you have gone too far.

* On older transmitters that cannot be fully tilt corrected it is not recommended to operate them with gross amounts of final clipping in the processor, large amounts of bass equalization, and positive modulation beyond 100%. Doing so (or attempting to do so) can overstress components in the transmitter leading to overheating and damage (or much worse).

Digital Output

Adjusts the *peak* audio signal level present at the AES/EBU digital output. The resulting output levels are simultaneously indicated in dBFS by the front panel and remote GUI bargraph meters. This control can be set to OFF or adjusted over the range of minus 35.90dBFS to 0dBFS.

In many recent AM transmitters with AES3 digital inputs 100% modulation is defined as -6dBFS, which means that enough headroom is (presumably) available in the transmitter to reach 200% positive modulation before its digital section "runs out of bits."

De-Emphasis

When checked, applies complimentary de-emphasis to the AES3 digital output.

PreDelay

When checked, the *AES Output* signal is taken *before* the *Diversity Delay*.

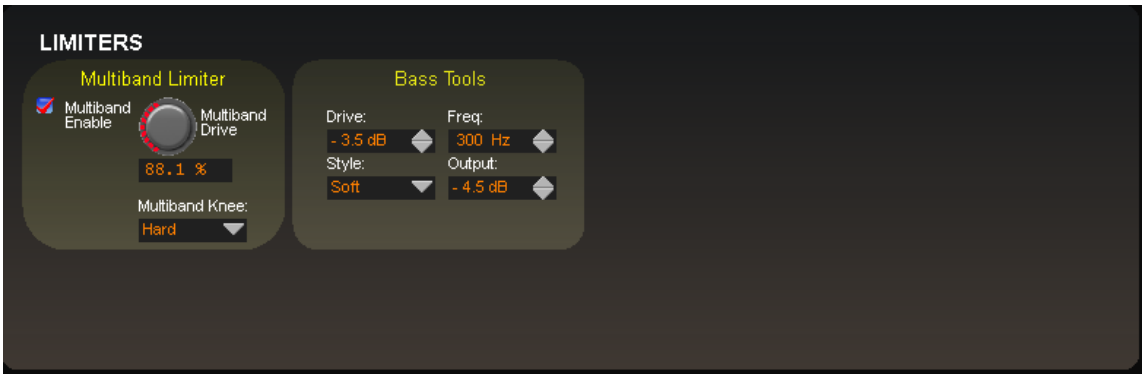
VP-8IP FM-HD Process Mode

The following discussion is unique to the VP-8IP Audio Processor when it is operating in its *FM-HD* mode.

There are two menu screens that are pertinent to the *FM-HD* mode, used for adjusting how the processor's final limiting scheme sounds and interfaces with external transmission equipment following the VP-8IP. These are the *Limiters* and *FM-HD* screens, the latter being visible and selectable via the *FM-HD* tab when VP-8IP is in its *FM-HD* mode.

Audio processing within the four band AGC takes place in the Sum and Difference domain which allows tasteful management of the FM-HD stereo image without creating codec-teasing artifacts.

LIMITERS MENU - FM-HD MODE



Limiters Screen in FM-HD Mode

Multiband Limiter

The VP-8IP contains our exclusive, fully adjustable "swish-free" eight band FM-HD final limiter algorithm. This limiter is especially effective at controlling peak energy without generating unnatural swishing and "phasey" artifacts that outdated and simplistic multiband algorithms, even with fewer bands, generate.

There are no caveats for using the eight band limiter for FM-HD, though it is probably inappropriate to drive the limiters continuously beyond 3-6dB of indicated gain reduction. We recommend operating the limiters in a manner that allows them to do three things: gently increase RMS energy (loudness), shape the HF spectrum to make it more suitable for the FM-HD medium, and control longer duration transients that the preceding four band AGC/compressor intentionally misses in order to keep the processor sounding musical.

Therefore for best musicality we recommend that the limiters not be operated with extremely short attack and release times. The reasoning behind this is twofold:

1. Extremely short attack times, while affording virtually perfect peak control, remove so much transient energy as to make the sound too homogenized and lifeless.
2. Even though our limiters have delayed release built, in extremely short release times may increase certain coding artifacts. This is because the codec will see the

resulting intermodulation distortion as sidebands around a real signal frequency and, not knowing that it is not audio, will code it along with the audio and reveal it as an unmasked artifact.

Please open several of our Factory FM-HD presets and observe the settings in the eight band limiter section to see what we mean.

The eight band limiter's operating parameters are fully adjustable in order to accommodate a wide variety of program material and competitive needs. It also contains the embedded Vorsis Bass Tools which is tightly integrated within the limiter structure to intelligently control bass program energy.

Multiband Enable

This check box determines whether or not any processing is applied by the eight band multiband limiter. When the eight band section is not enabled, audio peak control is handled solely by the look ahead limiter.

Multiband Drive

This control determines the amount of input signal applied to the eight band limiter and, therefore, how hard it is being driven. Lower *Multiband Drive* settings create a more gentle sound while still controlling peak energy, while higher *Multiband Drive* settings make the sound more aggressive by increasing short term audio density. Very high drive settings resulting in more than 6-8dB of continuous gain reduction are typically unnecessary.

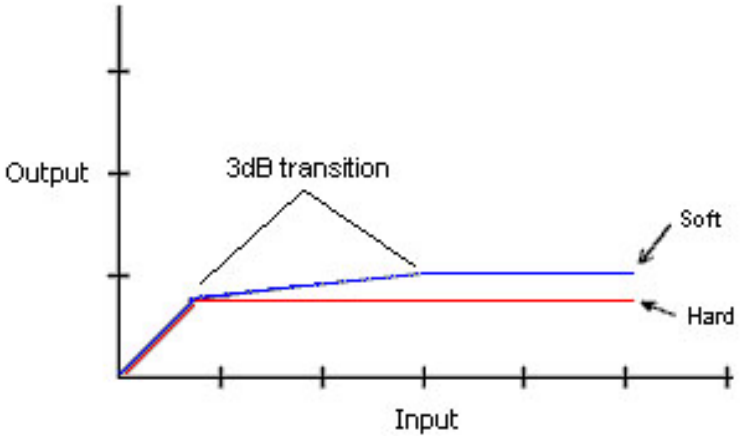
The absolute *Multiband Drive* control setting is dependent upon the setting of the *Makeup* control in the four band AGC's control screen. For this reason, the control is not labeled in dB Drive, but instead is labeled in percent (%). With typical settings of the four band *Makeup* control (+24dB to +30dB) the correct eight band limiter *Multiband Drive* control setting will usually be between 85% and 95%. The full adjustment range is 0% to 100%.

Higher *Multiband Drive* settings cause deeper gain reduction in the eight band limiter (as well as higher levels of limiting in the final look ahead limiter if the *Lim Drive* is not first reduced before the *Multiband Drive* is advanced). Therefore, *Multiband Drive* and *Lim Drive* are the pair of controls most useful for trading off "loudness," "density," and "distortion."

Multiband Knee

Each of the eight bands' threshold, attack, and release are adjustable, as is the overall transfer function (knee hardness). The knee shape determines how the limiters behave near the threshold of limiting. The graphic below illustrates the difference between the two transfer functions.

In the *Hard* mode, limiting action does not occur until the threshold within a particular band is precisely reached. The sound of this setting is more open and free because no limiting activity occurs *until* the threshold is achieved.



Comparison of Hard and Soft Limiter Modes

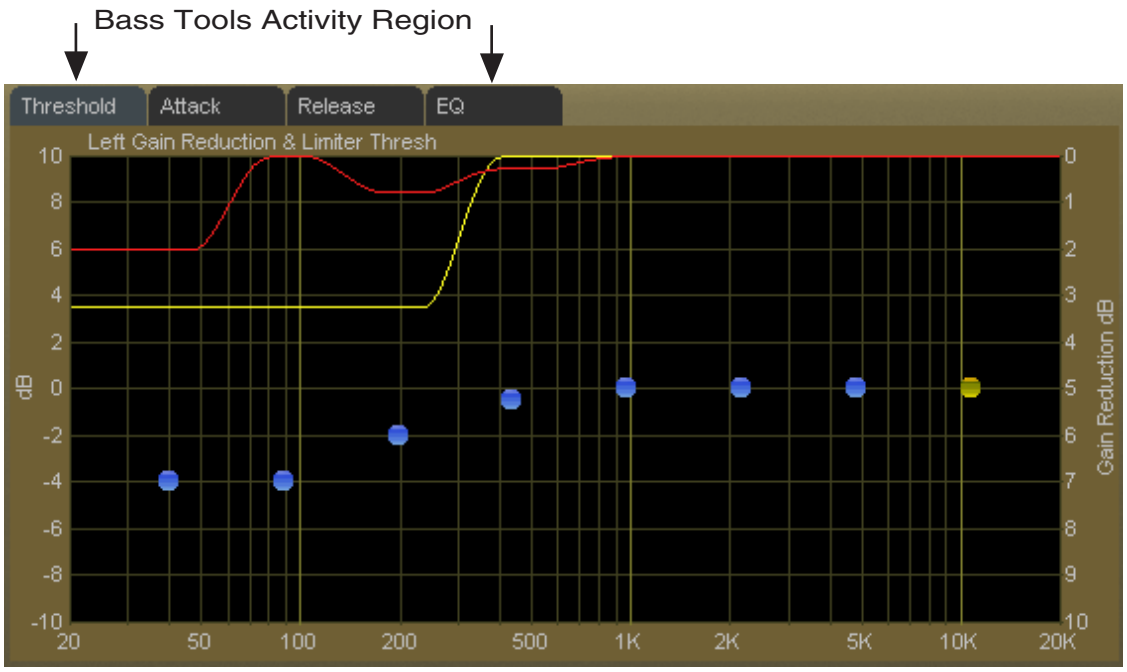
In the *Soft* mode, limiting starts to occur at a level that is 3dB below the actual limiter threshold setting and the transfer function at this point is relatively soft. That is, the audio waveform is not abruptly limited, but instead is allowed to gradually increase over an additional 3dB window until the ultimate limit threshold is reached, whereupon no further amplitude increase is allowed.

Because of this behavior, the sound of the *Soft* function is more obvious than the *Hard* setting. Until the ultimate limit threshold (the setting of the band threshold) is finally reached it is also less precise as a true limiter.

Bass Tools

Vorsis Bass Tools allows extremely low program frequencies to be processed without causing dynamic gain changes and without generating undesirable intermodulation artifacts in higher frequency signals. Its primary purpose is to allow the thumps of kick drums and other percussion instruments to sound natural in their "percussiveness" without pumping or other undesired artifacts. The algorithm operates in left/right mode treating both channels identically.

Our laboratory research on codec behavior rather surprisingly revealed that it is perfectly acceptable to process very low frequency material using the Bass Tools algorithm provided that the harmonics generated during the process are well controlled spectrally. Since our bass algorithm does not allow significant harmonic levels above the third or fifth (somewhat dependent the Bass Tools *Style* setting) there is no audible downside to using it with coded audio. Furthermore, the ability to utilize the Bass Tools on both the FM and HD sides of the processor allows the sound to be more evenly matched as receiver-based digital to analog crossfades are encountered.



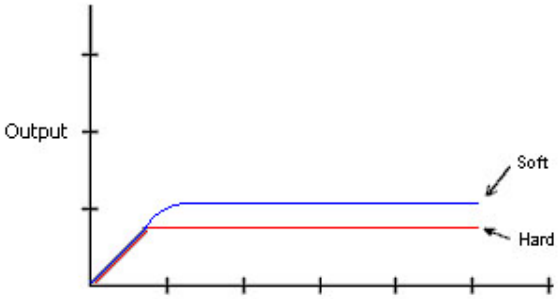
Bass Tools Activity Display - 20Hz up to VBMS Frequency Setting

Drive

This control adjusts the amount of Input Drive to the Bass Tools algorithm. Higher drive settings make it work harder and generate higher levels of harmonics. The *Drive* control adjusts the amount of drive to the Bass Tools and is adjustable over a range of -6dB to +12dB. Adjustment is best done by ear with percussive source material.

Style

Two Bass Tools operating styles, plus Off, are available. The *Hard* setting has a transfer function occupying approximately 1dB of transition πamplitude. The waveform control is fairly abrupt and generates low order harmonics.



Comparison of Bass Tools Operating Modes

The *Soft* setting operates with a transfer function occupying approximately 2dB of transition amplitude. Therefore the control is less sharp than the hard setting, generating more low order harmonics. While the two settings may appear to be almost the same, the 1dB difference in transfer ratio creates a *significant* difference in how the two Bass Tools styles *sound* on actual program material. When Off is selected, the Bass Tools processing is bypassed.

Frequency

The *Freq* control is simple. All audio below the frequency selected is processed by Bass Tools. Smoother bass can be had with settings above 200Hz, which is recommended.

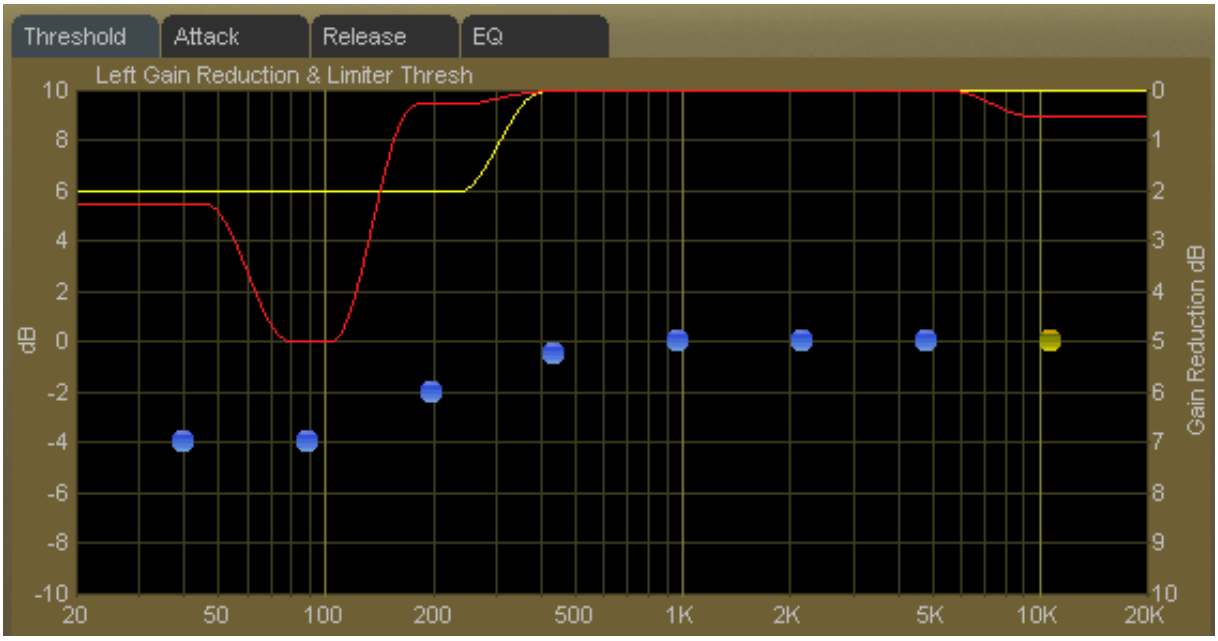
Output

The Bass Tools *Output* control adjusts the output of the Bass Tools algorithm and can be thought of as a "low bass mix" control. Settings between -6dB and -3.0dB add extra bass to the overall mix, while settings lower than -6.0dB reduce it. Both the Bass Tools *Drive* and the *Output* controls are best adjusted by ear using percussive bass material. The *Output* control is adjustable between -12dB and 0dB.

Multiband Limiter Adjustments

Threshold Screen

Clicking on the *Threshold* tab on the graphic opens up the adjustment screen for the limiter thresholds for each of the eight bands.



Multiband Limiter Threshold Screen

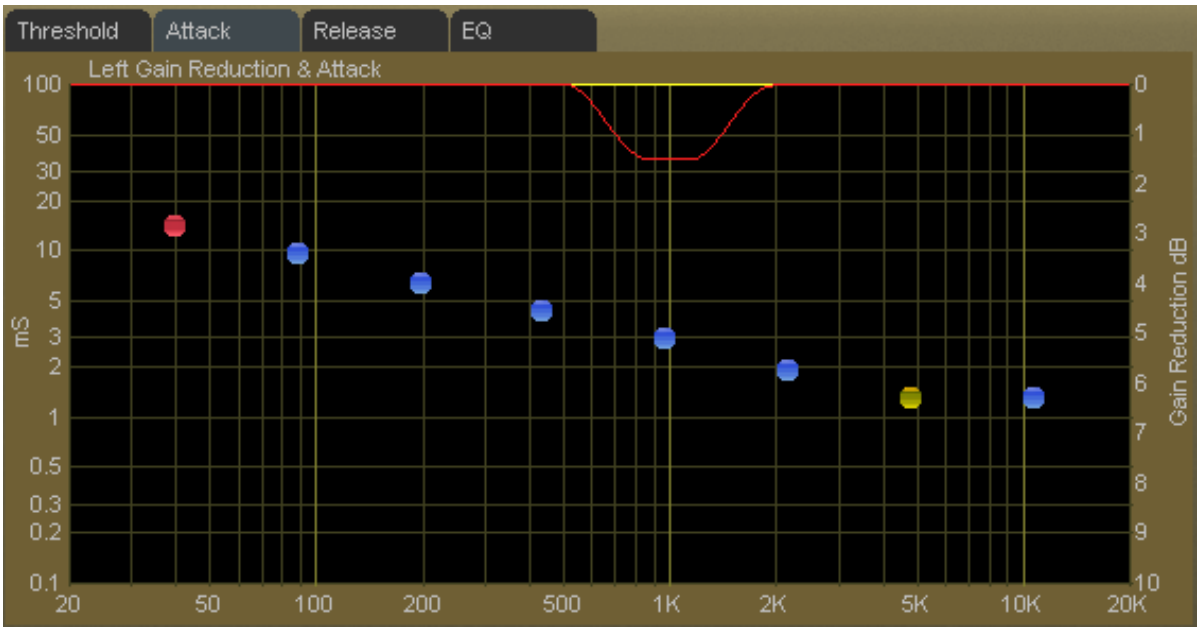
The limiter thresholds are adjustable over a +/-10dB range. This wide range permits frequency conscious limiting if needed.

Each of the limiter thresholds can be adjusted separately or in groups according to the manner described in the earlier *VP-8IP Graphical Interface Operation* section. Lower settings (more negative numbers) reduce the threshold, causing limiting to occur at lower audio levels. Conversely, higher thresholds cause limiting to occur at higher audio levels.

Note: Radically different settings between adjacent bands won't cause strange sounds – there is no swishing in the Vorsis multiband limiter algorithm.

Attack Time Screen

Clicking on the *Attack* tab opens up the attack adjustment screen.



Multiband Limiter Attack Screen

The attack time of a limiter is a measure of how fast it responds to an audio peak when an audio peak exceeds the limiter threshold. Shorter attack times cause the limiter to respond to audio peaks quicker, affording more accurate peak control, but excessively short attack times, while creating "perfect" peak control, can literally suck the life out of musical material.

At the other extreme, excessively long attack times can allow audio peaks to escape the limiters largely unmanaged, leaving them to be controlled solely by the final look ahead output limiter.

The attack times for each band are adjustable from 0.1 milliseconds (100 microseconds) to 100 milliseconds. As explained above, faster attack times exert more control over audio peaks, while slower attack times allow peaks to escape the limiter section with less control.

Attack times are generally adjusted to taste, keeping in mind that the slower the attack time the more work the limiter must do, while conversely the faster the attack time the fewer short term dynamics the program material will have.

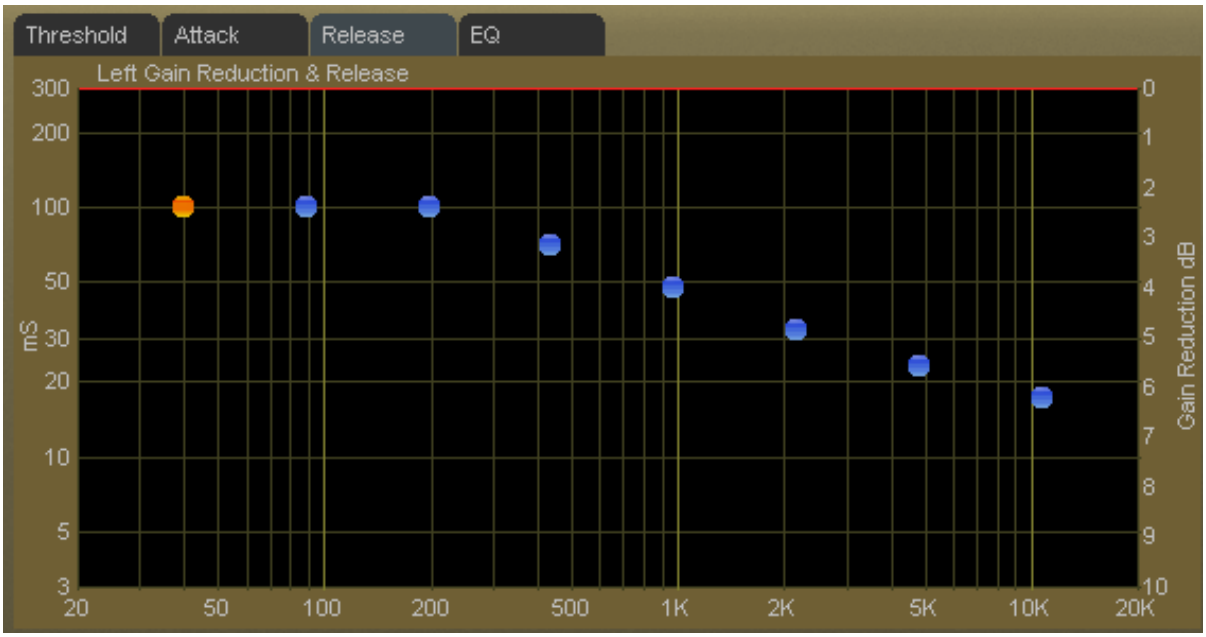
Generally, lower frequencies require slower attack times and higher frequencies require faster attack times for a given amount of control.

The attack times of each band may be individually or collectively modified using the GUI operating instructions provided earlier.

"When it comes to adjusting an audio processor there are really no right or wrong settings - what sounds right and works right in the specific application is right!"

Release Time Screen

Clicking on the *Release* tab opens up the release time adjustment screen.



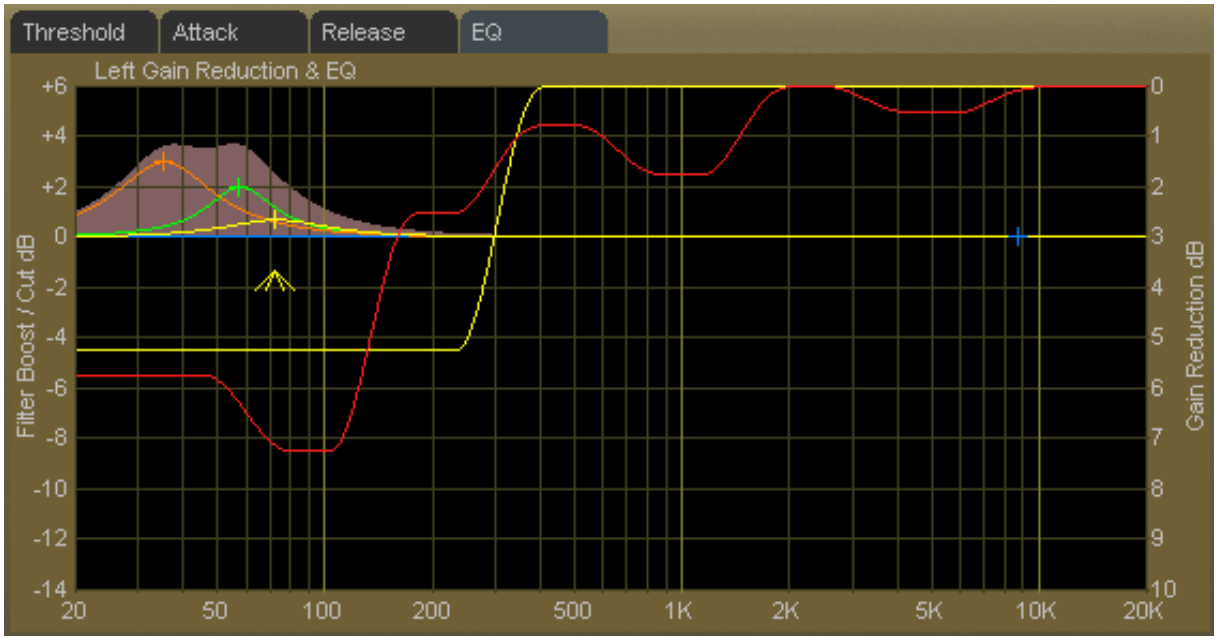
Multiband Limiter Release Time Screen

The release time of a limiter determines how fast it recovers from an audio peak that has caused a momentary decrease in gain. Shorter release times create a more dense sound, and excessively short release times might increase intermodulation distortion (the mixing of low and high frequency energy – or a muddying of the sound). The multiband limiters all have Delayed Release timing built in to minimize unintentional intermodulation distortion.

The release times for each band are adjustable from 300 milliseconds down to 3 milliseconds. Faster release allows a more rapid recovery after a limiting event, while slower release times do the opposite. Release times should be adjusted to taste, keeping in mind that the slower the release time the more dynamic the sound, but the more likely that audible ducking may be heard.

Final Equalization Screen

Clicking on the *EQ* tab opens up the final equalization screen.



Multiband Limiter Equalization (EQ) Screen

Under certain conditions it may be desired to alter the sound exiting the eight band limiter. A ***small*** amount of equalization performed after limiting can open up the sound or provide a *slight* enhancement to its overall feel. Minor equalization tweaks can be accomplished after the limiters without radically disturbing the peak levels emerging from the eight band algorithm.

Only very small amounts of *boost* equalization are recommended once spectral limiting has been accomplished in the eight band section. In fact, the adjustment range has been limited to +3dB/-12dB in order to curtail the temptation to perform aggressive equalization where it is the most inappropriate. Even this range is quite large considering the effect that can be created.

Small adjustments in EQ boost – on the order of 1dB or 2dB at most – are sufficient to alter the overall tonality of the eight band section.

Equalization boosts are always approached with caution because they can add significant energy that must be controlled later by the look ahead limiter (or clippers).

On the other hand, equalization *cuts* can be made as desired because they *remove*, not *add*, energy.

FM-HD MENU

FM-HD Screen

Because the bitrate of the associated FM-HD technology is high enough as to not (yet) require specialized algorithms in order to achieve great on-air sound the *FM-HD* screen does not need the myriad controls that the other processing modes require. Therefore there are only a few controls on the *FM-HD* screen to adjust the characteristics of the output signal. This is because "creative" and final limiting has already been achieved with the controls within the eight band Spectral and Look Ahead Limiters.

FM-HD Look Ahead Limiter

The FM-HD output is accurately peak controlled by a highly oversampled feed forward limiter with a 0.5 millisecond (500 microseconds) look ahead time. The control path (not the audio path) is highly oversampled in order to achieve extremely precise peak control. The primary goal of oversampling in the control sidechain is better peak control through finer definition of the audio waveform at the input to the detector. Oversampling of the audio signal itself is not only unnecessary, but wastes DSP resources. The graphic below illustrates why oversampling is more effective when the control signal is oversampled.

By sampling the audio waveform with finer resolution, peaks that would be missed because they would fall "between the samples" can be reliably detected. Note that if *both* the audio *and* control signals were oversampled, peaks would be missed again because their relative time alignment would be identical to the non-oversampled case!

FM-HD Look Ahead Limiter Controls

The graphic on the right shows the controls available for the look ahead limiter. The following text describes each control according to function.

Look-ahead Limiter Enable

When this check box is checked the look ahead limiter is enabled.

Left/Right Linked

When this box is checked the look ahead limiter's left and right channel *control* signals are common. This prevents unnatural stereo image shifts when one channel needs to reduce the signal in order to control peak excursions.

Lim (Limiter) Drive

This controls the amount of drive to the look ahead limiter. The gain range of this control is +12dB/-18dB. The resolution of this control is 0.01dB, fine enough for the most exacting drive level adjustments.

Attack Time

This control adjusts the attack time of the look ahead limiter. Because the look ahead is 0.5mS, an attack time setting of 0.2mS creates a limiter that attacks 0.3mS before the peak arrives at the limiter's output port. Likewise, a setting of 0.5mS creates a limiter with zero attack time (because the look ahead is 0.5mS). All settings above 0.5mS create a conventional limiter with the control calibration skewed a negative 0.5 millisecond because of the look ahead time. The full range is 0.2mS to 100mS.

Very fast attack times can afford very accurate peak control but at the expense of musicality and program dynamics. Therefore you should always choose the fastest attack time consistent with how you wish the output of the processor to sound, realizing that adjusting the limiter for longer attack times will, as a consequence of those longer attack times, allow some audio peaks to escape uncontrolled. How the attack time is adjusted is always a tradeoff between the accuracy of peak control and the musicality of the processor, with attack time settings above approximately 6 milliseconds being far more musical (in our opinion) than those settings below 6 milliseconds.

Release Time

Controls the primary (fast) release time of the look ahead limiter.

Slower settings (longer release times) cause the peak limiter to recover more slowly from limiting; release times that are set too slow may allow undesirable signal ducking to occur because the audio level remains reduced too long after a peak limiting event.

Conversely, fast release times increase signal density, but release times that are too fast may result in increased intermodulation distortion that can "muddy" the sound. When processing for a perceptual codec we recommend release times no faster than 50 milliseconds in order to minimize the generation (and coding) of close-in difference frequency intermodulation components that can result from very fast release times.

The overall range of the control is from 33 milliseconds to 330 milliseconds with an ideal setting for most applications falling between 70 and 130 milliseconds.

Delayed Release

Controls the secondary (slow) release of the look ahead limiter. The *Delayed Release* control has no effect unless the *Delayed Release* check box is checked (see below).

When Delayed Release is engaged via the *Delayed Release* check box the limiter will release the first 3dB of gain reduction at the main *Release* setting and the remainder of the release at the *Delayed Release* setting. Delayed Release affords a drastic reduction in intermodulation distortion and generally allows deeper limiting depth without incurring the usual intermodulation distortion penalty.

Use of the Delayed Release mode is always recommended when processing ahead of a perceptual codec. As we discussed before, very fast limiter release times can generate close-in intermodulation products around desirable signals and this may degrade the perceived quality of the audio at the output of the codec more than just the action of the codec or limiters themselves would imply.

The *Delayed Release* control offers adjustments between 100 milliseconds and 1.000 seconds. We cannot recommend a *Delayed Release* setting below approximately 160 milliseconds when processing for a perceptual codec. An ideal setting for the majority of applications will fall between settings of 300 and 600 milliseconds.

Phase Reverse

When this check box is checked the electrical phase of the left and right signals are flipped 180 degrees.

Left Right Reverse

When this check box is checked the left and right channels are swapped. If it is noticed that the left and right channels appear to be interchanged at the output of the VP-8IP, please check this control first to ensure that it is not checked!

Analog Output

Controls the output amplitude of the analog output signal appearing on the left and right channel XLR's on the rear panel. This control can be set to OFF or to levels from -79.95dB (or -57.95dBu) to 0.00dB (or +24.0dBu) in 0.05dB steps.

Digital Output

Adjusts the *peak* audio signal level present at the AES/EBU digital output. The resulting output levels are simultaneously indicated in dBFS by the front panel and remote GUI bargraph meters. This *Digital Output* control can be set to OFF or adjusted over the range of minus 35.95dBFS to 0.00dBFS.

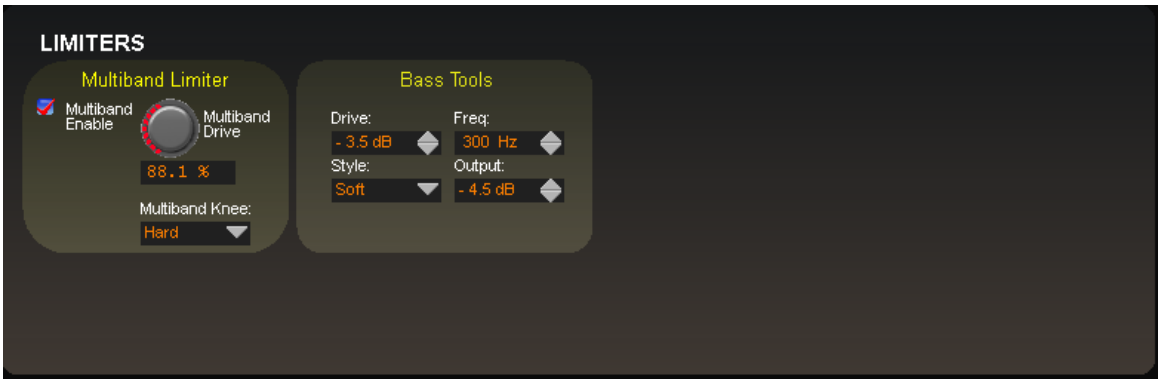
VP-8IP AM-HD Process Mode

The following discussion is unique to the VP-8IP Audio Processor when it is operating in its *AM-HD* mode. This mode is also ideal for processing prior to the Codecs utilized for DAB and DRM, ensuring the very best overall sound at the listener's receiver.

There are two menu screens that are pertinent to the *AM-HD* mode, used for adjusting how the processor's final limiting scheme sounds and how it interfaces with external transmission equipment following the VP-8IP. These are the *Limiters* and *AM-HD* screens, the latter being visible and selectable via the *AM-HD* tab when VP-8IP is in its *AM-HD* mode.

Audio processing within the four band AGC takes place in the Sum and Difference domain which allows tasteful management of the *AM-HD* stereo image without creating codec-teasing artifacts.

LIMITERS MENU - AM-HD MODE



Limiters Screen in AM-HD Mode

Multiband Limiter

The VP-8IP contains our exclusive, fully adjustable eight band AM-HD Spectral Limiter algorithm. This limiter is especially effective at controlling peak energy without generating unnatural swishing and "phasey" artifacts that outdated and simplistic multiband algorithms, even with fewer bands, generate.

There are no caveats for using the eight band limiter for AM-HD, though it is probably inappropriate to drive the limiters continuously beyond 3-6dB of gain reduction. We recommend operating the limiters in a manner that allows them to do three things: gently increase RMS energy (loudness), shape the HF spectrum to make it more suitable for the AM-HD medium, and control longer duration transients that the preceding four band AGC/compressor intentionally misses in order to keep the processor sounding musical.

Therefore for best musicality we recommend that the limiters not be operated with extremely short attack and release times. The reasoning behind this is twofold:

1. Extremely short attack times, while affording virtually perfect peak control, remove so much transient energy as to make the sound too homogenized and lifeless.

2. Extremely short release times, even though our limiters have delayed release built in, may increase certain coding artifacts. This is because the codec will see the resulting intermodulation distortion as sidebands around a real signal frequency and, not knowing any better, will code it along with the audio as an unmasked artifact.

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Multiband Enable

This check box determines whether or not any processing is to be applied by the eight band multiband limiter. When the eight band section is not enabled, audio peak control is handled solely by the look ahead limiter.

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This control determines the amount of input signal applied to the eight band limiter and, therefore, how hard it is being driven. Lower *Multiband Drive* settings create a more gentle sound while still controlling peak energy, while higher *Multiband Drive* settings make the sound more aggressive by increasing short term audio density. Very high drive settings resulting in more than 6-10dB of continuous gain reduction are typically unnecessary.

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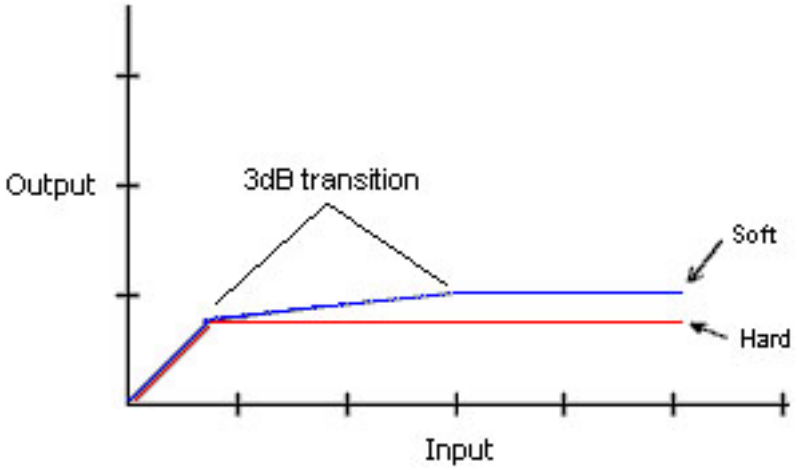

Comparison of Hard and Soft Limiter Modes

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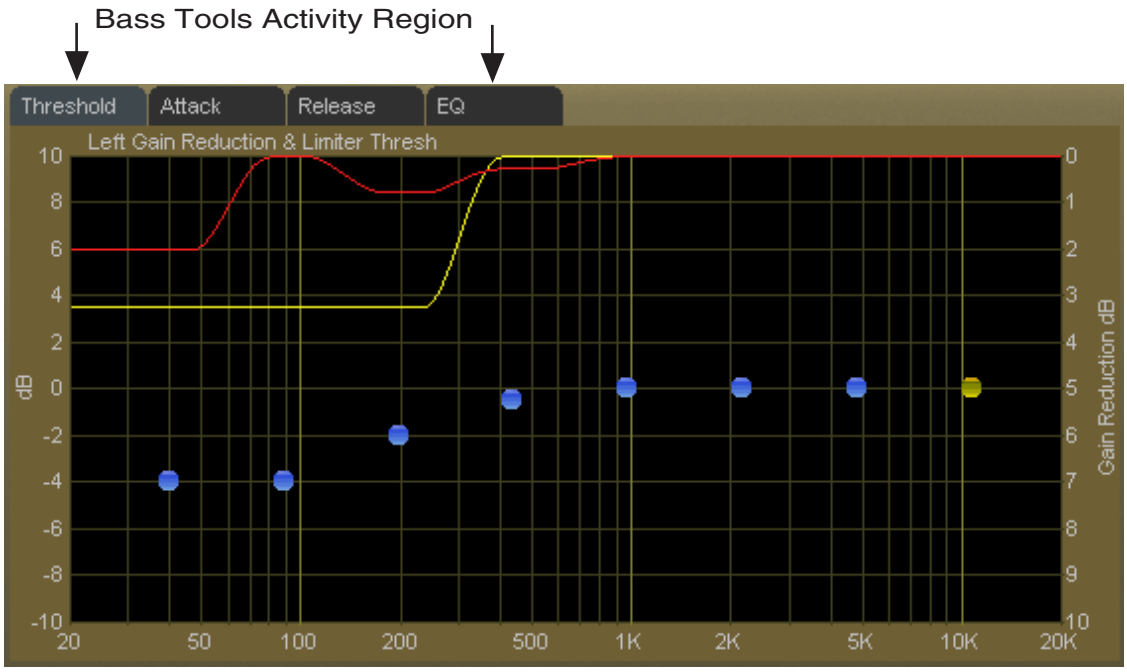

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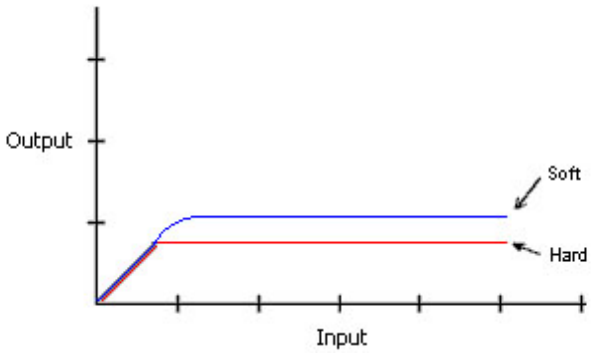

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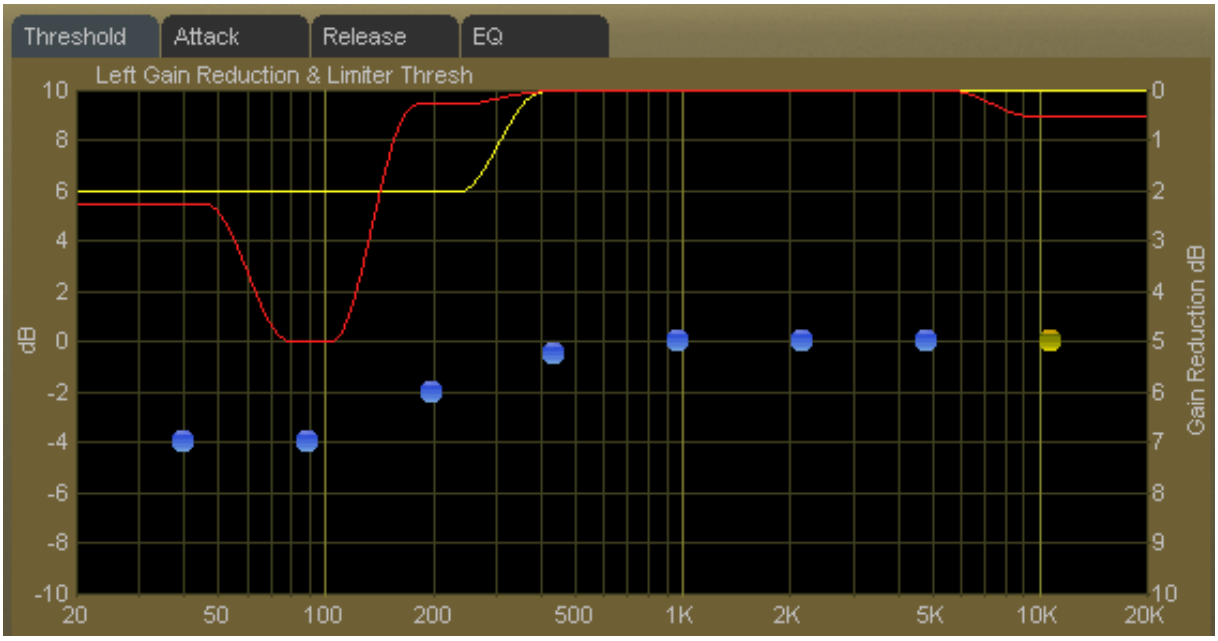

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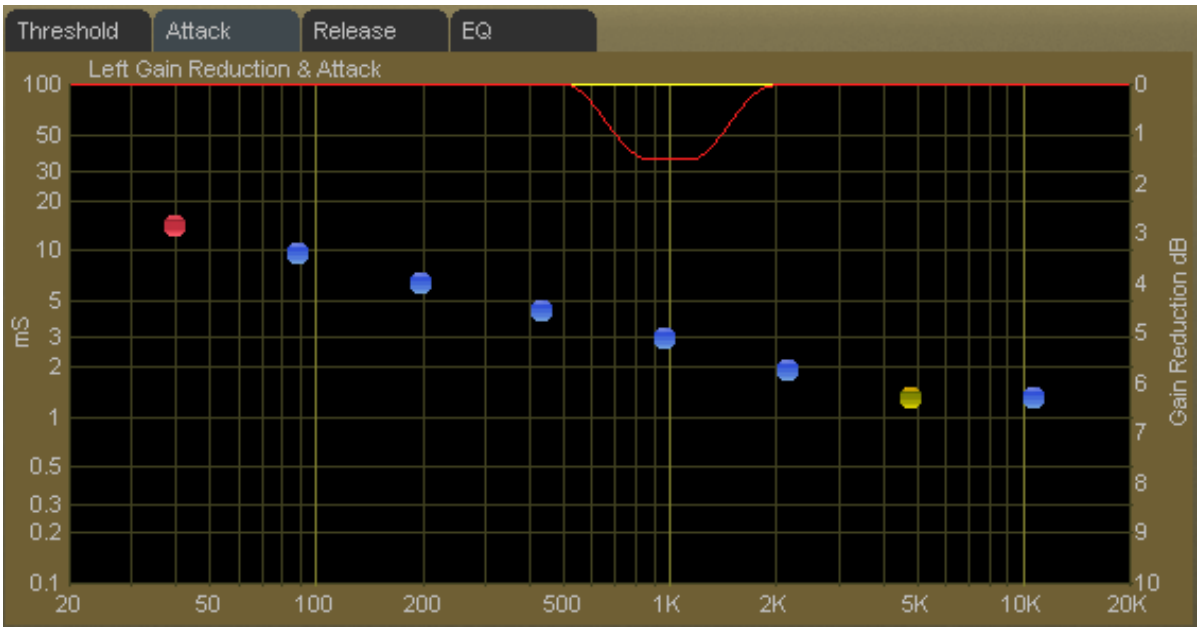
Attack Time Screen

Clicking on the *Attack* tab opens up the attack adjustment screen.



Multiband Limiter Attack Screen

The attack time of a limiter is a measure of how fast it responds to an audio peak when an audio peak exceeds the limiter threshold. Shorter attack times causes the limiter to respond to audio peaks quicker, which affords more accurate peak control, noting that excessively short attack times, while creating "perfect" peak control, can literally suck the life out of musical material.

At the other extreme, excessively long attack times can allow audio peaks to escape the limiters largely unmanaged, leaving them to be controlled solely by the final look ahead output limiter.

The attack times for each band are adjustable from 0.1 milliseconds (100 microseconds) to 100 milliseconds. As explained above, faster attack times exert more control over audio peaks, while slower attack times allow peaks to escape the limiter section with less control.

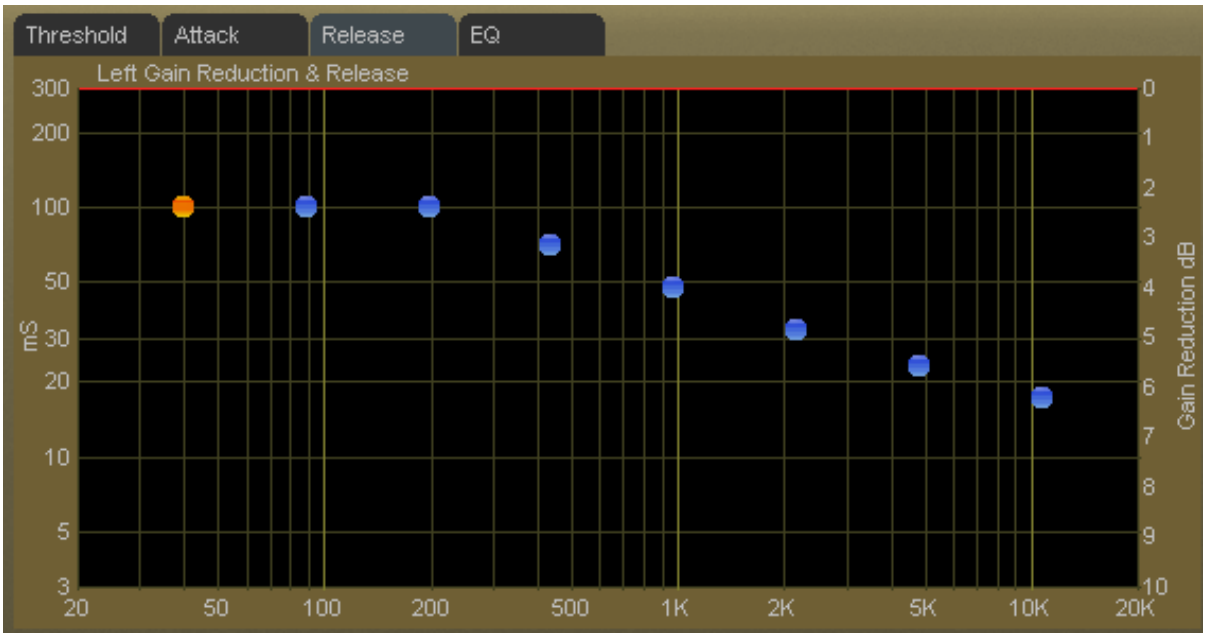
Attack times are generally adjusted to taste, keeping in mind that the slower the attack time the more work the limiter must do, while conversely the faster the attack time the fewer short term dynamics the program material will have.

Generally, lower frequencies require slower attack times and higher frequencies require faster attack times for a given amount of control, though this statement is not, nor should it even be, "written in stone." When it comes to setting up an audio processor there is really no right or wrong setting – what *sounds* right and *works* right in the specific application *is* right!

The attack times of each band may be individually or collectively modified using the GUI operating instructions provided earlier.

Release Time Screen

Clicking on the *Release* tab opens up the release time adjustment screen.



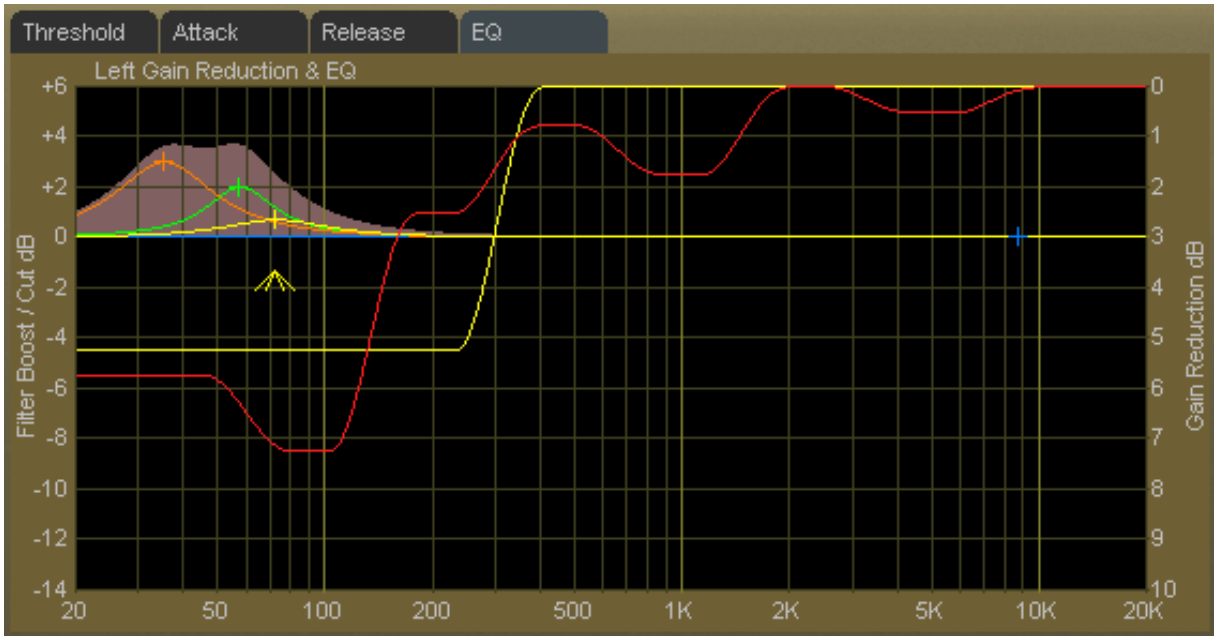
Multiband Limiter Release Time Screen

The release time of a limiter determines how fast it recovers from an audio peak that has caused a momentary decrease in gain. Shorter release times create a more dense sound, and excessively short release times might increase intermodulation distortion (the mixing of low and high frequency energy – or a muddying of the sound). The multiband limiters all have Delayed Release timing built in to minimize unintentional intermodulation distortion.

The release times for each band are adjustable from 300 milliseconds down to 3 milliseconds. Faster release allows a more rapid recovery after a limiting event, while slower release times do the opposite. Release times should be adjusted to taste, keeping in mind that the slower the release time the more dynamic the sound, but the more likely that audible ducking may be heard.

Final Equalization Screen

Clicking on the *EQ* tab opens up the final equalization screen.



Multiband Limiter Equalization (EQ) Screen

Under certain conditions it may be desired to alter the sound exiting the eight band limiter. A ***small*** amount of equalization performed after limiting can open up the sound or provide a *slight* enhancement to its overall feel, and minor equalization tweaks can be accomplished after the limiters without radically disturbing the peak levels emerging from the eight band algorithm.

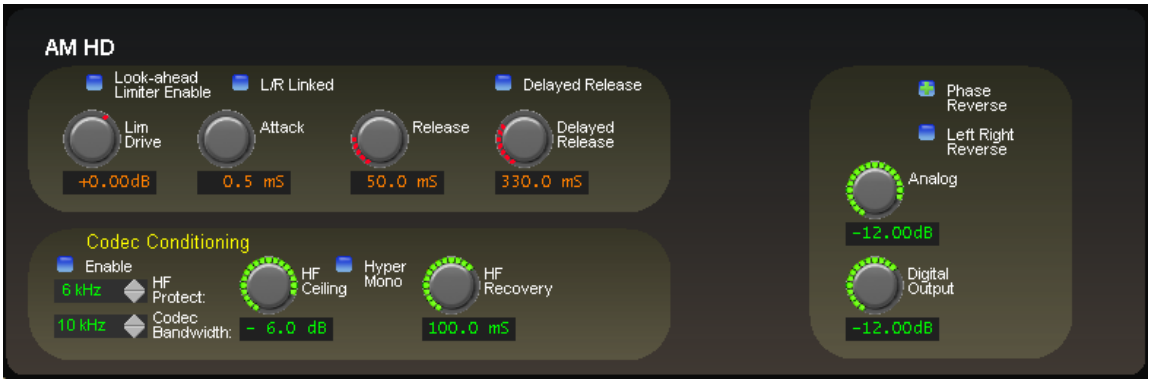
Only very small amounts of *boost* equalization are recommended once final limiting has been accomplished in the eight band section. In fact, the adjustment range has been limited to +3dB/-12dB in order to curtail the temptation to perform aggressive equalization where it is the most inappropriate, and even this range is quite large considering the effect that can be created.

Small adjustments in EQ boost – on the order of 1dB or 2dB at most – are sufficient to alter the overall tonality of the eight band section.

Equalization boosts are approached with caution because they can add significant energy that must be controlled later by the look ahead limiter (or clippers).

On the other hand, equalization *cuts* can be made as desired because they *remove*, not *add*, energy.

AM-HD MENU



AM-HD Screen

Codec Bandwidth

This control adjusts the bandwidth of a high order, linear phase FIR low pass filter that removes all high frequency energy above its setting. This control permits the output audio bandwidth to be restricted if required, and settings between 4kHz and 16kHz are available in 1kHz steps. When the control is set to its OFF position the audio output bandwidth is 24kHz, the Nyquist frequency of the output section's 48kHz sampling rate.

Enable

When *Enable* is selected the VP-8IP enables a specialized high frequency management algorithm that artistically minimizes "splashiness" in the high frequency spectrum. Such "splashiness" is known to exacerbate coding artifacts, especially when the Codec is operating at very low bitrates.

HF Protect Freq

When the HF Protect algorithm is active (see above) it will work to minimize coding artifacts caused by certain kinds of high frequency program energy. The *HF Protect* control adjusts the frequency where control action begins. This control works in concert with the *Ceiling* control (below).

HF Ceiling

The *Ceiling* control adjusts the audio threshold where the HF Protect algorithm will begin to operate. The more negative the control's setting in dB, the sooner (amplitude domain-wise) the high frequency control will take place.

A Note About Attack Time

There is no user-adjustable attack time for the HF Protect algorithm. This is because its *reaction time* is determined automatically by evaluating certain characteristics of the input program energy using FFT (Fast Fourier Transform) analysis before the energy actually reaches the algorithm. Because of the complexity and critical nature of this measurement, and its interaction with other aspects of the HF Protect algorithm, the reaction time is a complex combination of factors that would be extremely difficult to control, let alone optimize, with knobs, and therefore it has not been made field adjustable.

HF Recovery

The primary recovery rate of the HF Protect algorithm is program-adaptive and non-adjustable but the secondary recovery was made adjustable in order to allow a user to optimize it for his particular application. The fastest *HF Recovery* setting is 33mS while the slowest is 330mS. We recommend a starting setting of approximately 80mS for most music-based program formats. Voice only programming may use a faster recovery setting.

A Recovery rate calculator operates in the background to automatically determine the maximum recovery speed consistent with low distortion to ensure that gain control-induced Codec intermodulation distortion can never be generated regardless of control settings by the user and the input program material.

The above means that even if a user has set the *Recovery* control to its fastest possible setting, the Recovery algorithm may (and will!) automatically set it slower under certain program circumstances in order to prevent codec-induced close-in intermodulation products.

Hyper Mono

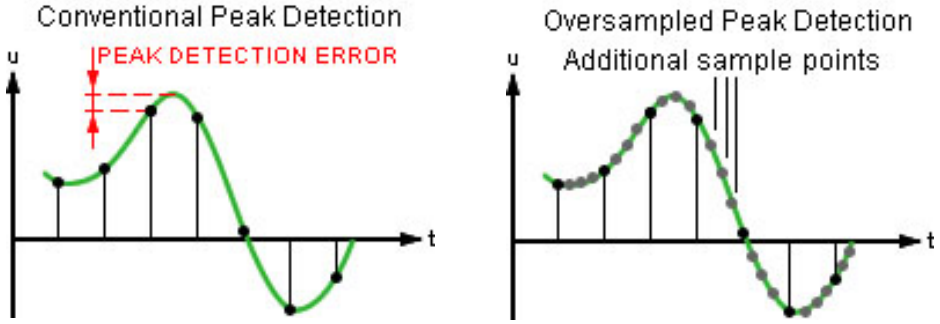
When this box is checked, weak or incorrect L-R information will be removed from the input audio, leaving a true monophonic signal where the left and right channels are absolutely identical.

Intuitively, such an algorithm may seem to be contrary to good sound; however, this is an acceptable tradeoff because the sound of most low bit rate codecs attempting to code unnecessary or inappropriate L-R material is anything but "good."

Hyper Mono will improve the overall sound quality by removing information that the codec should not be utilizing its fixed bit pool to encode.

AM-HD Look Ahead Limiter

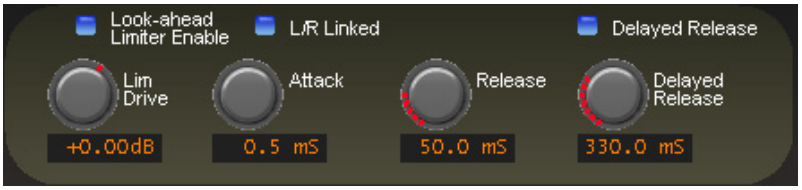
The AM-HD output is accurately peak controlled by a highly oversampled feed forward limiter with a 0.5 millisecond (500 microseconds) look ahead time. The control path (not the audio path) is highly oversampled in order to achieve extremely precise peak control. The primary goal of oversampling in the control sidechain is better peak control through finer definition of the audio waveform at the input to the detector. Oversampling of the audio signal itself is not only unnecessary, but wastes DSP resources. The graphic below illustrates why oversampling is more effective when the control signal is oversampled.



By sampling the audio waveform with finer resolution, peak excursions that would be missed otherwise because they would fall "between the samples" can be reliably detected. Note that if the audio *and* control signals were both oversampled, peaks would be missed again because their relative time alignment would be identical!

AM-HD Look Ahead Limiter Controls

The graphic on the right shows the controls available for the look ahead limiter. The following text describes each control according to function.



Look-ahead Limiter Enable

When this check box is checked the look head limiter is enabled.

L/R Linked

When this box is checked the look ahead limiter's left and right channel *control* signals are common. This prevents unnatural stereo image shifts when one channel needs to reduce the signal in order to control peak excursions.

Lim (Limiter) Drive

This controls the amount of drive to the look ahead limiter. The gain range of this control is +12dB/-18dB. The resolution of this control is 0.01dB, fine enough for the most exacting output level adjustments.

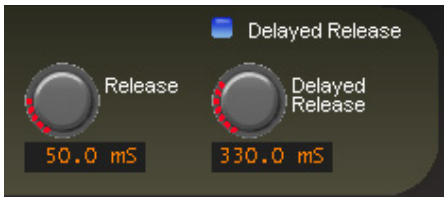
Attack

This control adjusts the attack time of the look ahead limiter. Because the look ahead is 0.5mS, an attack time setting of 0.2mS creates a limiter that attacks 0.3mS before the peak arrives at the limiter's output port. Likewise, a setting of 0.5mS creates a limiter with zero attack time (because the look ahead is 0.5mS). All settings above 0.5mS create

a conventional limiter with the control calibration skewed a negative 0.5 millisecond because of the look ahead time. The full range is 0.2mS to 100mS.

Very fast attack times can afford very accurate peak control but at the expense of musicality and dynamics. You should always choose the fastest attack time consistent with how you wish the output of the processor to sound, realizing that adjusting the limiter for longer attack times will, as a consequence of those longer attack times, allow peaks to escape uncontrolled. How the attack time is adjusted is always a tradeoff between the accuracy of peak control and the musicality of the processor, with attack time settings below approximately 6 milliseconds sounding far less musical (to our ears) than settings above 6 milliseconds.

Release



Controls the primary (fast) release time of the look ahead limiter.

Slower settings (longer release times) cause the peak limiter to recover more slowly from limiting; release times that are set too slow may allow undesirable signal ducking to occur because the audio level remains low for too long after a peak limiting event.

Conversely, fast release times increase signal density, but release times that are too fast result in increased intermodulation distortion which can "muddy" the sound. When processing for a perceptual codec we recommend release times no faster than 50 milliseconds in order to minimize the generation (and coding) of close-in difference frequency intermodulation components that can result from very fast release times.

The overall range of the control is from 33 milliseconds to 330 milliseconds with an ideal setting for most applications falling between 70 and 130 milliseconds.

Delayed Release

Controls the secondary (slow) release of the look ahead limiter. The *Delayed Release* control has no effect unless the *Delayed Release* check box is checked (see below).

When Delayed Release is engaged via the *Delayed Release* check box the limiter will release the first 3dB of gain reduction at the main Release setting, with the remainder of the release occurring at the *Delayed Release* setting. Delayed Release affords a drastic reduction in intermodulation distortion and generally allows deeper limiting depth without incurring the usual intermodulation distortion penalty.

Use of the Delayed Release mode is highly recommended when processing ahead of a perceptual codec. As we discussed above, very fast limiter release times can generate close-in intermodulation products around desirable signals and this may degrade the perceived quality of the audio at the output of the codec more than just the action of the codec or limiters themselves would imply.

The *Delayed Release* control offers adjustments between 100 milliseconds and 1.000 seconds. We cannot recommend a *Delayed Release* setting below approximately 160 milliseconds when processing for a perceptual codec. An ideal setting for the majority of applications will fall between settings of 300 and 600 milliseconds.

Phase Reverse

When this check box is checked the electrical phase of the left and right signals are flipped 180 degrees. This control is useful for matching up the audio phase of the signal passing through VP-8IP to that of a parallel, external signal path.

Left Right Reverse

When this check box is checked the left and right channels are swapped. If it is noticed that the left and right channels appear to be interchanged at the output of the VP-8IP, please check this control first to ensure that it is not checked!

Analog

Controls the output amplitude of the analog output signal appearing on the left and right channel XLR connectors on the rear panel. This output control can be set to OFF or to levels from -79.95dB (or -57.95dBu) to 0.00dB (or +24.0dBu) in 0.05dB steps.

Digital Output

Adjusts the *peak* audio signal level present at the VP-8IP's AES/EBU digital output. The resulting output levels are simultaneously indicated in dBFS by the front panel and remote GUI bargraph meters. This control can be set to OFF or adjusted over the range of minus 35.95dBFS to 0.00dBFS.

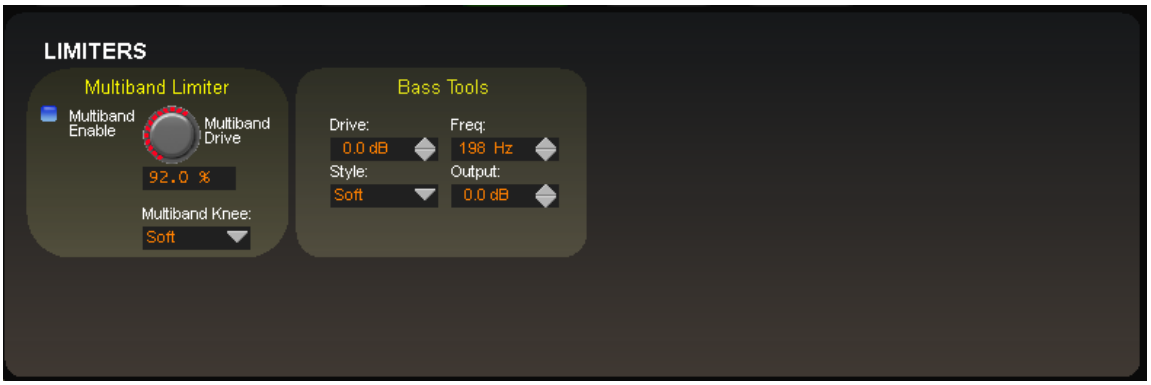
VP-8IP MP3/AAC>48K Process Mode

The following discussion is unique to the VP-8IP Audio Processor when it is operating in its *MP3/AAC>48K* mode.

There are two menu screens that are unique to the *MP3/AAC>48K* mode and used for adjusting how the processor's final limiting scheme sounds and interfaces with external transmission equipment. These are the *Limiters* and *MP3/AAC>48K* screens, the latter being visible and selectable via the *MP3/AAC>48K* tab only when the VP-8IP is in that mode.

Audio processing within the four band AGC takes place in the Sum and Difference domain, which allows tasteful management of the stereo image without creating codec-teasing artifacts.

LIMITERS MENU - MP3/AAC>48K MODE



Limiters Screen in AAC/MP3>48K Mode

Multiband Limiter

The VP-8IP contains our exclusive, fully adjustable "swish-free" eight band final limiter algorithm. This limiter is especially effective at controlling program energy without generating unnatural swishing and "phasey" artifacts that outdated and simplistic multiband algorithms, even with fewer bands, often generate.

There are no caveats for using the eight band limiter, though it is probably unnecessary to drive the limiters continuously beyond 3dB of indicated gain reduction. We generally recommend operating the limiters in a manner that allows them to do three things: gently increase RMS energy (loudness), shape the HF spectrum to make it more suitable for the codec medium, and control longer duration transients that the preceding four band AGC/compressor intentionally misses in order to keep the processor sounding musical.

For best musicality we recommend that the multiband limiters not be operated with extremely short attack and release times. The reasoning behind this is twofold:

1. Extremely short attack times, while affording virtually perfect energy control, remove so much transient energy as to make the sound too homogenized and lifeless (in our opinion).
2. Even though our limiters have delayed release built-in extremely short release times may increase certain coding artifacts. This is because the codec will see the resulting intermodulation distortion as sidebands around a real signal frequency and, not knowing that it is not audio, code it along with the audio and reveal it as an unmasked artifact.

Please open several of our Factory MP3/AAC>48K presets and observe the settings in the eight band limiter section to see what we mean.

The eight band limiter's operating parameters are fully adjustable in order to accommodate a wide variety of program material and competitive needs. It also contains the embedded Vorsis Bass Tools which is tightly integrated within the limiter structure to intelligently control bass program energy.

Multiband Enable

This check box determines whether or not any processing is applied by the eight band multiband limiter. When the eight band section is not enabled, audio peak control is accomplished solely by the look ahead limiter.

Multiband Drive

This control determines the amount of input signal applied to the eight band limiter and, therefore, how hard it is being driven. Lower *Multiband Drive* settings create a more gentle sound while still controlling program energy, while higher *Multiband Drive* settings make the sound more aggressive by increasing short term audio density. Very high *Multiband Drive* settings resulting in more than 6-8dB of continuous gain reduction in any band is unnecessary.

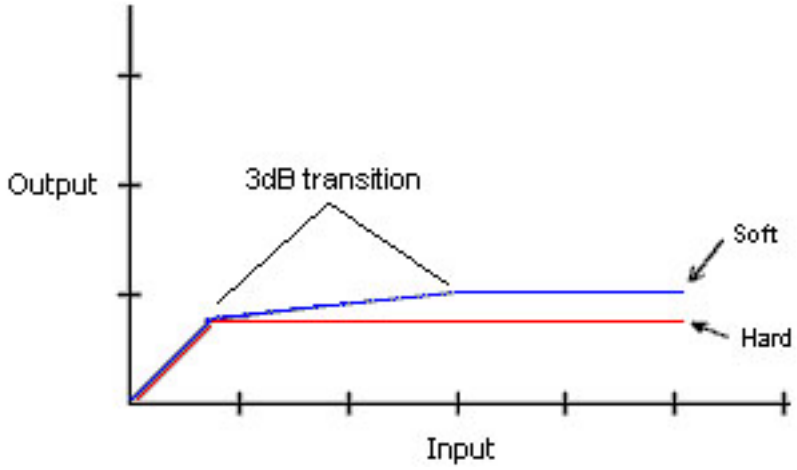
The absolute *Multiband Drive* control setting is also somewhat dependent upon the setting of the *Makeup* control in the four band AGC's control screen. For this reason, the control is not labeled in dB Drive, but instead is labeled in percent (%). With typical settings of the four band AGC's *Makeup* control (+24dB to +30dB) the correct eight band limiter *Multiband Drive* control setting will usually be between 85% and 95%. The full adjustment range is 0% to 100%.

Higher *Multiband Drive* settings cause deeper gain reduction in the eight band limiter (as well as higher levels of limiting in the final look ahead limiter if the *Lim Drive*, located in the *MP3/AAC>48K* screen, is not first reduced before the *Multiband Drive* is advanced). Therefore, *Multiband Drive* and *Lim Drive* are the pair of controls most useful for trading off "loudness," "density," and "distortion." To allow easy adjustment we've placed the *Multiband Drive* control in *both* the *Limiters* and *MP3/AAC>48K* screens to make toggling back and forth between screens unnecessary during adjustment.

Multiband Knee

Each of the eight bands' threshold, attack, and release are adjustable, as is the overall transfer function (knee hardness). The knee shape determines how the limiters behave at or near their thresholds of limiting. The graphic below illustrates the difference between the two transfer functions.

In the *Hard* mode, limiting action does not occur until the threshold within a particular band is precisely reached. The sound of this setting is more open and "free" because no limiting activity occurs *until* the threshold is achieved.



Comparison of Hard and Soft Limiter Modes

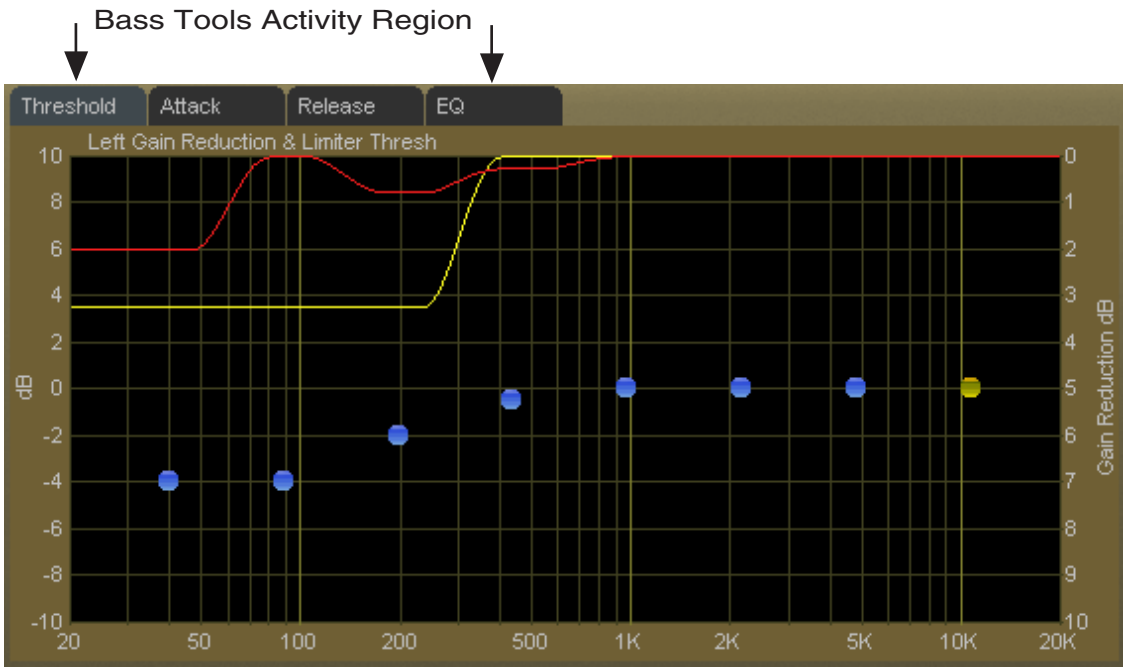
In the *Soft* mode, limiting starts to occur at a level that is 3dB below the actual limiter threshold setting and the transfer function at this point is relatively soft. That is, the audio waveform is not abruptly limited, but instead is allowed to gradually increase over a 3dB window until the ultimate limit threshold is reached, whereupon no further amplitude increase is allowed.

Because of this behavior, the sound of the *Soft* function is more obvious than the *Hard* setting. Until the ultimate limit threshold (the setting of the band threshold) is finally reached it is also less precise as a true limiter.

Bass Tools

Vorsis Bass Tools allows extremely low program frequencies to be processed without causing dynamic gain changes and without generating undesirable intermodulation artifacts in higher frequency signals. Its primary purpose is to allow the thumps of kick drums and other percussion instruments to sound natural in their "percussiveness" without pumping or other undesired artifacts. The algorithm operates in left/right mode treating both channels identically.

Our laboratory research on codec behavior rather surprisingly revealed that it is perfectly acceptable to process very low frequency material using the Bass Tools algorithm provided that the harmonics generated during the process are well controlled spectrally. Since our bass algorithm does not allow significant harmonic levels above the third or fifth (somewhat dependent the Bass Tools *Style* setting) there is no audible downside to using it with coded audio. Furthermore, the ability to utilize the Bass Tools on both the FM and HD sides of the processor allows the sound to be more evenly matched as receiver-based digital to analog crossfades are encountered.



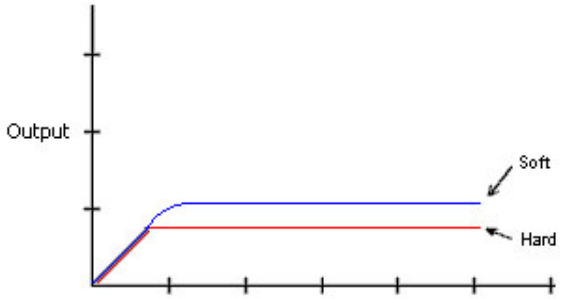
Bass Tools Activity Display - 20Hz up to VBMS Frequency Setting

Drive

This control adjusts the amount of Input Drive to the Bass Tools algorithm. Higher drive settings make it work harder and generate higher levels of harmonics. The *Drive* control adjusts the amount of drive to the Bass Tools and is adjustable over a range of -6dB to +12dB. Adjustment is best done by ear with percussive source material.

Style

Two Bass Tools operating styles, plus Off, are available. The *Hard* setting has a transfer function occupying approximately 1dB of transition amplitude. The waveform control is fairly abrupt and generates low order harmonics.



Comparison of Bass Tools Operating Modes

The *Soft* setting operates with a transfer function occupying approximately 2dB of transition amplitude. Therefore the control is less sharp than the hard setting, generating more low order harmonics. While the two settings may appear to be almost the same, the 1dB difference in transfer ratio creates a *significant* difference in how the two Bass Tools styles *sound* on actual program material. When Off is selected, the Bass Tools processing is bypassed.

Frequency

The *Freq* control is simple. All audio below the frequency selected is processed by Bass Tools. Smoother bass can be had with settings above 200Hz, which is recommended.

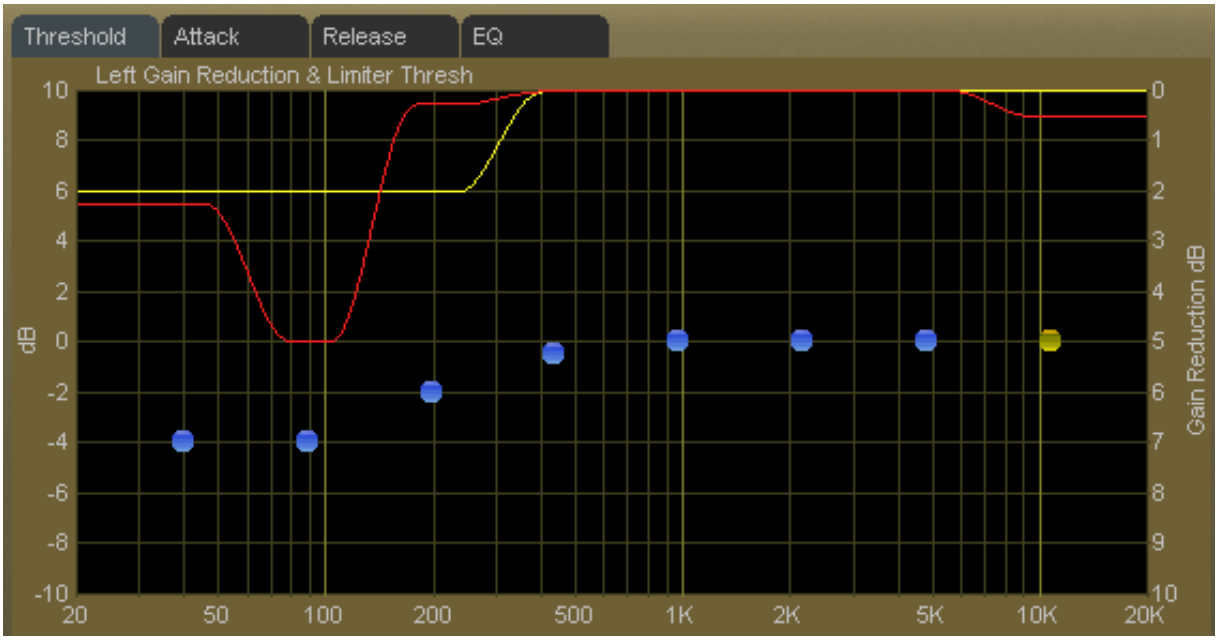
Output

The Bass Tools *Output* control adjusts the output of the Bass Tools algorithm and can be thought of as a "low bass mix" control. Settings between -6dB and -3.0dB add extra bass to the overall mix, while settings lower than -6.0dB reduce it. Both the Bass Tools *Drive* and the *Output* controls are best adjusted by ear using percussive bass material. The *Output* control is adjustable between -12dB and 0dB.

Multiband Limiter Adjustments

Threshold Screen

Clicking on the *Threshold* tab on the graphic opens up the adjustment screen for the limiter thresholds for each of the eight bands.



Multiband Limiter Threshold Screen

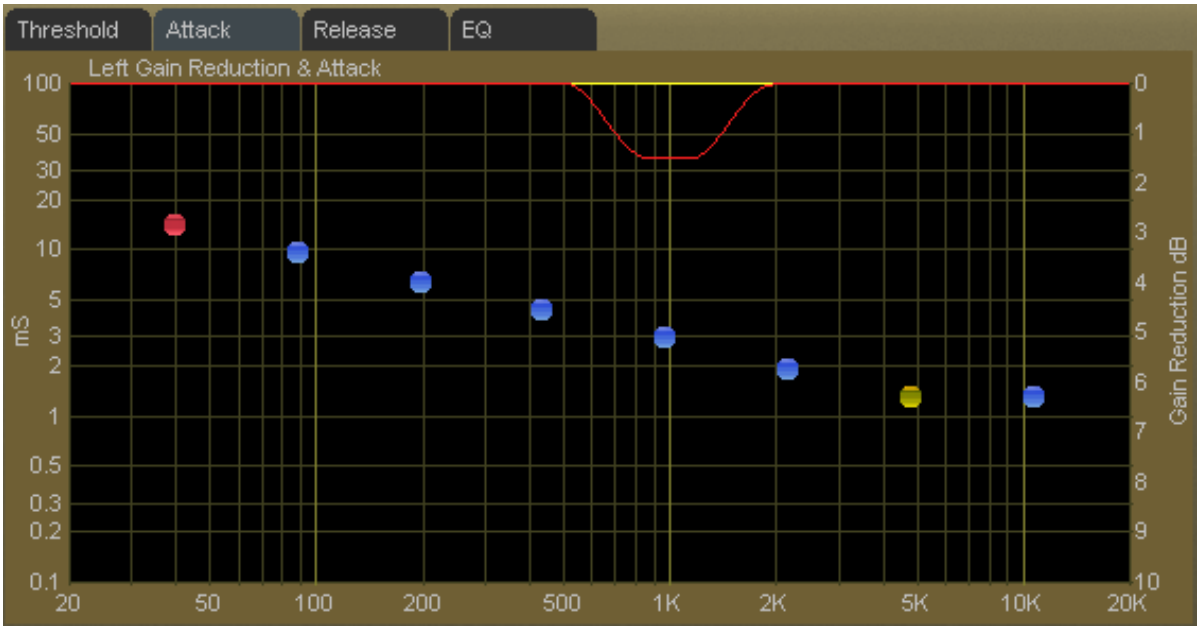
The limiter thresholds are adjustable over a +/-10dB range. The thresholds are adjustable over a wide range to permit frequency conscious limiting to be created if needed.

Each of the limiter thresholds can be adjusted separately or in groups according to the manner described in the earlier *VP-8IP Graphical Interface Operation* section. Lower settings (more negative numbers) reduce the threshold, causing limiting to occur at lower audio levels. Conversely, higher thresholds cause limiting to occur at higher audio levels.

Note: Radically different settings between adjacent bands won't cause strange sounds – there is no swishing in the Vorsis multiband limiter algorithm.

Attack Time Screen

Clicking on the *Attack* tab opens up the attack adjustment screen where blue dots (yellow/orange when selected) set the attack time of each of the eight limiter bands.



Multiband Limiter Attack Screen

The attack time of a limiter is a measure of how fast it responds to an audio peak when it exceeds the limiter threshold. Shorter attack times cause the limiter to respond to audio peaks quicker, affording more accurate peak control, but excessively short attack times, while creating "perfect" peak control, can literally suck the life out of musical material.

At the other extreme, excessively long attack times can allow audio peaks to escape the limiters largely unmanaged, leaving them to be controlled solely by the final look ahead output limiter.

The attack times for each band are adjustable from 0.1 milliseconds (100 microseconds) to 100 milliseconds. As explained above, faster attack times exert more control over audio peaks, while slower attack times allow peaks to escape the limiter section with less control.

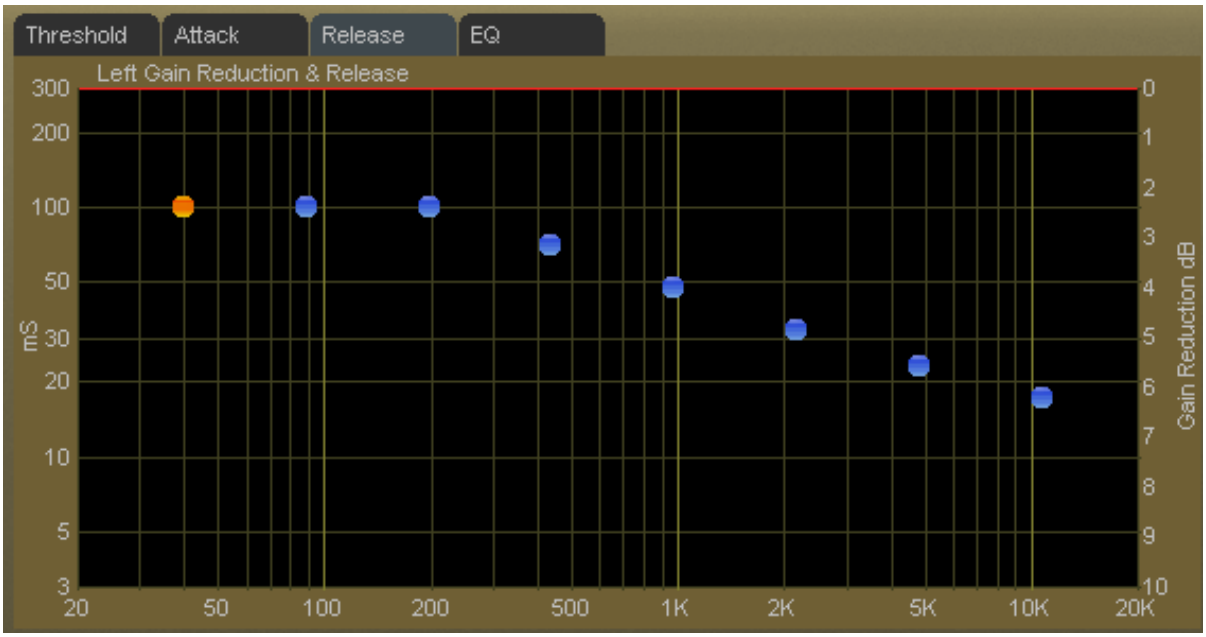
Attack times are generally adjusted to taste, keeping in mind that the slower the attack time the more work the final limiter must do. Conversely, the faster the attack time the fewer short term dynamics the program material will have.

Generally, lower frequencies require slower attack times and higher frequencies require faster attack times for a given amount of control.

The attack times of each band may be individually or collectively modified using the GUI operating instructions provided earlier (Chapter 4, page 4-15).

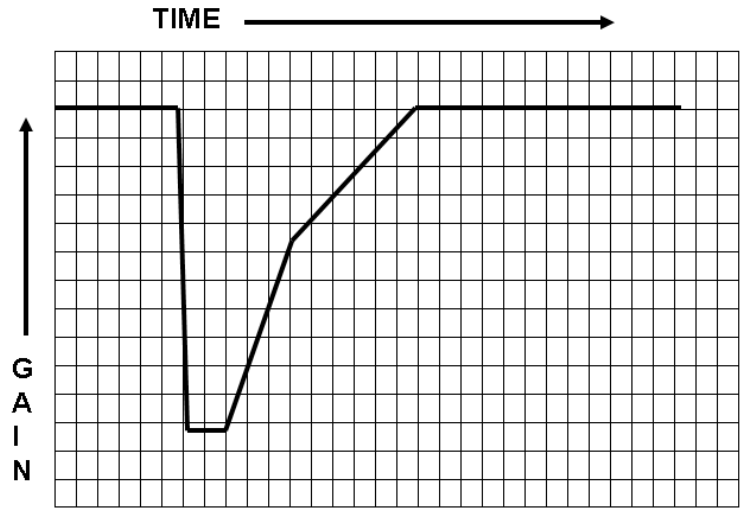
Release Time Screen

Clicking on the *Release* tab opens up the release time adjustment screen.



Multiband Limiter Release Time Screen

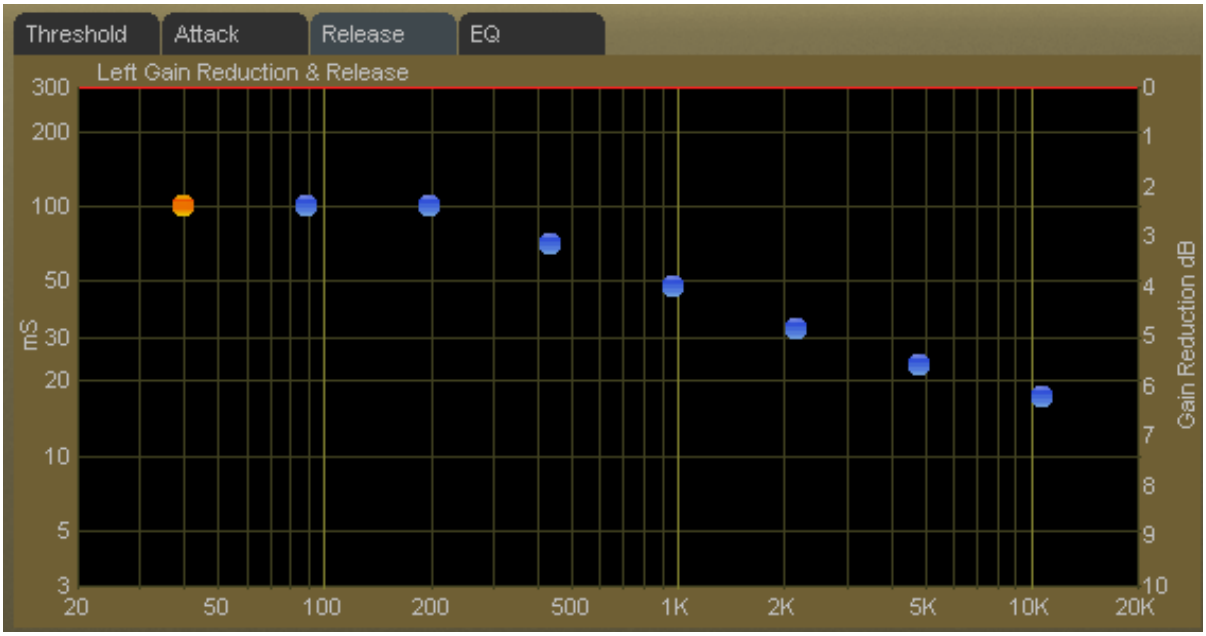
The release time of a limiter determines how fast it recovers from an audio peak once the peak has caused a momentary decrease in gain. Shorter (faster) release times create a denser sound, and excessively short release times might increase intermodulation distortion (the mixing of low and high frequency energy – or a muddying of the sound). The multiband limiters in the VP-8IP have Delayed Release timing built-in to minimize unintentional intermodulation distortion. See the graph below for a plot of this behavior (note that all gain trajectories are linear).



Limiter attack, followed by Gain Hold, followed by 3dB of Primary Release Time, followed by recovery at The Delayed Release Setting.

The release times for each band are adjustable from 300 milliseconds down to 3 milliseconds. Faster release allows a more rapid recovery after a limiting event, while slower release times do the opposite. Release times should be adjusted to taste, keeping in mind that the slower the release time the more dynamic the sound, but the more likely that audible ducking may be heard when a limiter enters the attack phase.

We very generally recommend that the eight band multiband limiter release times be operated with a tapering slope of about 150 milliseconds on the lowest band to approximately 30 milliseconds on the highest band as shown below. This is not a hard and fast rule, but a general suggestion that will sound good on a wide variety of program material. Your ears should always be the final judge on any setting.



Release Time Note:

The eight band limiter's Release times may be adjusted all the way down to 3 milliseconds, which is far *too fast* a setting for frequencies below about 1kHz. The reason we allow this to be done is that certain custom audio textures, especially in the bass region, may be created by *purposely* operating the lower frequency limiters with extremely fast release times.

Such settings, in effect, create distortion – on purpose. But because the resulting distortion is constrained to the narrow frequency bands within each of the VP-8IP's multiband limiters it is possible to create aesthetically pleasing warmth on the low end without it sounding like blatant distortion. This kind of adjustment is usually done only to create an effect, it must be done with great care, and it is therefore not recommended as standard operating procedure.

Final Equalization Screen

Because post limiter equalization is undesirable in a low-bitrate codec-based application there is no post limiter EQ available in the *MP3/AAC>48K* mode.

The Vorsis Spectral Energy Gate™

In order to successfully transmit pleasing sounding audio at low bitrates through a codec-based system it can be helpful to modify the spectral energy of the incoming program content (as it will be later seen by the codec) as program conditions change. The VP-8IP is equipped with a specially designed Vorsis algorithm that accomplishes exactly this.

The VP-8IP's Spectral Energy Gate™ is a complex multiband control structure. It is designed to manage audio program energy that has been deemed to be potentially troublesome for a codec. The single task of the Spectral Energy Gate™ is to find, through a series of look ahead calculations, opportunities to lower the priority of certain audio information that would, due to its spectral or temporal characteristics, contribute little to reconstructing a pleasing audio signal by the decoder.

By dynamically decreasing the workload of the encoder in real time, more bits are available for coding the most prominent audio signals. The resulting audio quality is not decreased as would be expected, but is actually increased, enhancing the listening experience.

The ears are *always* the best tools to use for adjusting processing controls. However, unlike the MP3/AAC<48K mode, in the VP-8IP's MP3/AAC>48K mode the Spectral Energy Gate™ is fully automatic and therefore requires no special user adjustment.

Note: We strongly suggest that the output of the VP-8IP NOT be used to evaluate the subjective sound quality of the codec system. Rather, listen to the output of the decoder as this is where sound quality should always be judged. This is because in order to alter the audio fed to the codec in such a way that it sounds better at the output of the decoder, the output of the VP-8IP may not sound very pleasing. This is no cause for concern – simply use the output of the decoder as the sound quality reference.

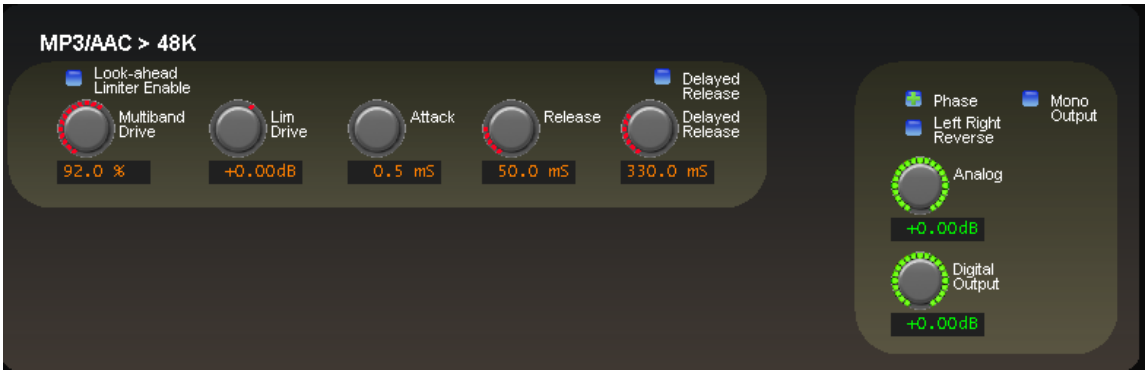
- The upside of the above is that maximum possible sound quality will be extracted from the combination of the VP-8IP + encode/decode process.
- The downside of the above is that the latency through the audio path, including all audio buffering, may approach several seconds which will necessarily make adjustment more time consuming.

Our specialized Vorsis algorithms can greatly improve the sound quality of low bitrate audio.

The VP-8IP's settings as well as the resulting sound quality rely strongly on the encoder being used (only "decoders" are defined by official standard... encoders are not!)

Please be aware that results are strongly biased by the quality of the incoming program material, the pedigree of the encoder in use and, of course, the frailties of our human and ever present personal preferences.

MP3/AAC>48K MENU



MP3/AAC>48K Screen

When operating in this mode it is assumed that the bitrate of the associated codec technology is high enough as to not yet require highly specialized algorithms in order to achieve great sound. Therefore the *MP3/AAC>48K* screen does not need myriad controls and the highly advanced algorithms that the VP-8IP's lower bitrate processing mode requires.

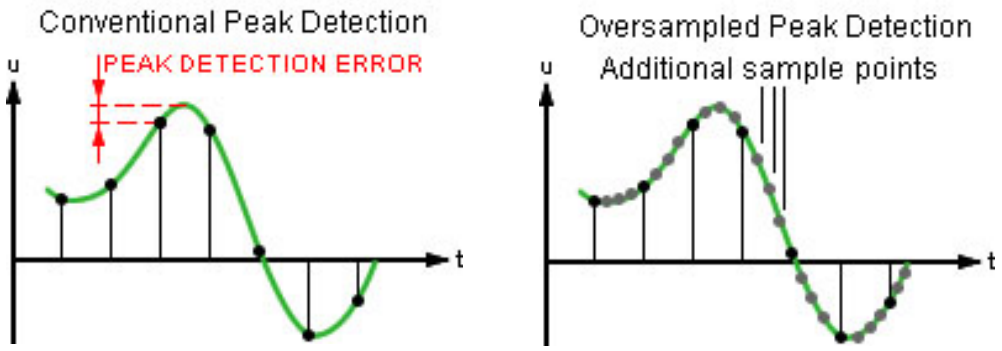
The *MP3/AAC>48K* screen is available for adjusting the characteristics of the look ahead limiter and Output signals,

Multiband Drive

This is a duplication of a control also found in the eight band *Limiters* screen. It is duplicated here in order to enable fine adjustments of the eight band limiter's sound without having to toggle back and forth between the *MP3/AAC>48K* and *Limiters* screens.

MP3/AAC>48K Look Ahead Limiter

The MP3/AAC>48K output may be accurately peak controlled by a highly oversampled feed forward limiter with a 0.5 millisecond (500 microseconds) look ahead time. The control path (not the audio path) is highly oversampled in order to achieve extremely precise peak control. The primary goal of oversampling in the control sidechain is better peak control through finer resolution of the audio waveform at the input to the detector. Oversampling of the audio signal itself is not only unnecessary, but wastes DSP resources. The graphic below illustrates why oversampling is more effective when the control signal is oversampled.



By sampling the audio waveform with finer resolution, peaks that would normally be missed because sample rates would fall "between the samples" are reliably detected. Even higher oversampling of the control signal would improve the accuracy of control, though the point of diminishing returns is rapidly reached – our lab research revealed that oversampling the *control* signal by more than about 4x over the audio has no real benefit in an audio application.

Note that if the audio *and* the control signals were *both* oversampled, peaks would be missed again because their relative time alignment, that is, peaks occurring between samples, would be identical to the non-oversampled case!

MP3/AAC>48K Look Ahead Limiter Controls

The graphic on the right shows the controls available for the look ahead limiter. The following text describes each control according to function.

Look-ahead Limiter Enable

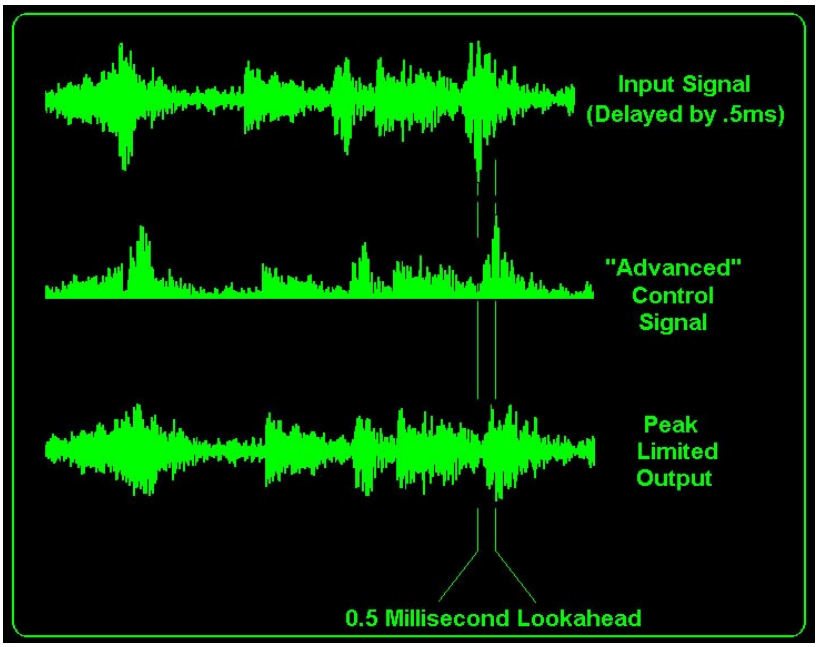
When this check box is checked the look head limiter is enabled. When the box is not checked no peak limiting is performed and the peak output levels will be uncontrolled and therefore undefined!

Lim (Limiter) Drive

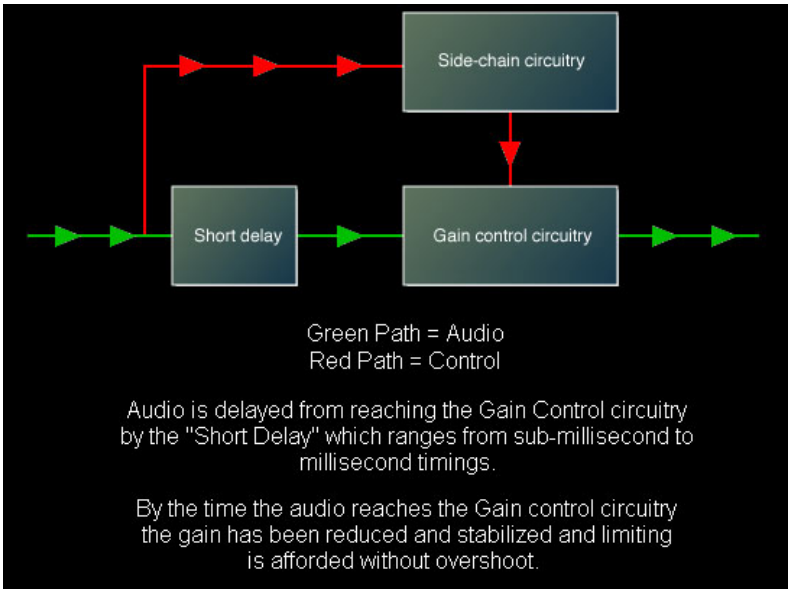
This controls the amount of drive to the look ahead limiter, noting that, unlike other VP-8IP operating modes there are no clippers available – all limiting must be done by the look ahead limiter. The gain range of this control is -18dB to+12dB. The resolution of this control is 0.05dB, fine enough for the most exacting drive adjustments.

Attack

This control adjusts the attack time of the look ahead limiter. In essence, a look ahead limiter "sees into the future" so that it can adapt to a new signal peak's level before the peak actually arrives. The graphics below illustrate how this works.



With a look ahead time of 0.5mS an attack time setting of 0.2mS allows the limiter to attack 0.3mS *before* the peak. Likewise, a setting of 0.5mS creates a limiter with zero attack time. All settings above this create a conventional peak limiter with the attack time calibration skewed negative 0.5 milliseconds – an attack setting of 1.0 millisecond is actually an attack time of 0.5 milliseconds because of the 0.5 millisecond look ahead which is *always* active.



Very fast attack times afford accurate energy control at the expense of musicality. Always choose the fastest attack time consistent with how you wish the output of the processor to sound, realizing that adjusting the limiter for longer attack times will allow some audio peaks to escape uncontrolled. How the attack time is adjusted is always a tradeoff between the accuracy of peak control and the musicality of the processor.

The accuracy of the calibration of the Analog and Digital Output controls is highly dependent on the setting of the look ahead limiter's attack time. This is because if the attack time is not short enough to catch all audio peaks, there will be overshoots and some peaks will escape – this will add to the peak level of the output. The magnitude of these overshoots will be highly program dependent and will almost certainly be unpredictable.

Because the look ahead limiter is the only mechanism by which peak levels at the output of the VP-8IP are accurately controlled, if the attack time is slowed down (set to larger numbers) in order to create a sound that is more open and less restrained then due consideration must be given to the resulting peak output levels. Why?

1. If the Digital and/or Analog output controls are operated near their maximum output levels and the look ahead limiter is operated with slow attack times, clipping will almost certainly occur in the digital domain before the signal ever leaves the VP-8IP.
2. If the Digital and/or Analog output level controls are operated at some negative number and the look ahead limiter is operated with slow attack times, the peak output level of the VP-8IP will be higher than the calibration of those controls may indicate. This can cause clipping and/or signal overload in the device that the VP-8IP is feeding.
3. The calibration of the Digital and Analog output level controls is valid only when the look ahead limiter is being operated with an attack time of 0.5 milliseconds or less which ensures that the peak output level is being defined by the peak limiter's threshold and not by some arbitrary program-related overshoot caused by limiter attack times that are simply too slow.

Release

Slower settings (longer release times) cause the peak limiter to recover more slowly from limiting; release times that are set too slow may allow undesirable signal ducking to occur because the audio level remains reduced too long after a peak limiting event.

Fast release times increase signal density, but release times that are too fast, even with the Delayed Release feature enabled, may result in some intermodulation distortion. When processing for a perceptual codec we recommend release times no faster than 60 milliseconds in order to minimize the coding of close-in intermodulation components resulting from fast release times.

The overall adjustment range of the control is from 33 milliseconds to 330 milliseconds, with an ideal setting for most applications falling between 70 and 130 milliseconds.

Delayed Release

Controls the secondary (slow) release of the look ahead limiter. The *Delayed Release* control has no effect unless the *Delayed Release* check box is checked.

When Delayed Release is engaged the limiter will release the first 3dB of gain reduction at the time constant set by the main *Release* setting. The remainder of the gain recovery will occur at the setting of the *Delayed Release* time control. The purpose of Delayed Release is to reduce intermodulation distortion, permitting deeper limiting depth without incurring an intermodulation distortion penalty.

Use of the Delayed Release mode is always recommended when processing ahead of a perceptual codec. As we discussed before, very fast limiter release times can generate close-in intermodulation products around desirable program signals which can degrade the perceived quality of the audio at the output of the codec more than just the action of the codec or limiters themselves would imply.

The *Delayed Release* control offers adjustments between 100 milliseconds and 1.000 seconds, though we do not recommend *Delayed Release* settings below approximately 190 milliseconds when processing for a perceptual codec operating at bitrates below about 48kBPS. An ideal setting for the majority of applications will fall between settings of 180 and 600 milliseconds.

Mono Output

When selected, the output of the VP-8IP is mixed to Mono – the left and right channels carry identical information, even if the input is in stereo. This control is useful for forcing an overall mono mixdown when desired.

Phase

When a Plus (+) shows in this check box the electrical phase of the left and right signals are in phase with the signal applied to the input of the VP-8IP. When a Minus (-) sign shows in this box the electrical phase of the left and right signals are flipped 180 degrees from the signal applied to the input of the VP-8IP. To change this setting merely click on it and its status will toggle.

Left Right Reverse

When this check box is checked the left and right channels are swapped. If it is noticed that the left and right channels appear to be interchanged at the output of the VP-8IP, please check this control first to ensure that it is not checked!

Analog

Controls the output amplitude of the analog output signal appearing on the left and right channel XLR's on the rear panel. This control can be set to OFF or to levels from -79.95dB (or -57.95dBu) to 0.00dB (or +24.0dBu) in 0.05dB steps.

Digital Output

Adjusts the *peak* audio signal level present at the AES/EBU digital output. The resulting output levels are simultaneously indicated in dBFS by the front panel and remote GUI bargraph meters. This control can be set to OFF or adjusted over the range of minus 35.95dBFS to 0dBFS.

Peak Output Level Note:

As mentioned previously, the "peak level" accuracy of the output level controls is highly dependent on the setting of the *Attack* time of the Look Ahead Limiter.

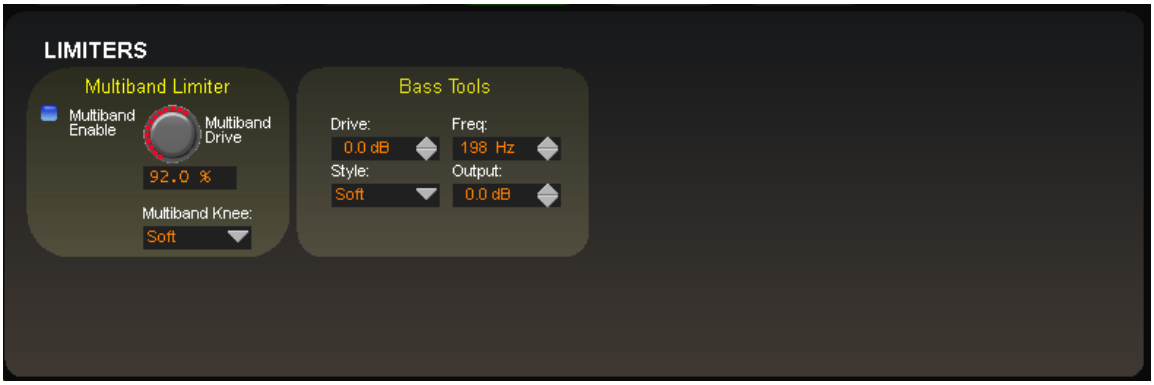
VP-8IP MP3/AAC<48K Process Mode

The following discussion is unique to the VP-8IP Audio Processor when it is operating in its *MP3/AAC<48K* mode.

There are two menu screens that are unique to the *MP3/AAC<48K* mode and used for adjusting how the processor's final limiting scheme sounds and interfaces with external transmission equipment. These are the *Limiters* and *MP3/AAC<48K* screens, the latter being selectable via the *MP3/AAC<48K* tab only when VP-8IP is in that codec processing mode.

Audio processing within the four band AGC takes place in the Sum and Difference domain, allowing tasteful management of the *MP3/AAC<48K* mode's stereo image without creating codec-teasing artifacts.

LIMITERS MENU - MP3/AAC<48K MODE



Limiters Screen in AAC/MP3<48K Mode

Multiband Limiter

The VP-8IP contains our exclusive, fully adjustable "swish-free" eight band final limiter algorithm. This limiter is especially effective in controlling program energy without generating unnatural swishing and "phasey" artifacts that outdated and simplistic multiband algorithms usually generate.

There are no caveats for using the eight band limiter other than it is inappropriate to operate the limiters continuously beyond 3dB of limiting when used with codecs. We generally recommend operating the limiters in a manner that allows them to do three things: gently increase RMS energy (loudness), shape the instantaneous HF spectrum to make it more suitable for the MP3/AAC<48K medium, and control longer duration transients that the preceding four band AGC/compressor intentionally misses in order to maintain an unrestrained sound quality.

For best musicality we recommend that the multiband limiters not be operated with extremely short attack and release times. The reasoning behind this is twofold:

1. Extremely short attack times, while affording virtually perfect energy control, remove so much transient energy that it makes the sound too homogenized and lifeless.
2. Even though our limiters utilize a delayed release algorithm, extremely short release times may increase certain coding artifacts. This is because fast release

times increase intermodulation distortion and the codec, seeing these new signals and not knowing that it is not audio, code it along with the audio to later reveal it as an unmasked artifact.

Please open several of our Factory MP3/AAC<48K presets and study the settings we've used in the eight band limiter section to see what we mean.

The eight band limiter's operating parameters are fully adjustable in the lower graphical screen in order to accommodate a wide variety of program material and competitive needs. It also contains the embedded Vorsis Bass Tools, which is tightly integrated within the limiter structure to intelligently control bass program energy. Please see the section of the VP-8IP Manual dedicated to GUI operation for how to accomplish the adjustment of the individual operating parameters of the eight-band section.

Multiband Enable

This check box determines whether or not any processing is applied by the eight band multiband limiter. When the eight band section is not enabled, audio peak control is handled solely by the look ahead limiter.

Multiband Drive

This control determines the amount of input signal applied to the eight band limiter and, therefore, how hard it is being driven. Lower *Multiband Drive* settings create a more gentle sound while still controlling program energy, while higher *Multiband Drive* settings make the sound more aggressive by increasing short term audio density. Very high drive settings resulting in more than 6-8dB of continuous gain reduction in any band is typically unnecessary as well as undesired.

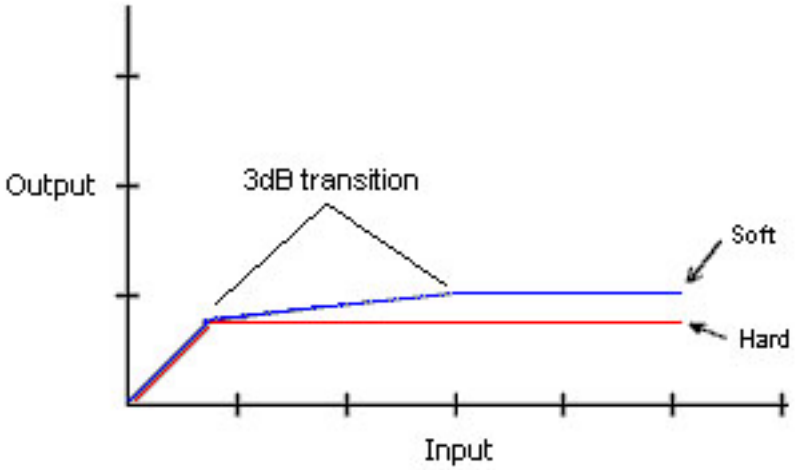
The *Multiband Drive* control setting is also somewhat dependent upon the setting of the *Makeup* control in the four band AGC's control screen. For this reason, the *Multiband Drive* control is not labeled in dB Drive, but instead is labeled in percent (%). With typical settings of the four band AGC's *Makeup* control (+20dB to +30dB) the correct eight band limiter *Multiband Drive* control setting will usually end up between 85% and 95%. The full adjustment range of the control is 0% to 100%.

Higher *Multiband Drive* settings cause deeper gain reduction in the eight band limiter (as well as higher levels of limiting in the final look ahead limiter if the *Lim Drive*, located in the *MP3/AAC<48K* screen, is not first reduced before the *Multiband Drive* is advanced). Therefore, *Multiband Drive* and *Lim Drive* are the pair of controls most useful for trading off "loudness," "density," and "distortion."

Multiband Knee

Each of the eight bands' threshold and attack and release times is adjustable, as is the overall transfer function (knee hardness). The knee shape determines how the limiters behave near their thresholds of limiting. The graphic below illustrates the difference between the two transfer functions.

In the *Hard* mode, limiting action does not occur until the threshold within a particular band is reached. This setting is more open sounding because no limiting activity occurs *until* the threshold is achieved and once limiting does begin, the compression ratio is very high.



Comparison of Hard and Soft Limiter Modes

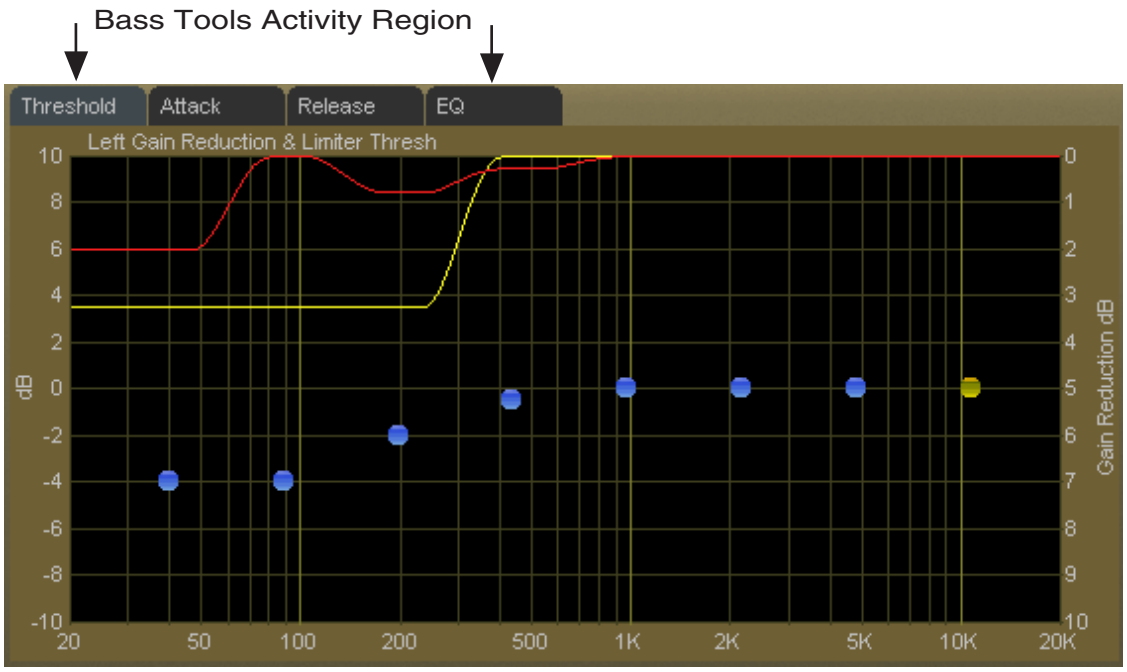
In the *Soft* mode, limiting starts to occur at a level that is 3dB below the user-set limiter threshold and the transfer function is relatively soft. That is, the audio waveform is not abruptly limited, but instead is allowed to gradually increase over a 3dB window until the ultimate limit threshold is reached, whereupon no further amplitude increase is allowed.

Because of this behavior, the sound of the *Soft* function is more obvious than the *Hard* setting, even though the latter is actually more aggressive in controlling levels. In this mode, and until the ultimate limit threshold is reached, the limiters operate more like variable ratio multiband compressors than multiband limiters.

Bass Tools

Vorsis Bass Tools allows extremely low program frequencies to be processed without causing dynamic gain changes and without generating undesirable intermodulation artifacts in higher frequency signals. Its primary purpose is to allow the thumps of kick drums and other percussion instruments to sound natural in their "percussiveness" without pumping or other undesired artifacts. The algorithm operates in left/right mode treating both channels identically.

Our laboratory research on codec behavior rather surprisingly revealed that it is perfectly acceptable to process very low frequency material using the Bass Tools algorithm provided that the harmonics generated during the process are well controlled spectrally. Since our bass algorithm does not allow significant harmonic levels above the third or fifth (somewhat dependent the Bass Tools *Style* setting) there is no audible downside to using it with coded audio. Furthermore, the ability to utilize the Bass Tools on both the FM and HD sides of the processor allows the sound to be more evenly matched as receiver-based digital to analog crossfades are encountered.



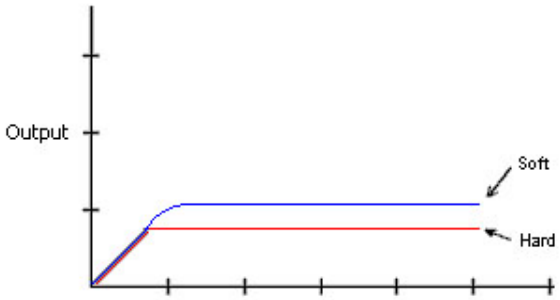
Bass Tools Activity Display - 20Hz up to VBMS Frequency Setting

Drive

This control adjusts the amount of Input Drive to the Bass Tools algorithm. Higher drive settings make it work harder and generate higher levels of harmonics. The *Drive* control adjusts the amount of drive to the Bass Tools and is adjustable over a range of -6dB to +12dB. Adjustment is best done by ear with percussive source material.

Style

Two Bass Tools operating styles, plus Off, are available. The *Hard* setting has a transfer function occupying approximately 1dB of transition amplitude. The waveform control is fairly abrupt and generates low order harmonics.



Comparison of Bass Tools Operating Modes

The *Soft* setting operates with a transfer function occupying approximately 2dB of transition amplitude. Therefore the control is less sharp than the hard setting, generating more low order harmonics. While the two settings may appear to be almost the same, the 1dB difference in transfer ratio creates a *significant* difference in how the two Bass Tools styles *sound* on actual program material. When Off is selected, the Bass Tools processing is bypassed.

Frequency

The *Freq* control is simple. All audio below the frequency selected is processed by Bass Tools. Smoother bass can be had with settings above 200Hz, which is recommended.

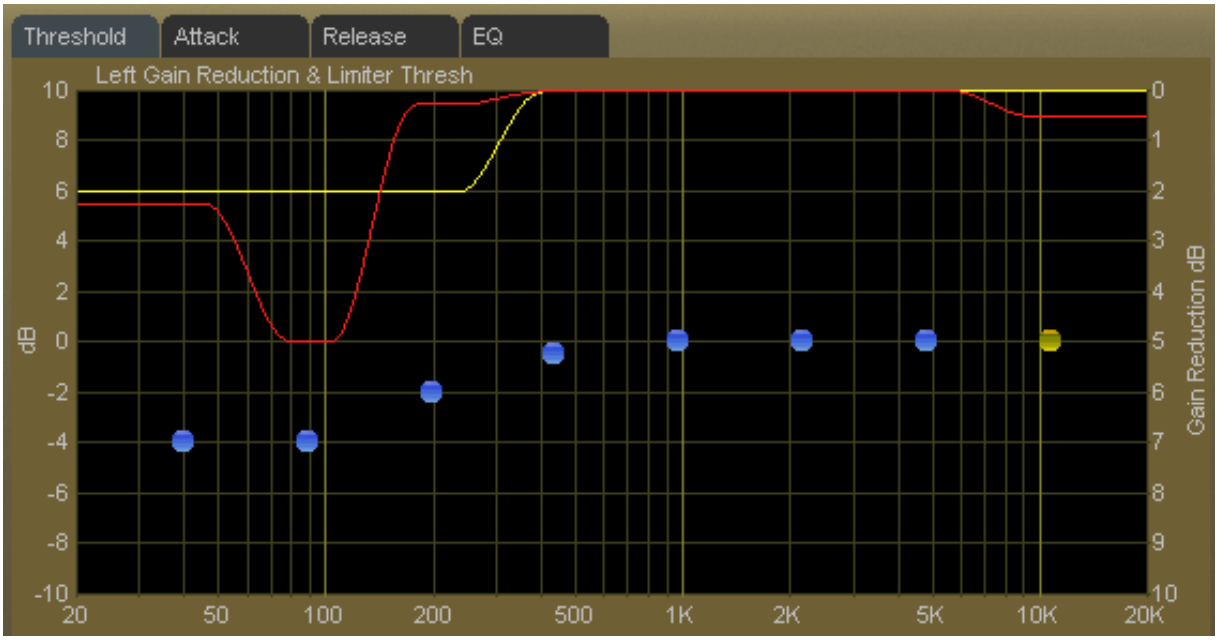
Output

The Bass Tools *Output* control adjusts the output of the Bass Tools algorithm and can be thought of as a "low bass mix" control. Settings between -6dB and -3.0dB add extra bass to the overall mix, while settings lower than -6.0dB reduce it. Both the Bass Tools *Drive* and the *Output* controls are best adjusted by ear using percussive bass material. The *Output* control is adjustable between -12dB and 0dB.

Multiband Limiter Adjustments

Threshold Screen

Clicking on the *Threshold* tab on the graphic opens up the adjustment screen for the limiter thresholds for each of the eight bands.



Multiband Limiter Threshold Screen

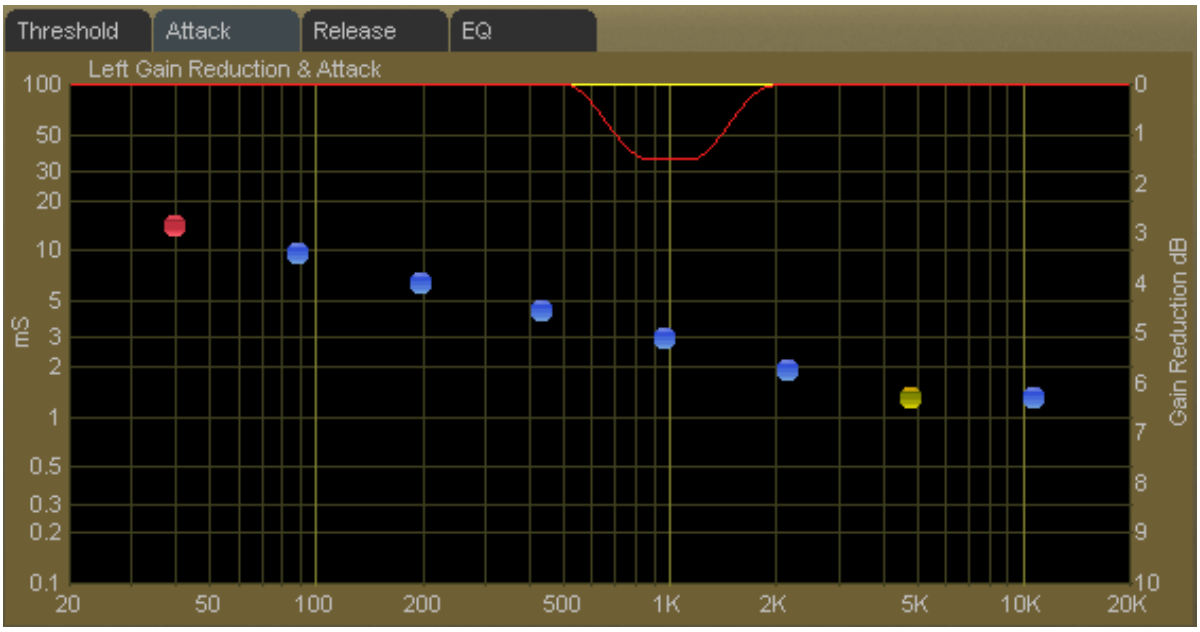
The limiter thresholds are adjustable over a +/-10dB range permitting frequency conscious limiting to be created if needed.

Lower *Threshold* settings (more negative numbers) reduce the threshold, causing limiting to occur at lower audio levels. Conversely, higher thresholds cause limiting to occur at higher audio levels.

Note: Radically different settings between adjacent bands won't cause strange sounds – there is ***no*** swishing possible in the Vorsis multiband limiter algorithm.

Attack Time Screen

Clicking on the *Attack* tab opens up the attack adjustment screen where blue dots (yellow/orange when selected) set the attack time of each of the eight limiter bands.



Multiband Limiter Attack Screen

The attack time of a limiter is a measure of how fast it responds to an audio peak when an audio peak exceeds the limiter threshold. Shorter attack times cause the limiter to respond to audio peaks quicker, affording more accurate peak control, but excessively short attack times, while creating "perfect" peak control, can literally suck the life out of musical material.

The attack times for each band are adjustable from 0.1 milliseconds (100 microseconds) to 100 milliseconds. As explained above, faster attack times exert more control over audio peaks, while slower attack times allow peaks to escape the limiter section with less control.

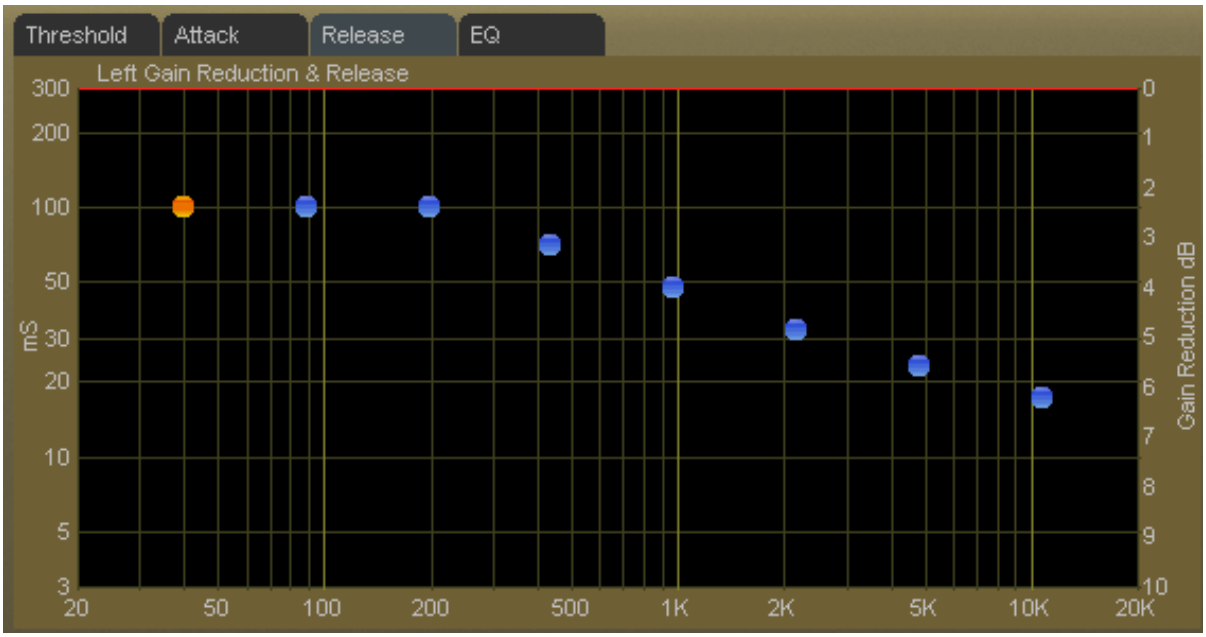
Attack times are generally adjusted to taste, keeping in mind that the slower the attack time the more work the final limiter must do. Conversely, the faster the attack time the fewer short term dynamics the program material will have.

Generally, for best musicality lower frequencies should employ slower attack times and higher frequencies faster attack times for a given amount of control.

The attack times of each band may be individually or collectively modified using the GUI operating instructions provided earlier (Chapter 4, page 4-15).

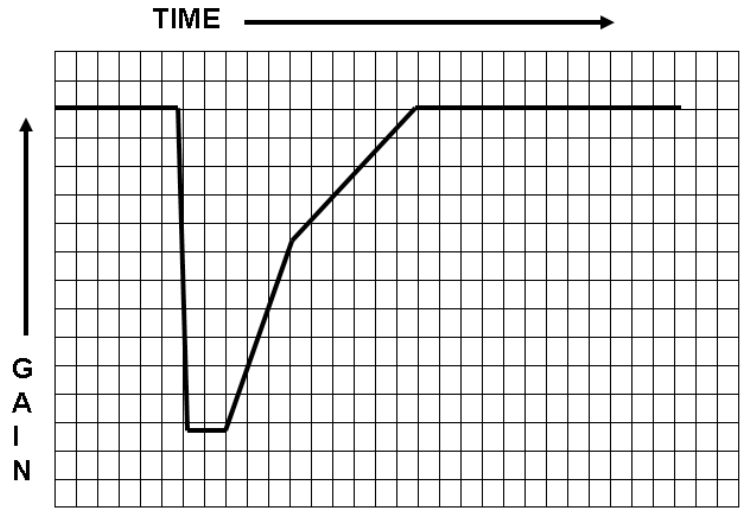
Release Time Screen

Clicking on the *Release* tab opens up the release time adjustment screen.



Multiband Limiter Release Time Screen

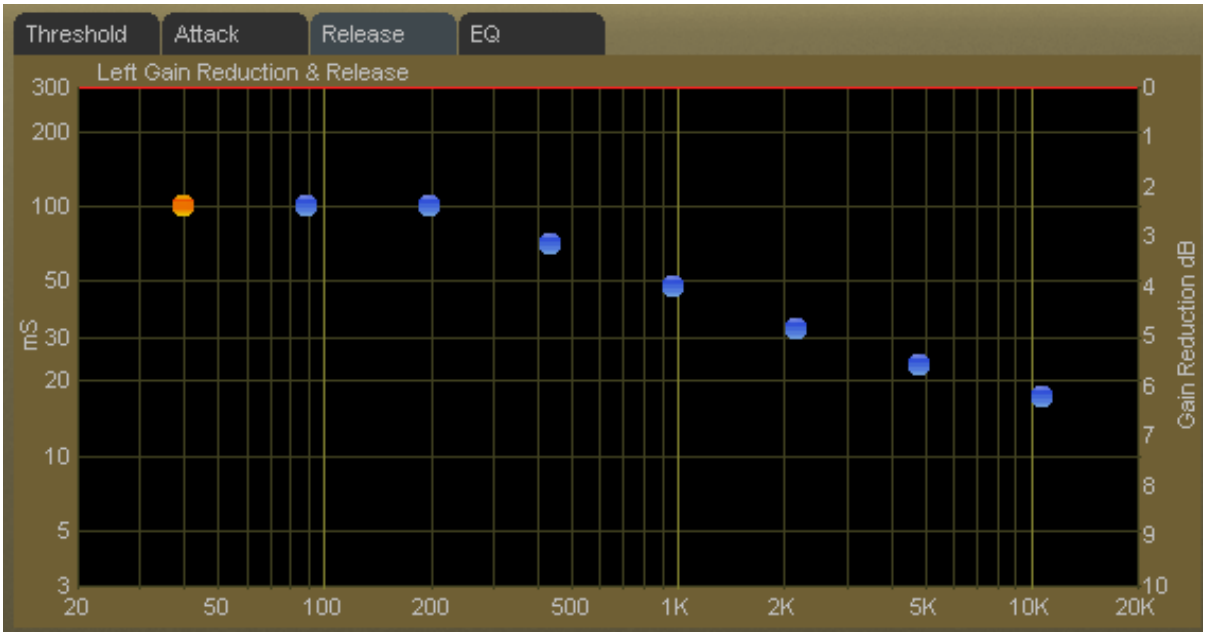
The release time of a limiter determines how fast it recovers from an audio peak after the peak has caused a momentary decrease in gain. Shorter (faster) release times create a denser sound, and excessively short release times might increase intermodulation distortion (the mixing of low and high frequency energy – or a muddying of the sound). The multiband limiters in the VP-8IP have built-in Delayed Release timing to minimize unintentional intermodulation distortion. See the graph below for a plot of this behavior.



Limiter attack, followed by Gain Hold, followed by 3dB of Primary Release Time, followed by recovery at The Delayed Release Setting.

The release times for each band are adjustable from 300 milliseconds down to 3 milliseconds. Faster release allows a more rapid recovery after a limiting event, while slower release times do the opposite. Release times should be adjusted to taste, keeping in mind that the slower the release time the more dynamic the sound, but the more likely that audible ducking may be heard.

We very generally recommend that the multiband limiter release times be operated with a tapering slope of about 150 milliseconds on the lowest band to approximately 30 milliseconds on the highest band as shown below. This is not a hard and fast rule, but a general suggestion that will sound good on a wide variety of program material. Your ears should always be the final judge on any setting.



Release Time Note:

The eight band limiter's *Release* times may be adjusted all the way down to 3 milliseconds, which is far too fast a setting for frequencies below about 1kHz. The reason we allow this to be done is that certain audio textures, especially in the bass region, may be created by *purposely* operating the lower frequency limiters with extremely fast release times.

In effect, such drastic settings create harmonic and narrow-band IM distortion. But because the resulting distortion is constrained to the quite narrow frequency bands of the VP-8IP's multiband limiters it is possible to create aesthetically pleasing "warmth" on the low end without blatant and irritating distortion.

This type of adjustment is usually done only to create an effect. It is done only on the lower three or four bands, it must be done with great care, and we do not recommend it as standard operating procedure. Please contact us for assistance or advice if you would like to experiment with this capability and are not familiar with how best to use it.

Final Equalization Screen

There is no post limiter EQ available in either the *MP3/AAC<48K* or *MP3/AAC>48K* modes.

MP3/AAC<48K MENU

MP3/AAC<48K Screen

When operating in this mode it is assumed that the bitrate of the associated codec technology is low enough to require highly specialized algorithms in order to achieve great sound. Consequently this mode is more complex and there are additional controls on the *MP3/AAC<48K* screen compared to that of the higher bitrate mode.

Multiband Drive

This is a duplication of a control also found in the eight band *Limiters* screen. It is duplicated here in order to enable fine adjustments of the eight band limiter's sound without having to toggle back and forth between the *MP3/AAC<48K* and *Limiters* screens as the final limiter is adjusted.

The Vorsis Spectral Energy Gate™

In order to successfully transmit pleasing sounding audio at low bitrates through a codec-based system it can be helpful to modify the spectral distribution of incoming program energy (as it will be seen by the codec) as program conditions change. The VP-8IP is equipped with a specially designed Vorsis algorithm (with its associated controls) that accomplishes exactly this.

This sophisticated audio processing algorithm, the exclusive Spectral Energy Gate™ , is tailored specifically for use with low bitrate perceptual codecs. Though it can be utilized with all audio codecs, it really shines when used in combination with codecs operating at bitrates below approximately 48kBPS.

The VP-8IP's Spectral Energy Gate™ is a complex multiband control structure for managing audio program energy that, through real-time analysis in look ahead fashion, has been deemed to be potentially troublesome for a low bitrate codec.

The Spectral Energy Gate™ operates in conjunction with the controls shown on the *MP3/AAC<48k* Screen and though it is multiband in structure it is completely independent of the multiband AGC and eight band limiter.

The purpose of the Spectral Energy Gate™ is to find, through a series of look ahead calculations, opportunities for lowering the priority of certain audio information that, because of its spectral or temporal characteristics would contribute minimal improvement during reconstruction of the signal at the decoder.

Dynamically decreasing the workload of the encoder in real time allows more bits to be available for coding the more prominent (and therefore more audible) audio data. The resulting audio quality is actually increased, which enhances the listening experience.

Adaptive Bandwidth Controller™ Controls

The VP-8IP also incorporates an Adaptive Bandwidth Controller™ as one of its tools for managing the behavior of low bitrate codecs. There are five controls associated with the Adaptive Bandwidth Controller™ which will be discussed in order.

Codec Sample Rate

The *Codec Sample Rate* control has two positions, *48/32/24k* and *44.1/22.05k* which correspond to different multiband operating parameters within the Adaptive Bandwidth Controller™ .

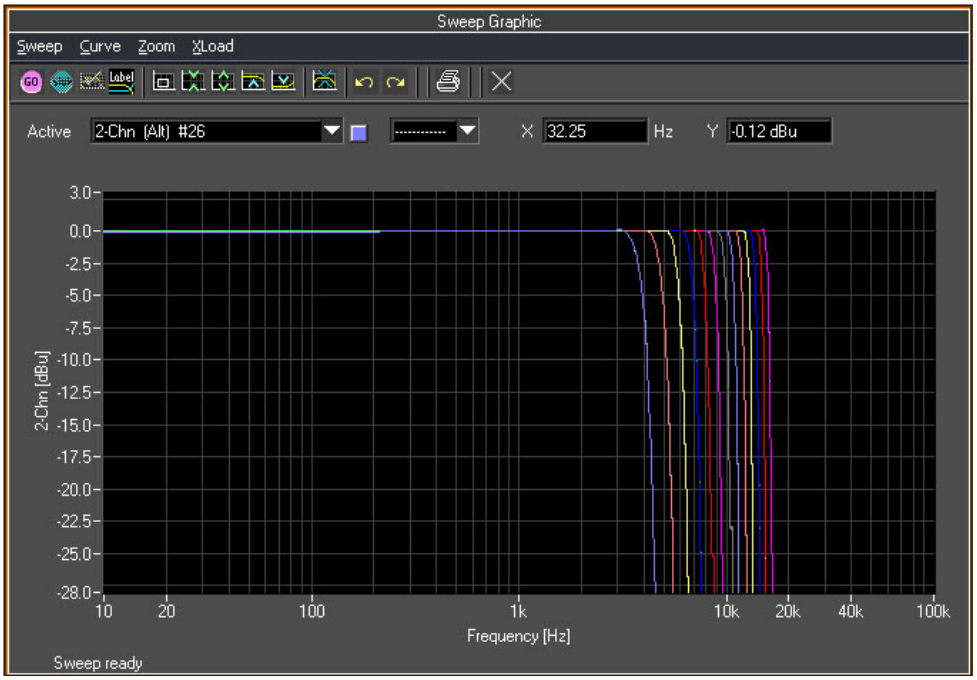
The differences between the two settings serve important roles in achieving the best sound from codecs at different sample rates. We recommend utilizing the 48/32/24kHz or 44.1/22.05kHz position for codecs operating at those sample rates. Sample rates lower than 22.05kHz may benefit most from the use of the 44.1/22.05kHz setting.

Low Pass Filter

The purpose of the low pass filter is to remove program energy higher than the frequency selected by the control. By removing audio energy above a certain frequency, higher subjective audio quality can be achieved by the codec when operating at a particular bitrate. Low pass filters become especially important at very low bitrates because they remove energy that the codec does not need to contend with in order to provide a pleasing listening experience at the decoder output (we consider any bitrate below 32kHz to be very low).

The graphic at right is a family of sweeps displaying the characteristics of the Low Pass Filter when set at each of its available cutoff frequencies.

Note the steep low-pass characteristics and the perfect ruler-flat passband flatness at each filter setting.



Codec Bitrate

When the bitrate is very low, special care must be taken to prevent the codec from generating annoying artifacts that would not be present if higher bitrates were used. While it is impossible to prevent all codec-related artifacts when operating at very low bitrates we've found that special processing of certain types of program material can go a long way towards creating a pleasing listening experience. And while such special processing may reduce certain forms of program information, this behavior can actually make the codec sound better than if such processing were not utilized.

Mask Thresh and Mask Recovery

The *Codec Bitrate* control operates in conjunction with the *Mask Thresh* and *Mask Recovery* controls, which belong to the algorithm that manipulates incoming audio to reduce codec artifacts commonly referred to as "splashiness" and "swirliness." It operates by analyzing and controlling certain characteristics of the incoming program after all other processing has been performed by the VP-8IP.

Achieving the Proper Settings

There is naturally some interaction between the *Mask Thresh* and *Mask Recovery* controls, although achieving the proper settings is easy if the following guidelines are used.

First, set the *Codec Bitrate* control according to the bitrate being used. Next set the *Mask Recovery* control to 330mS and the *Mask Thresh* control to 0dB. Then adjust the *Mask Thresh* control counter-clockwise until the onset of high frequency dulling is just barely detected. Once this point is achieved, the control should be backed off by 1dB (adjust it one dB more positive).

Next the *Mask Recovery* control is adjusted until the point is found where any remaining dulling effect seems minimized. Once this point is found, add 30mS to the setting and move the control to that new value.

Example:

Suppose the *CodecBitrate* control is set to 32kBPS and the *Mask Recovery* control is at 330mS. While adjusting the *Mask Thresh* it is determined that the point where high frequency dulling becomes noticeable is -7.0dB. Therefore the proper setting of the control is -7dB plus 1dB, or -6.0dB.

While adjusting the *Mask Recovery* control it is noticed that residual dulling becomes minimized with the control set to 70mS. Therefore the correct setting will be 70mS + 30mS, or 100mS.

The *Codec Bitrate*, *Mask Thresh* and *Mask Recovery* controls are always adjusted for the best subjective results at the output of the codec's decoder.

Each control's setting is highly dependent on outside factors not associated with the VP-8IP's internal algorithms and a deep understanding of the algorithms living under the hood is not required in order to extract maximum audio performance. Therefore an explanation as to the inner workings of the Bitrate and Mask algorithms is unnecessary because the ears are always the best tools to use for adjusting these and other processing controls.

Note: The above suggests (and strongly so) that the output of the VP-8IP should NOT be used to evaluate the subjective sound quality of the codec system. Rather, it is the output of the decoder that should be used to judge sound quality. This is because, in order for the VP-8IP processing to alter the audio fed to the codec in such a way that it sounds better at the output of the decoder, the output of the VP-8IP might not sound very pleasing. This is no cause for concern – simply use the output of the decoder as the sound quality reference.

- The upside of the above is that maximum possible sound quality will be extracted from the encode/decode process.
- The downside of the above is that the latency through the audio path, including all audio buffering, may approach several seconds which will necessarily make adjustment more time consuming.

Our specialized Vorsis algorithms can greatly improve the sound quality of low bitrate audio.

The VP-8IP's settings as well as the resulting sound quality rely strongly on the encoder being used (only "decoders" are defined by official standard... encoders are not!)

Please be aware that results are strongly biased by the quality of the incoming program material, the pedigree of the encoder in use and, of course, the frailties of our human and ever present personal preferences.

MP3/AAC<48K Look Ahead Limiter

The MP3/AAC<48K output may be accurately peak controlled by a highly oversampled feed forward limiter with a 0.5 millisecond (500 microseconds) look ahead time. The control path (not the audio path) is highly oversampled in order to achieve extremely precise peak control. The primary goal of oversampling in the control sidechain is better peak control through finer resolution of the audio waveform at the input to the detector. Oversampling of the audio signal itself is not only unnecessary, but wastes DSP resources. The graphic below illustrates why oversampling is more effective when the control signal is oversampled.

By sampling the audio waveform with finer resolution, peaks that would normally be missed because they would fall "between the samples" are reliably detected. Even higher oversampling of the control signal would improve the accuracy of control, though the point of diminishing returns is rapidly reached – our lab research revealed that oversampling the *control* signal by more than about 4x over the audio has no real benefit in an audio application.

Note that if the audio *and* the control signals were *both* oversampled, peaks would be missed again because their relative time alignment, that is, peaks occurring between samples, would be identical to the non-oversampled case!

MP3/AAC<48K Look Ahead Limiter Controls

The graphic on the right shows the controls available for the look ahead limiter. The following text describes each control according to function.

Look-ahead Limiter Enable

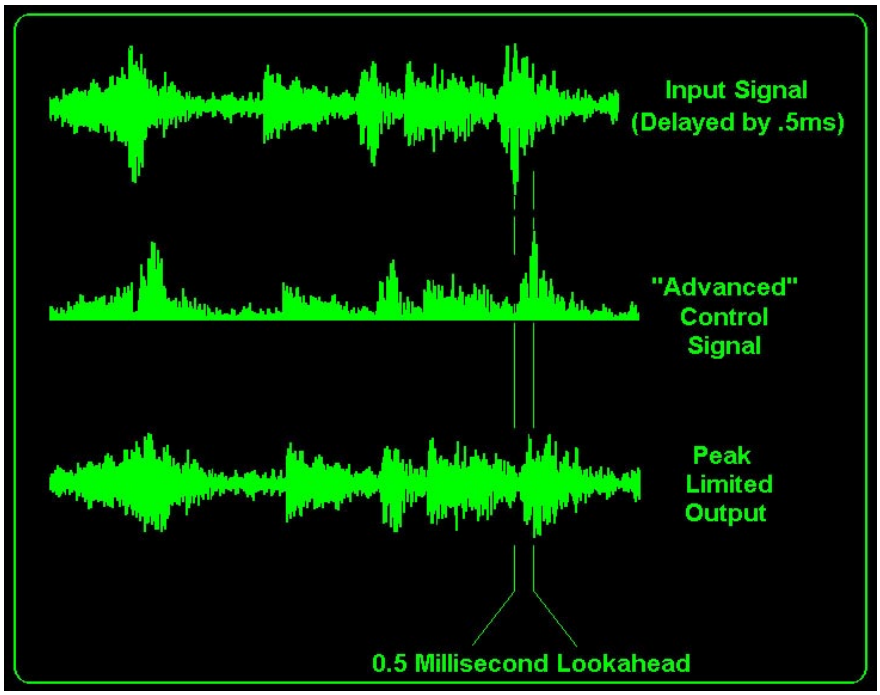
When this check box is checked the look head limiter is enabled. When the box is not checked no peak limiting is performed and the peak output levels will be uncontrolled and therefore undefined!

Lim (Limiter) Drive

This controls the amount of drive to the look ahead limiter, nothing that, unlike other VP-8IP operating modes there are no clippers available – all limiting must be done by the look ahead limiter. The gain range of this control is -18dB to+12dB. The resolution of this control is 0.01dB, fine enough for the most exacting drive adjustments.

Attack

This control adjusts the attack time of the look ahead limiter. In essence, a look ahead limiter "sees into the future" so that it can adapt to a new signal peak's level before the peak actually arrives at the control mechanism. The image below demonstrates how this works.



With a look ahead time of 0.5mS an attack time setting of 0.2mS creates a limiter that attacks 0.3mS before the peak. Likewise, a setting of 0.5mS creates a limiter with zero attack time (because the look ahead is 0.5mS). All settings above 0.5mS create a conventional peak limiter with the attack time calibration skewed negative 0.5 milliseconds. That means that an attack setting of 1.0 millisecond is actually an attack time of 0.5 milliseconds because of the 0.5 millisecond look ahead which is always active. The full adjustment range is 0.2mS to 100mS.

Very fast attack times afford accurate peak control at the expense of musicality. Always choose the fastest attack time consistent with how you wish the output of the processor to sound, realizing that adjusting the limiter for longer attack times will allow some audio peaks to escape uncontrolled. How the attack time is adjusted is always a tradeoff between the accuracy of peak control and the musicality of the processor.

Note that the accuracy of the Analog and Digital Output controls, that is, the peak output levels occurring at their settings, is highly dependent on the setting of the look ahead limiter's attack time. This is because if the attack time is not short enough to catch all audio peaks, there will be overshoots and some peaks will escape adding to the peak level of the output. The magnitude of these overshoots will be highly program dependent and will almost certainly be unpredictable.

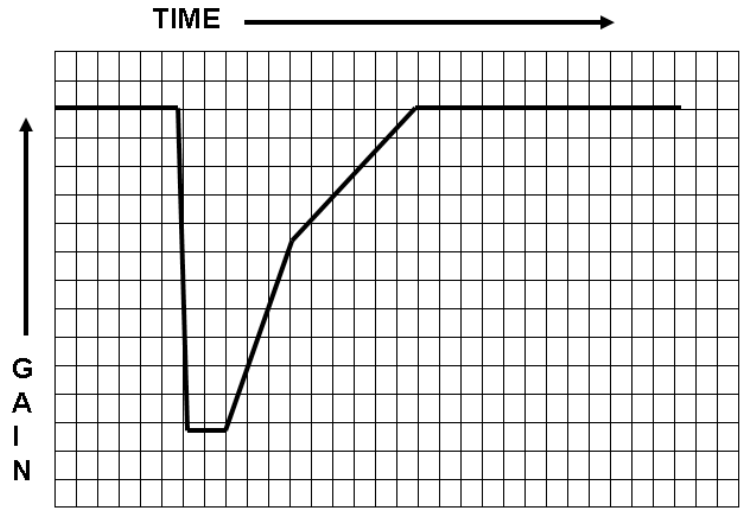
Because the look ahead limiter is the only mechanism by which peak levels can be accurately controlled, if the attack time of this limiter is slowed down (set to larger numbers) in order to create a sound that is more open and less restrained then due consideration must be given to the resulting peak output levels. Why?

1. If the Digital and/or Analog output controls are operated near their maximum output levels settings and the look ahead limiter is operated with slow attack times, clipping will almost certainly occur in the digital domain before the signal ever leaves the VP-8IP.
2. If the Digital and/or Analog output level controls are operated at some negative number and the look ahead limiter is operated with slow attack times, the peak output level of the VP-8IP will be higher than what the output control's calibration might indicate. This can cause clipping and/or signal overload in the device that the VP-8IP is feeding.
3. The calibration of the Digital and Analog output level controls is valid only when the look ahead limiter is being operated with an attack time of 0.5 milliseconds or less which ensures that the peak output level is being defined by the peak limiter's threshold and not by some arbitrary program-related overshoot caused by limiter attack times that are simply too slow.

Release

Release controls the primary (fast) release time of the look ahead limiter.

The look ahead limiter utilizes the same smart release algorithm that the multiband limiter is equipped with. In essence, release is delayed temporarily (regardless of whether *Delayed Release* is selected) in a program-controlled manner that minimizes distortion due to limiting. The graphic below illustrates this behavior.



Limiter attack, followed by Gain Hold, followed by 3dB of Primary Release Time, followed by recovery at The Delayed Release Setting.

Slower settings (longer release times) cause the peak limiter to recover more slowly from limiting; release times that are set too slow may allow undesirable signal ducking to occur because the audio level remains reduced too long after a peak limiting event.

Conversely, fast release times increase signal density, but release times that are too fast, even with the Delayed Release feature enabled, may result in some intermodulation distortion that can "muddy" the sound. When processing for a perceptual codec we

recommend release times no faster than 60 milliseconds in order to minimize the coding of close-in difference frequency intermodulation components resulting from fast release times.

The overall adjustment range of the control is from 33 milliseconds to 330 milliseconds with an ideal setting for most applications being between 70 and 130 milliseconds.

Delayed Release

Controls the secondary (slow) release of the look ahead limiter. The *Delayed Release* control has no effect unless the *Delayed Release* check box is checked.

When *Delayed Release* is engaged via the *Delayed Release* check box the limiter will recover the first 3dB of gain reduction at the time constant set by the main *Release* setting. The remainder of the gain recovery occurs at the *Delayed Release* time constant.

The purpose of Delayed Release is to afford a drastic reduction in intermodulation distortion which generally allows deeper limiting depth without incurring the usual intermodulation distortion penalty.

Use of the Delayed Release mode is always recommended when processing ahead of a perceptual codec. As we discussed before, very fast limiter release times can generate close-in intermodulation products around desirable program signals, which may degrade the perceived quality of the audio at the output of the codec more than just the action of the codec or limiters themselves would imply. The use of Delay Release mitigates this distortion, improving the sound of the codec at any practical bitrate.

The *Delayed Release* control offers adjustments between 100 milliseconds and 1.000 seconds, though we do not recommend *Delayed Release* settings below approximately 190 milliseconds when processing for a perceptual codec operating at bitrates below about 48kBPS. An ideal setting for the majority of applications will fall between 180 and 600 milliseconds.

Mono Output

When selected, the output of the VP-8IP is mixed to Mono – the left and right channels carry identical information, even if the input is in stereo. This control is useful for forcing mono when desired.

Phase

When a Plus (+) shows in this check box the electrical phase of the left and right signals are in phase with the signal applied to the input of the VP-8IP. When a Minus (-) sign shows in this box the electrical phase of the left and right signals are flipped 180 degrees from the signal applied to the input of the VP-8IP. To change this setting merely click on it and its status will toggle.

Left Right Reverse

When this check box is checked the left and right channels are swapped. If it is noticed that the left and right channels appear to be interchanged at the output of the VP-8IP, please check this control first to ensure that it is not checked!

Analog

Controls the output amplitude of the analog output signal appearing on the left and right channel XLR's on the rear panel. This control can be set to OFF or to levels from -79.95dB (or -57.95dBu) to 0.00dB (or +24.0dBu) in 0.05dB steps.

Digital Output

Adjusts the *peak* audio signal level present at the AES/EBU digital output. The resulting output levels are simultaneously indicated in dBFS by the front panel and remote GUI bargraph meters. This control can be set to OFF or adjusted over the range of minus 35.95dBFS to 0dBFS.

Peak Output Level Note:

As mentioned previously, the "peak level" accuracy, that is, the calibration, of the output level controls is highly dependent on the setting of the *Attack* time of the Look Ahead Limiter.

Appendix

Contents

Parameters, Units and Ranges.....A-2

Parameters, Units and Ranges

Approximately following the VP-8IP's signal path, these are the values and ranges appropriate to each type of processing.

System Level

Headroom:	20dB
Nominal Operating Level:	-20dBFS digital

Analog Line Input

Type:	Electronic Differential
Input Impedance:	> 10Kohm (bridging)
Optimum Source Impedance:	< 1Kohm

Digital Line Input

Data Standard:	AES3 (AES/EBU)
Data Amplitude:	Per AES3-2003 assuming minimum allowable output signal amplitude of 2V and minimum allowable input signal amplitude of 200mV

Input Gain Adjustment Ranges

Gain Adjustment:	-36dB to +12dB
Gain Adjustment Increments:	0.5dB
Gain Calibration:	A gain control setting of 0.0 aligns an external 0dBFS signal with the VP-8IP's 0dBFS internal reference

Input Failsafe

Type:	Automatic
Analog Fail Cause:	Audio level below -48dBFS.
Response Time:	30 seconds
Digital Fail Cause 1:	Audio level below -48dBFS
Response Time:	30 seconds
Digital Fail Cause 2:	Corrupted or invalid AES data
Response Time:	Immediate

Audio Level Balance

Type:	Common to Analog and Digital inputs
Analog/Digital L/R Balance Range:	+/-12dB
Analog/Digital L/R Balance Increments:	0.5dB

Voice Symmetry Phase Rotation

Operating Modes:	In/Out
Filter Type:	4th Order Allpass

High-Pass Filter

HPF Filter Class:	24dB/octave Butterworth
Frequency range:	20Hz – 300Hz
HPF Insert Modes:	Off / Stereo L/R / Sum/Difference

Equalization - Two sections of four-band Parametric Equalization

EQ Set 1 – routable to pre or post-five-band AGC.
EQ Set 2 – post-multiband limiter.

Frequency:	20Hz – 20 kHz
Bandwidth:	0.2 – 3.0 octaves
Lift / Cut:	+/- 14.0dB pre/post four band AGC
	+3/-12.0dB post multiband limiter

Four-Band AGC/Compressor

Operates in sum/difference, or "matrix," mode with the gain linked at AGC time constants.

Drive Gain:	-79.0dB to +6.0dB in 0.5dB steps, plus OFF
Makeup Gain Range:	+18.0dB to +24.0dB, 0.5dB steps
AGC/Compressor Thresholds:	- 35dBFS to -65dBFS
AGC Threshold Backoff:	0dB to -6dB
AGC Attack:	50mS – 1500ms (1.5 sec)
AGC Release:	100mS – 7.0 seconds
AGC Band Coupling:	0dB to -6.0dB, 0.5dB steps
Compressor Attack:	3.0mS – 1000mS (1Sec)
Compressor Release:	20mS – 1000mS (1Sec)
Ratio:	2:1 – 20:1
AGC Gate Threshold:	-20.0dB to -79dBFS, plus OFF
Gate Thresh Trim:	+/- 6dB per band
Gated Modes:	Slow Recovery/Gain Freeze
Gate Delay:	50mS to 500mS

Crossover Frequencies

Low to Low Mid Band:	50 Hz to 198 Hz
Low Mid Band to High Mid Band:	250 Hz to 1.5 kHz
High Mid Band to High Band:	3.0 kHz to 8.0 kHz
Band Output Trims, Sum:	+/- 6dB
Band Output Trims, Difference:	+/- 6dB

Eight Band FM Output Peak Limiter

A four band parametric EQ follows the eight band limiter and feeds the Bass Tools with multiple operating styles, then an oversampled, zero overshoot look ahead limiter or main clipper with multiple operating styles, and then an FM diversity delay.

Overall Controls	
Multiband Drive:	80 – 100%
Bass Tools Drive:	- 6.0/+12.0dB
Bass Tools Style:	Soft/Hard
Bass Tools Output:	- 12.0/0.0dB
Bass Tools Frequency:	60 - 300Hz
Multiband Knee:	Soft/Hard
Multiband Threshold:	+/-10dB relative to Drive
Multiband Attack:	0.1ms – 100ms
Multiband Release:	3ms – 300ms
Pre-emphasis Modes:	Off/50uS/75uS
Pre-emphasis insert points:	Pre-MB Limiter
	Pre-look ahead limiter
	Post-look ahead limiter (pre-clipper)
Look Ahead Limiter Attack:	0.2ms – 100ms
Look Ahead Limiter Release:	33ms – 330ms
Look Ahead Limiter Delayed Release:	100ms – 1000ms (1Sec)
Main FM Clipper Styles:	Hard/Firm/Round
De-emphasis:	Complementary as applied in pre-emphasis.

FM

Stereo Encoder (FM Mode Only)

Reference grade stereo encoder with embedded composite processing, test oscillator, SCA digitizer, and balanced and unbalanced composite outputs.

Overall Controls	
Stereo Width Limiter:	10% to 100% in 10% steps, plus OFF
BS-412 MPX Power Controller	+9.0dB to -2.0dB in 0.05dB steps
Stereo Pilot Injection:	0 – 20%, 0.1% steps
Stereo Pilot Phase:	+/- 22.5 degrees reference to 38kHz.
SCA 1 Input:	Analog, 10kohm input impedance,
	+24dBu max. input level
SCA Input 1 Gain:	-79.95dB to +10.00dB, or OFF

SCA 2 Input:	Analog, 10kohm input impedance, +24dBu max. input level
SCA Input 2 Gain:	-79.95dB to +10.00dB, or OFF
TX 1 Output Level:	-79.95dB to +6.0dB, 0.05dB steps, or OFF
TX 2 Output Level:	-79.95dB to +6.0dB, 0.05dB steps, or OFF May be selected to provide pilot tone only
Composite Processor Modes:	Oversampled look ahead limiter Soft Clipper Hard Clipper
Stereo Encoder Output Operating Modes:	Analog L/R Analog L/R De-emphasized Unbalanced and Balanced Multiplex
Output signals available as:	
Digital:	AES 3-pin XLR (digital), and Stereo, pre or post Diversity Delay Baseband192
Analog:	Analog Left/Right Analog Left/Right, De-emphasized Composite Multiplex, TX1 and TX2
Test Oscillator:	
Waveform:	Sine Wave
Frequencies:	50Hz to 80kHz plus 31.25kHz Bessel
Output Level:	0 to 100% in 1.0% steps
Digital Output Level:	-35.90dB to 0.0dB, or OFF

FM

Eight Band AM Output Peak Limiter

A four band parametric EQ follows the eight band limiter and feeds the Bass Tools with multiple operating styles, then an oversampled, zero overshoot look ahead limiter or main clipper with multiple operating styles, and then an AM diversity delay.

Overall Controls	
Multiband Drive:	80 – 100%
Bass Tools Drive:	- 6.0/+12.0dB
Bass Tools Style:	Soft/Hard
Bass Tools Output:	- 12.0/0.0dB
Bass Tools Frequency:	60 - 300Hz
Multiband Knee:	Soft/Hard
Multiband Threshold:	+/-10dB relative to Drive
Multiband Attack:	0.1ms – 100ms

AM

Multiband Release:	3ms – 300ms
Pre-emphasis Modes:	Off/NRSC
Pre-emphasis insert points:	Pre-MB Limiter
	Pre-look ahead limiter
	Post-look ahead limiter (pre-clipper)
Look Ahead Limiter Attack:	0.2ms – 100ms
Look Ahead Limiter Release:	33ms – 330ms
Look Ahead Limiter Delayed Release:	100ms – 1000ms (1Sec)
Main FM Clipper Styles:	Hard/Firm/Round
De-emphasis:	Complementary as applied in pre-emphasis.

AM Transmission

This screen permits access to the controls unique to the VP-8IP's AM operating mode. Functionally it follows the eight band AM limiter.

Overall Controls

Bandwidth:	Full (20kHz), 10kHz NRSC, 7.5kHz, 6.5kHz 6kHz CCIR, 5.5kHz, 5kHz, and 4.5kHz
Output Mode:	Mono, Stereo, Stereo M/S
Positive Modulation:	100% to 150% in 1% steps
Digital Output:	AES 3-pin XLR (digital), and Stereo, pre or post Diversity Delay, pre or post DeEmphasis

AM

Test Oscillator:	
Waveform:	Sine Wave/Clipped Sine Wave
Frequencies:	25Hz to 12kHz
Level:	0% to 100% in 1.0% steps
Tilt Test:	Provides clipped sine wave at 50% of the "Level" amplitude

LF Tilt Equalizer:

Equalizer Type:	Parametric Bandpass
Frequency (F):	5.0Hz to 100Hz
Bandwidth (Q):	0.25 to 3.00
Level (L):	0.0dB to 6.00dB, in 0.5 steps

(MONO mode has LF Tilt Equalizers available for both transmitter outputs)

Outputs:

Analog 1:	-79.95dB to +0.0dB, 0.05dB steps, or OFF
Phase	0 or 180 degrees
Analog 2:	-79.95dB to +0.0dB, 0.05dB steps, or OFF
Phase	0 or 180 degrees

(Analog 2 is only available in MONO mode)

Eight Band FM-HD Peak Limiter

A four band parametric EQ follows the eight band limiter and feeds the Bass Tools with multiple operating styles, then an oversampled, zero overshoot look ahead limiter.

Overall Controls	
Multiband Drive:	80 – 100%
Bass Tools Drive:	- 6.0/+12.0dB
Bass Tools Style:	Soft/Hard
Bass Tools Output:	- 12.0/0.0dB
Bass Tools Frequency:	60 - 300Hz
Multiband Knee:	Soft/Hard
Multiband Threshold:	+/-10dB relative to Drive
Multiband Attack:	0.1ms – 100ms
Multiband Release:	3ms – 300ms
Look Ahead Limiter Attack:	0.2ms – 100ms
Look Ahead Limiter Release:	33ms – 330ms
Look Ahead Limiter Delayed Release:	100ms – 1000ms (1Sec)

FM-HD

FM-HD Output Screen Controls

Phase	0 or 180 degrees
Left/Right Reverse checkbox	
Analog Out:	-79.9dB to 0.0dB, 0.1dB steps, or OFF
Digital Out:	-35.95dB to 0.0dB, 0.05dB steps, or OFF

Eight Band AM-HD Peak Limiter

A four band parametric EQ follows the eight band limiter and feeds the Bass Tools with multiple operating styles, then an oversampled, zero overshoot look ahead limiter.

Overall Controls	
Multiband Drive:	80 – 100%
Bass Tools Drive:	- 6.0/+12.0dB
Bass Tools Style:	Soft/Hard
Bass Tools Output:	- 12.0/0.0dB
Bass Tools Frequency:	60 - 300Hz
Multiband Knee:	Soft/Hard
Multiband Threshold:	+/-10dB relative to Drive
Multiband Attack:	0.1ms – 100ms
Multiband Release:	3ms – 300ms
Look Ahead Limiter Attack:	0.2ms – 100ms
Look Ahead Limiter Release:	33ms – 330ms
Look Ahead Limiter Delayed Release:	100ms – 1000ms (1Sec)

AM-HD

AM-HD Output Screen Controls

Bandwidth:	Off, and 16kHz to 4kHz in 1kHz steps
HF Protect:	16kHz to 3kHz in 1kHz steps
Ceiling:	0dB to -48dB in 0.5dB steps
HF Protect checkbox	
Hyper Mono checkbox	
Recovery 33mS to 330mS	
Phase	0 or 180 degrees
Left/Right Reverse checkbox	
Analog Out:	-79.9dB to 0.0dB, 0.1dB steps, or OFF
Digital Out:	-35.95dB to 0.0dB, 0.05dB steps, or OFF

Eight Band MP3/AAC>48K Peak Limiter

A four band parametric EQ follows the eight band limiter and feeds the Bass Tools with multiple operating styles, then an oversampled, zero overshoot look ahead limiter.

Overall Controls	
Multiband/Drive:	80 – 100%
Bass Tools Drive:	- 6.0/+12.0dB
Bass Tools Style:	Soft/Hard
Bass Tools Output:	- 12.0/0.0dB
Bass Tools Frequency:	60 - 300Hz
Multiband Knee:	Soft/Hard
Multiband Threshold:	+/-10dB relative to Drive
Multiband Attack:	0.1ms – 100ms
Multiband Release:	3ms – 300ms

MP3/ACC >48K

MP3/AAC>48K Screen Controls

Multiband Drive (clone from previous screen):	0 – 100%
Limiter Drive:	-18/+12dB
Look Ahead Limiter Attack:	0.2ms – 100ms
Look Ahead Limiter Release:	33ms – 330ms
Look Ahead Limiter Delayed Release:	100ms – 1000ms (1Sec)
Phase	0 or 180 degrees
Left/Right Reverse checkbox	
Analog Out:	-79.95dB to 0.0dB, 0.05dB steps, or OFF
Digital Out:	-35.95dB to 0.0dB, 0.05dB steps, or OFF

Eight Band MP3/AAC<48K Peak Limiter

A four band parametric EQ follows the eight band limiter and feeds the Bass Tools with multiple operating styles, then an oversampled, zero overshoot lookahead limiter.

Overall Controls	
Multiband Drive:	80 – 100%
Bass Tools Drive:	- 6.0/+12.0dB
Bass Tools Style:	Soft/Hard
Bass Tools Output:	- 12.0/0.0dB
Bass Tools Frequency:	60 - 300Hz
Multiband Knee:	Soft/Hard
Multiband Threshold:	+/-10dB relative to Drive
Multiband Attack:	0.1ms – 100ms
Multiband Release:	3ms – 300ms

MP3/ACC <48K

MP3/AAC<48K Screen Controls

Multiband Drive (clone from previous screen):	0 – 100%
Codec Sample Rate:	48/32/24K / 44.1/22.05K
Low Pass Filter:	OFF/16 to 4kHz, 1kHz steps
Codec Bit Rate:	64/48/32/24/20/18 kBPS
Mask Threshold:	0/-48dB, 0.5dB steps
Recovery:	33ms – 330ms
Limiter Drive:	-18/+12dB
Look Ahead Limiter Attack:	0.2ms – 100ms
Look Ahead Limiter Release:	33ms – 330ms
Look Ahead Limiter Delayed Release:	100ms – 1000ms (1Sec)
Phase	0 or 180 degrees
Left/Right Reverse checkbox	
Analog Out:	-79.95dB to 0.0dB, 0.05dB steps, or OFF
Digital Out:	-35.95dB to 0.0dB, 0.05dB steps, or OFF

Presets

An advanced preset management system allows the creation, storing, and recall of 80 presets within the VP-8IP itself and an unlimited number on a host PC. All signal-processing parameters are contained in these presets, allowing the preservation of the complete processing environment. Optionally, Input and Output settings may be stored and recalled with presets.

Note that all VP-8IP presets are mode-based. That is, presets are stored along with data that remembers what operating mode was running at the time they were created.

Preset Encryption

Presets are encrypted using the VP-8IP internal 128 bit serial number as the encryption key.

Onboard Real-time clock

A highly accurate internal real-time clock allows the automatic recall of presets at pre-determined times. This clock may be set to free-run or be synchronized to an external on- or off-site SNTP server, or it may by synchronized to the GUI PC.

Day-Parting and Long-term Scheduling

A weekly preset scheduling establishes a weekly cycle of preset changes, allowing for automatic programmed day-by-day exclusions or additions.

Additionally, a separate long-term scheduling table permits the establishment of one-off preset changes for *any* future time and date.

General Purpose Input (GPI)

Eight optically isolated inputs are slaved to the first eight preset slots.

Presets may be recalled by providing the appropriate voltage to the associated GPI port to create a Logic High and may be either momentary or latching as desired. Voltages are applied between the appropriate GPI pin and Common with the GPI pin being the most positive voltage. Please see manual text for the required current limiting resistor for voltages above 3.3VDC.

Ethernet Interface

The VP-8IP contains an auto-sensing 10/100BaseT Ethernet switch with one port available on the outside of the unit for field use. Because the switch is auto-sensing it is not necessary to utilize a crossover cable when connecting a PC directly to the VP-8IP.

Software Remote Control

Supplied Windows® VP-8IP GUI software affords control of all system and processing parameters via a 10/100BaseT Ethernet interface.