# VP-8

## DIGITAL AUDIO PROCESSOR

# TECHNICAL MANUAL

VORSIS®

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

**VP-8 Digital Audio Processor Technical Manual - 1st Edition - Revised**
**[VP8GuiSetup_2_3_x(and above).exe]**



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* a division of Wheatstone Corporation

# 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.***

# Making Audio Processing History

In 2005 Wheatstone returned to its roots in audio processing with the creation of Vorsis, a new division of the company. The same year it introduced the model AP-3 Digital Signal Processor at the National Association of Broadcasters (NAB) convention in Las Vegas.

At the following year's NAB convention, and with more than 100,000 people in attendance Vorsis introduced the model HD P3 Digital Multiband Signal Processor and the revolutionary model AP-1000 Digital Spectral Processor. The AP-1000 garnered two coveted industry recognitions: the Radio World "Cool Stuff" award and the BE Radio "Pick Hit". With its incredible dual 31-band FM and HD "Fine Grain" limiters and intuitive graphical user interface the AP-1000 set a completely new performance standard for *high-end* broadcast audio processors.

At the 2007 NAB Show Vorsis introduced the FM-5 and AM-5HD processors, as well as the new and hot-selling M1 Microphone Processor.

One year later at the 2008 NAB Show, Vorsis again surprised the broadcast industry with the introduction of *five* new audio processing products:

**AP-2000 and FM-2000**: successors to the well-received AP-1000 introduced just two years earlier, the AP-2000 is targeted for stations needing the power of its dual 31-band FM and HD limiters, while the FM-2000 has all the features of the AP-2000 except the HD signal path.

**AM-10HD and FM-10HD**: successors to the AM-5HD and extremely popular FM-5. The AM-10HD and FM-10HD are equipped with dual signal chains for separately processing the standard analog and HD signals.

**VP-8**: the first <u>*multimode*</u> broadcast audio processor with six built-in processing modes for FM, AM, FM-HD, AM-HD, MP3/-AAC>48k, and MP3/-AAC<48k. Audio modes are changed by simply selecting them from within the Windows-based GUI - no reboot is required to change from one processing mode to another.

These new Vorsis products are equipped with one or more of the following exclusive Vorsis-designed algorithms:

**- *Vorsis Bass Management System© (VBMS)***: Finally… clean, deep, low bass without the distortion that plagues other audio processors. VBMS manages bass by preventing frequencies that could "inter-modulate" in the main clipper from doing so, resulting in a dramatically deep low end - and clean midrange - and clean highs – simultaneously.

- ***Sweet Spot Technology*©** ***(SST)***: A completely *new* five band dynamics controller that is specially designed for maximizing the consistency of the station's on-air presentation.

- ***VoiceMaster*©:** For the loudest and cleanest on-air voices ever, VoiceMaster is a special vocal power spectrum management tool for detecting and then specially processing live announcer voice.

- ***Refined Main Peak Controller*:** It and its companion distortion control algorithm have been further improved for even greater on-air loudness and clarity than before while simultaneously reducing harshness and distortion.

# The Vorsis Vision

Wheatstone's Vorsis Audio Processing division was created for one purpose — to "design audio processing tools that *improve* the sound of broadcasting."

Company founder and owner Gary Snow noted how harsh and distorted radio stations had become over the years as they tried to increase their on-air loudness with existing products. In 2005 he set a goal to design audio processors that could *improve* the sound of radio, not further destroy it.

Today the Vorsis development team has grown to over a dozen experts in digital signal processing, professional audio, and product design, and is led by broadcast industry veteran and audio processing expert Jeff Keith. Jeff joined Wheatstone in January 2007 and guides the Vorsis team as they design new and revolutionary audio processing products in Wheatstone's design and manufacturing facility in New Bern, North Carolina.

One hundred percent of Vorsis design, manufacturing, testing, and quality control are accomplished within the state-of-the-art Wheatstone facility to ensure that customers always receive the very highest quality products.

Wheatstone's Vorsis audio processing division offers the broadcast industry superior performance in our line of competently designed and feature-rich audio processing products, built to Wheatstone's exacting quality standards. Vorsis audio processors are a perfect match for customers who truly *care* about how they sound.

# VP-8 Technical Manual

# Table of Contents

## Chapter 1 – General Information

## Chapter 2 - Vorsis VP-8 Features



## Chapter 3 - Vorsis VP-8 GUI

## Chapter 4 - Vorsis VP-8 Process Modes



FM

AM-HD



MP3/ACC >48K

# General Information

## Chapter Contents

**VP-8 Digital Audio Processor**


# General Information

## Introduction

The Vorsis VP-8 takes a completely new approach to broadcast audio processing through new and surgically accurate audio processing algorithms created by the Vorsis product design team. The result is far more audio processing power as well as complete user control over that power. There are no hidden controls and no secret "back door" hiding things from the end user. Every control is labeled for exactly what it does, and *everything* is brought out to the end user. If you can think of the on-air sound you want your station to have, the VP-8 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-8 has clarity, power, and accuracy of sound not found in any other broadcast audio processor in its price range.

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

Vorsis is a division of Wheatstone Corporation of New Bern, NC.

**Wheatstone Corporate Headquarters**

Wheatstone Corporation is a world-leader in broadcast audio equipment design and manufacturing. Our product line includes audio consoles for the radio and television broadcast industries, audio control surfaces and digital audio networks, and now, our ever-growing and award winning line of Vorsis digital audio processors.

Wheatstone began in 1975 as the "Audioarts" company. By 1976, its product line had already grown to include high quality audio equalizers, electronic crossovers, and compressor/limiters. In 1977 its first audio mixing control console was introduced. There followed many years of success, under the names of Audioarts Engineering and Wheatstone Corporation, as the leading manufacturer of broadcast quality audio consoles and distribution systems.

In 2005 Wheatstone returned to its roots when it created the Vorsis AP3 and began a line of specialized signal processing products. Today the product line consists of the AP-2000, FM-2000, FM-5, FM-10HD, AM-10HD, VP-8, HD P3, AP3, and M1 Digital Audio Processors for radio.

Wheatstone's product line has expanded greatly since the early days, and Wheatstone is now a recognized world leader in radio broadcast consoles, television consoles, and many types of audio processing equipment.

# Overview


The VP-8 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 Vorsis® Multiband Dynamics Processing.
- Precision eight-band multiband limiter.
- Specialized codec preconditioning tailored for each of the 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-8 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 an audio source that is attached to the VP-8 but not even on the air to be monitored!

The most salient feature of the VP-8 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-8 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-8 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-8 is designed to fit into an industry standard 19” equipment rack, and requires one rack unit (1.75 inches) of vertical space. The VP-8 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-8 because its power consumption is low enough to not require one.

The VP-8 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-8 and devices immediately above and/or below it.

***WARNING!* Under no circumstances should the VP-8 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-8 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-8 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-8 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 increased jitter.

## Where to Install the VP-8

The recommended location for the VP-8 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-8.

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-8 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-8.

The VP-8 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-8 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-8 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.

### EAS – United States Emergency Alert System

If the EAS generator is placed before the VP-8, the levels of the signaling tones and FSK data may be disturbed by the processing. Therefore it is highly recommend that the EAS unit be placed *after* the processing.

If the EAS unit must be installed prior to it, the VP-8 should be configured so that when the EAS tones are being sent the VP-8 is switched to a bypass mode. This can be accomplished by using the VP-8's GPI interface and by assigning a special preset to the EAS function that has internal processing defeated and audio levels calibrated to provide proper modulation levels during EAS transmissions.

## Energizing

Assuming the Vorsis VP-8 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-8; 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-8 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-8'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 Vorsis VP-8 GUI (see Chapter 3).

The two DB-9 connectors provide 8 GPI inputs for the first eight presets and 4 GPO outputs.

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

## Audio Inputs

Input audio can be applied to either the analog left/right inputs or to an AES3 digital input. The AES3 input automatically accepts sample rates between 32 kHz and 96 kHz.

Automatic audio failover from one input to the other is a user selectable option. Failover from AES3 to analog is instantaneous if based on invalid or missing bits in the AES3 stream. If the digital input is selected, Failover has been chosen, and no audio (<-48dBFS) is present at the AES input for 30 seconds, it will switch to the analog input.

Failover from analog to AES3 is based on the same silence sense algorithm responding to audio below -48dBFS for more than 30 seconds. Automatic switchover will only occur if Failover has been enabled.

### 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

### FM Path Analog Out—XLR-M

Output audio for FM path 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.

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

### FM Path AES Out—XLR-M

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

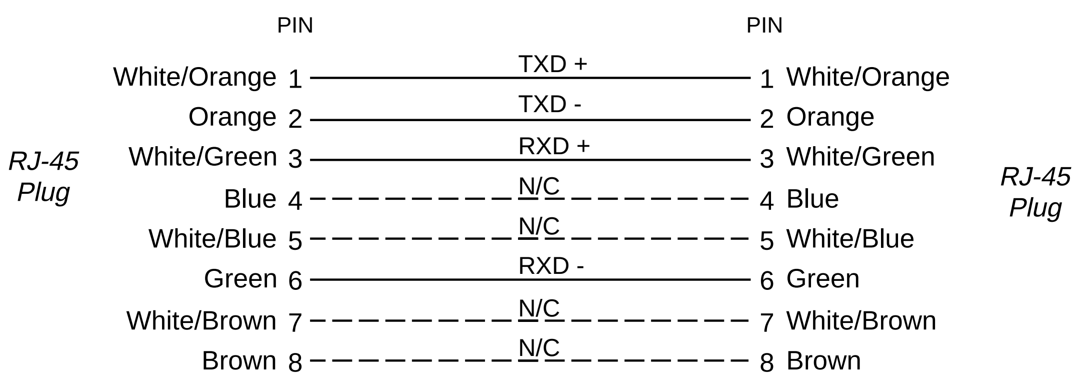
The VP-8 contains its own internal 100Base-T Ethernet switch with one port available on the outside of the product reserved for user connections. The Ethernet switch is equipped with auto-sensing and therefore this port will accept either a straight-through or crossover cable.

Networked systems are normally connected to the VP-8 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-8 with no network in between, it is normal to use a crossover cable.

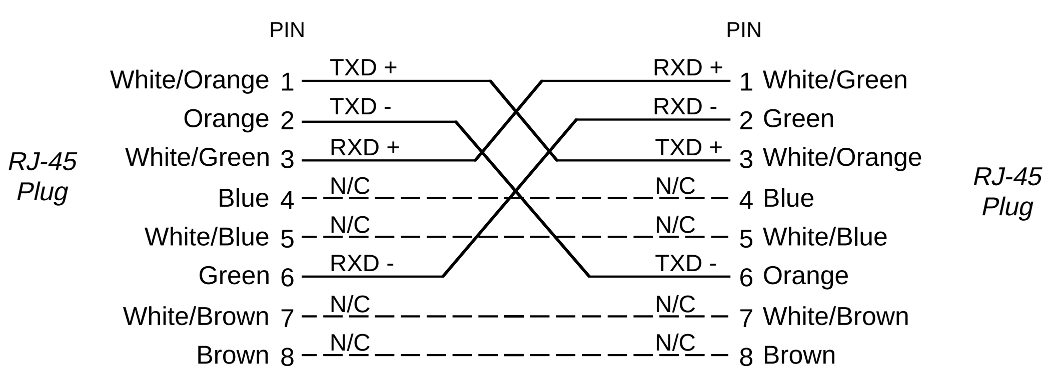
### Ethernet—RJ-45

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 CABLE**



## General Purpose Interface (GPI/GPO)

The VP-8 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-8.

### 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-8 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 (+)

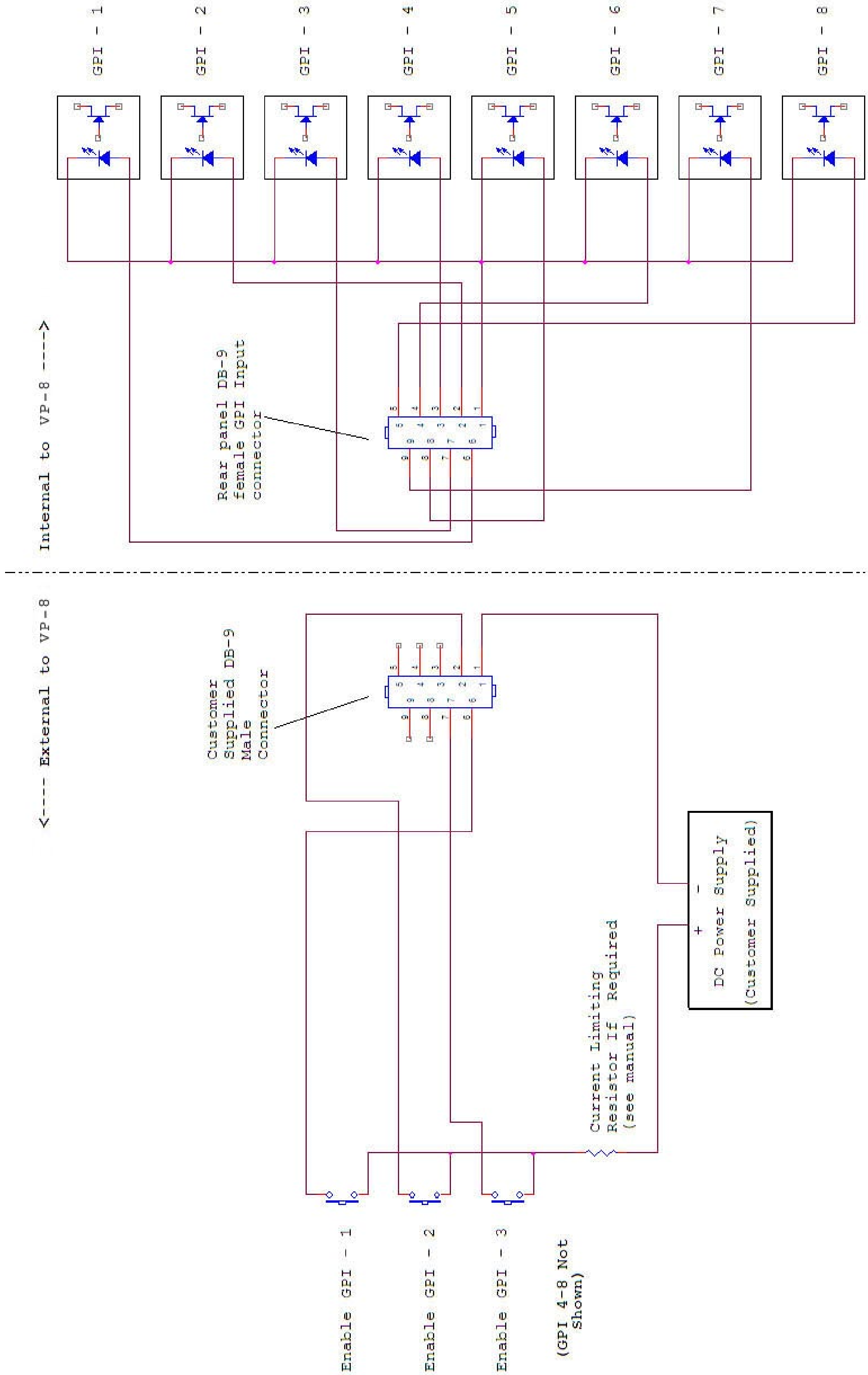
The GPI input circuits are optoisolated by devices with a 1,500VAC isolation rating.

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-8 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-8'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 <u>not</u> unlock factory presets.

## GP0

The VP-8 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 <u>isolated</u> return circuits — that is, they <u>do not</u> 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

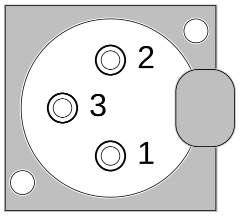
Pin 4 – GPO 3 Return
Pin 8 – GPO 3 Source

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

# Analog XLR Connections

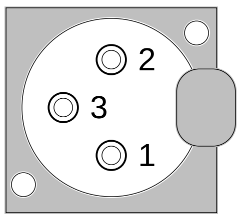
## Analog In - XLR-F

### 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

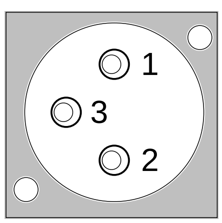
### XLR RT-F




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

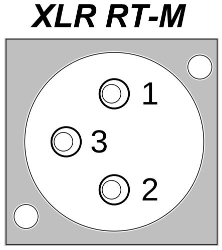
## Analog Out - XLR-M

### 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

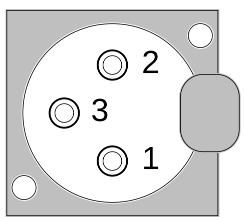
### XLR RT-M




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

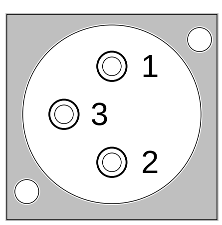
# Digital XLR Connections

## AES In - XLR-F




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

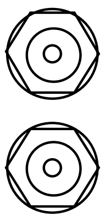
## AES Out - XLR-M




PIN 1 XLR SH - AES OUT SH
PIN 2 XLRHI - AES OUT HI
PIN 3 XLR LO - AES OUT LO

# BNC Connections

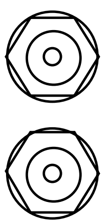
**BNC - SCA**



**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**

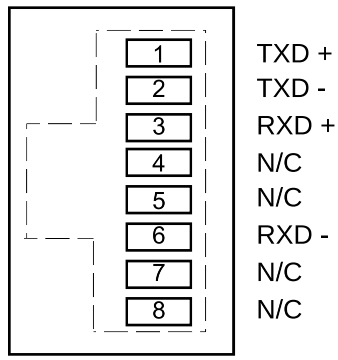


**1**
PIN 1 BNC 1 HI - TX 1 OUT HI - CENTER PIN
PIN 2 BNC 1 SH - TX 1 OUT SH - SHELL

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

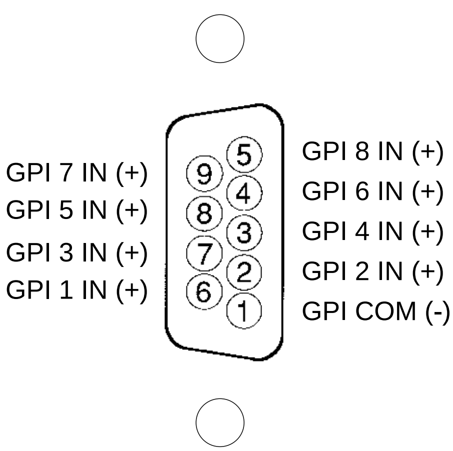
# Ethernet - RJ-45

### *RJ-45 ETH*

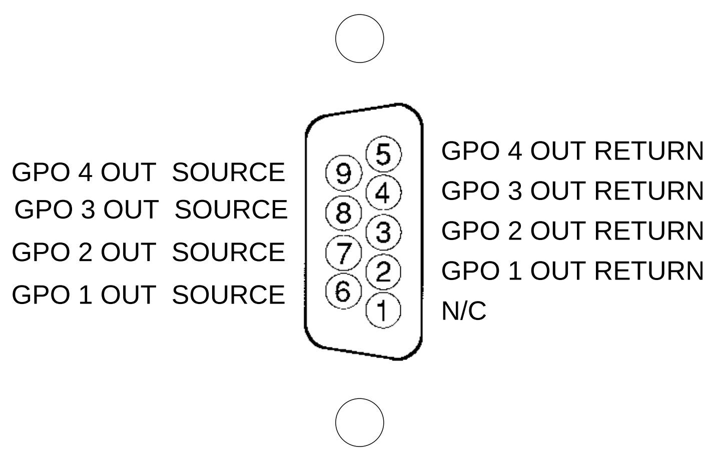



# General Purpose Interface - DB-9

### *GPI - DB-9*




### *GPO - DB-9*



# Vorsis VP-8 Features

## Chapter Contents

# Vorsis VP-8 Quick Start Setup Guide

We know that you're probably in a hurry to get on the air with your new VP-8. 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 the VP-8 Operating Guide 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:

| | |
|---|---|
| Vorsis VP-8 Unit | Quantity 1 |
| Vorsis GUI Software CD | Quantity 1 |
| AC Power Cord | Quantity 1 |
| Technical Manual | Quantity 1 |
| Vorsis Warranty Card | Quantity 1 |

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


**VP-8 Rear Panel Connections**

## Steps to Success

1. Install the VP-8 GUI 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-8 is 192.168.1.198. Refer to the Network Installation section for detailed information on reconfiguring the VP-8 IP address.
2. Install the VP-8 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 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.
4. Connect the Analog or Digital audio inputs as appropriate for your installation.
5. Start the VP-8 GUI, configure a device if you haven't done so already (see page 2-6), and then connect to the unit.
6. Click on the Input Menu tab in 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 VP-8 front panel Input and Output meters.
7. 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 at <u>no more than -12dBFS</u> with normal program levels being applied.
8. Make the audio output connections that are appropriate for your installation.
9. If using the Analog outputs pre-emphasized, navigate to the Stereo Encoder screen and select the checkbox 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 Stereo Encoder screen and select the checkbox for Analog L/R deemph (de-emphasized). Then adjust the TX 1 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-8, navigate to the FM OUT screen, and adjust the AES Output level appropriately.
12. If using the Composite outputs associated with the rear panel BNC connectors, navigate to the Stereo Encoder screen and select the checkbox 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 <u>composite stereo</u>, 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-8 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 0.01uF capacitor between XLR Pin #1 and XLR Pin #3.

We ship every Vorsis 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 Vorsis website for you to download and play with. Presets can easily be installed into your VP-8 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!

## Install and Connect the Vorsis VP-8

1. Install the VP-8 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.
2. Connect the audio input(s) to the appropriate XLR female input connector(s) on the rear panel.
3. Connect the line level analog outputs, the AES digital output, or the FM composite stereo output(s) as required in your installation.
4. To use the Vorsis VP-8 GUI software connect the rear panel Ethernet RJ-45 jack as appropriate for your intended use:
   a) For direct connection to a PC use a crossover CAT5 cable;
   b) For connecting into your LAN use a straight through CAT5 cable from the Ethernet switch or HUB.
5. Connect the AC power cord to the VP-8 rear panel AC power connector and then plug it into AC power. The VP-8 will power up and all DSP related signal paths should be intact within 20 seconds.

## Operating the VP-8 Locally

The front panel of the VP-8 has only two user controls, a rotary encoder with integral push switch, and a lighted push button located next to the rotary encoder. These controls are assigned to the following functions:


### Unlocking the Front Panel

The factory default passcode is “0000”. To unlock the front panel, press in and hold the encoder wheel until the “Pass 0000” prompt appears - the first two zeroes will be flashing. Press the encoder wheel to lock these digits in place and move the cursor to the second pair of digits. When the second pair of zeroes flashes, press the encoder wheel in. The message “OK” should appear. At this point the front panel is unlocked and will remain so for several minutes, after which automatic relocking will take place. You can also purposely lock it by pressing and holding the encoder wheel until the display reads “LOCKED”.


The front panel lock/unlock passcode can be changed using the remote control GUI software. Additionally, the front panel does not need to be unlocked in order to connect to the unit with the remote GUI software, or to adjust the headphone level.

### Taking Presets

If the front panel is unlocked, you can take a new preset. Simply turn the rotary encoder until the desired preset name is being 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 scrolling the new preset name, this time without flashing.

### 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.

## Operating the VP-8 Remotely—Installing the GUI Software

For remote operation via the GUI software supplied with the unit, insert the Vorsis VP-8 software CD into a Windows XP/2000 computer and follow the on screen instrucrions 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-8GUISetup_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-8 TCP/IP Address[*]

Once the GUI has been installed you must configure it and the VP-8 so that they can communicate with each other. This requires configuration for both the VP-8 (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-8’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:C2:23:xx:xx where xx:xx are the digits unique to your VP-8.

[*] *The VP-8 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-8 does not conflict with the address of any other device on the network. Please consult your friendly IT manager if necessary.*

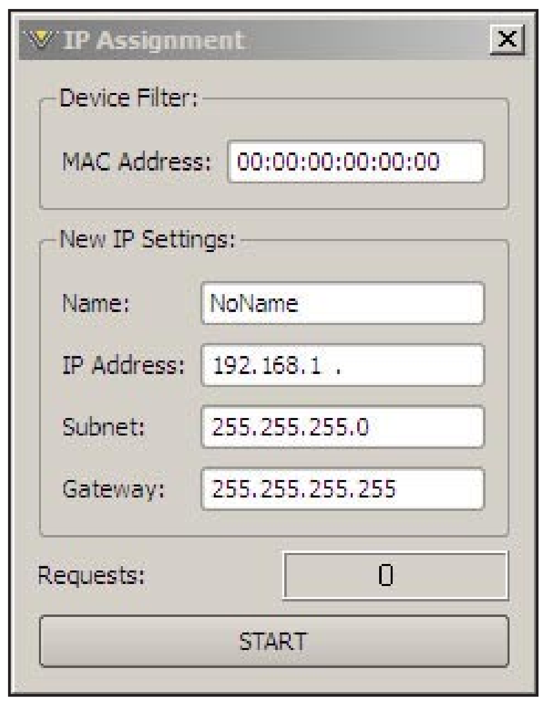



**Figure 1**
**VP-8 IP Address Configuration**

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

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

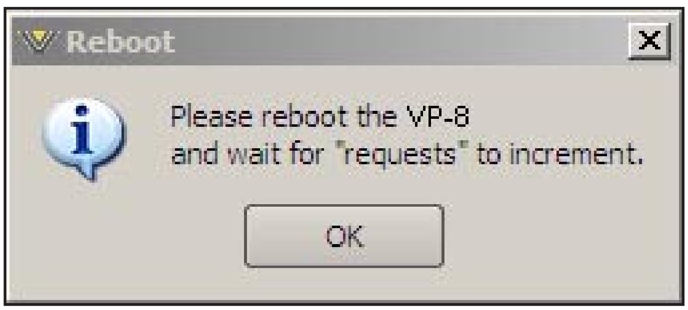
- Next, carefully enter your VP-8's MAC address in the MAC Address field. The MAC address must be carefully entered because the GUI will "send" the IP address information to the device that owns the MAC address that was entered in this step. If the MAC address is incorrect, the VP-8 will never "hear"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

- Next, enter a pet name for your VP-8, like "WABC VP8 2", etc.

- Then, enter the IP address that you wish the VP-8 to have, noting that this address must be unique if the VP-8 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:



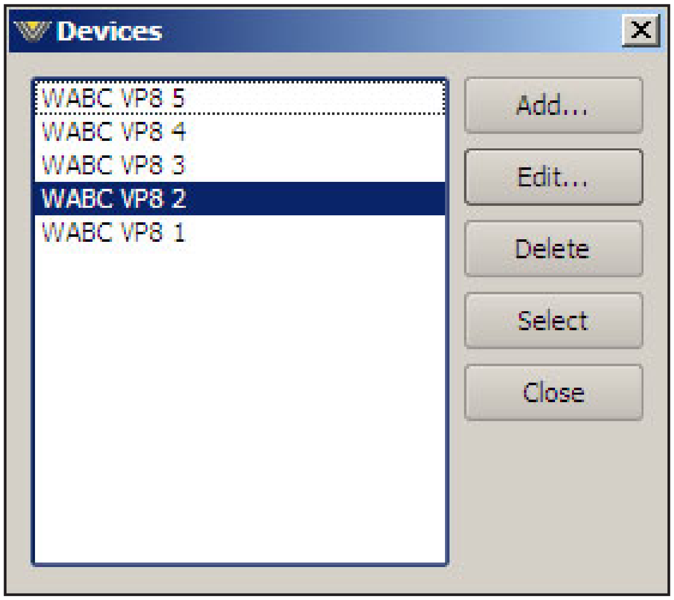

- Click OK, then remove power to the VP-8 for a few seconds, and then reapply it. As the VP-8 is booting up, the "Requests" number in the IP Address window should increment to something other than zero (please see the bottom of the image in Figure 1). When this occurs the VP-8 has been programmed with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

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

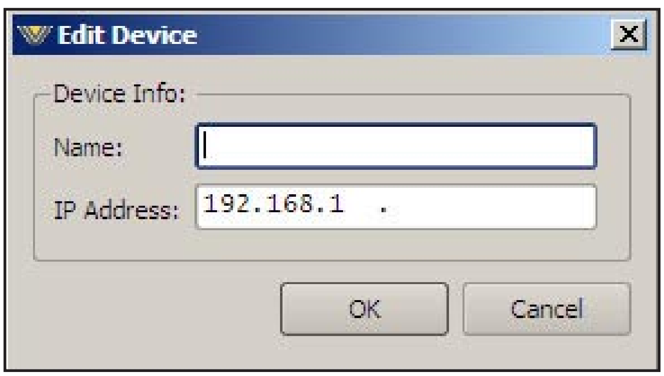
- Locate and click on the "Devices" button that is located along the right side of the GUI.


Location of "Devices" Button

- When this is done, 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 same pet name that you used when you assigned the IP address to the VP-8.

- Next enter the IP address that you configured the VP-8 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-8 device that you just added and click on "Select". This step tells the GUI specifically which VP-8 you want to connect to.

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

Online Status Indicator

Online/Offline Button


In the Status window you may see the message "Trying" as the GUI is handshaking with the VP-8. 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-8, 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-8 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-8 must be on the same network subnet. For instance, if the VP-8 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-8 and GUI PC cannot share the same IP address.

## Additional Notes on VP-8 GUI Connectivity

The Vorsis VP-8 uses both TCP port 55890 and the first available UDP port in the range of 60001 to 60010 during communications sessions.

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

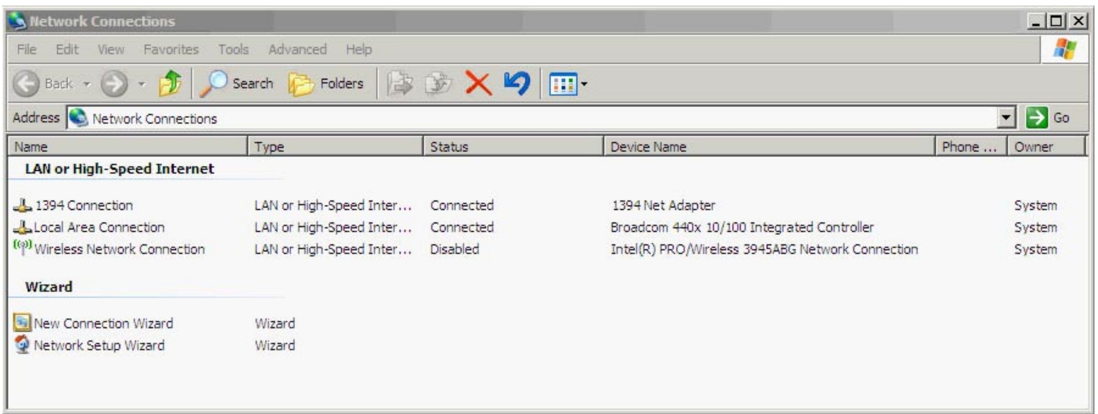
What this means is that network traffic on TCP port 55890 and the UDP port range 60001 - 60010 must be allowed on the portion of the network between the computer that the GUI is running on and the actual VP-8 hardware. If this is not true, then they will not be able to communicate with each other.

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.

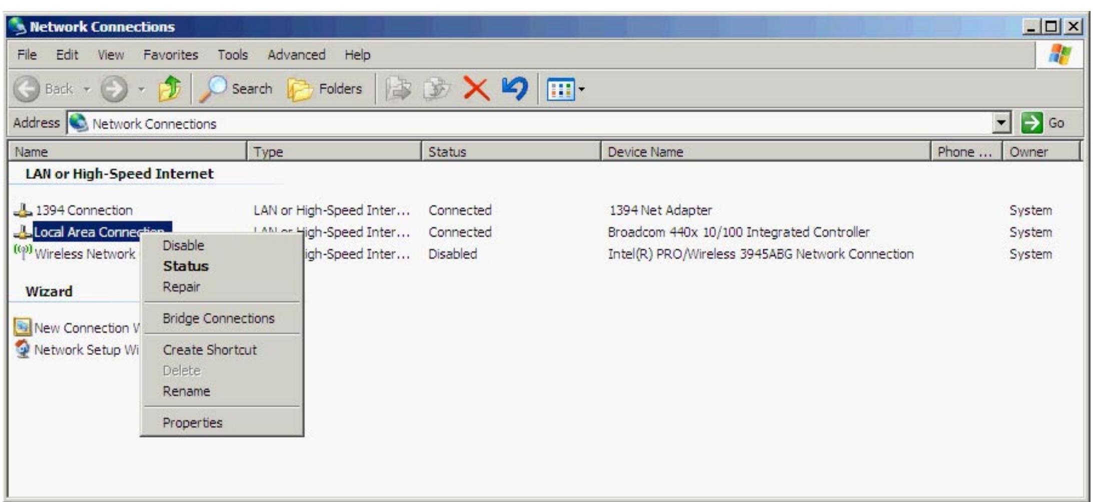
## A Known Windows Networking Issue

We've observed that certain builds of WindowsXP exhibit inconsistent behavior during either the IP address configuration process or later during attempts to connect to the VP-8 with the GUI. In virtually every case the cause has been identified to be old data being in the PC's ARP (Address Resolution Protocol) cache – the first place Windows looks for how to connect to a networked device.

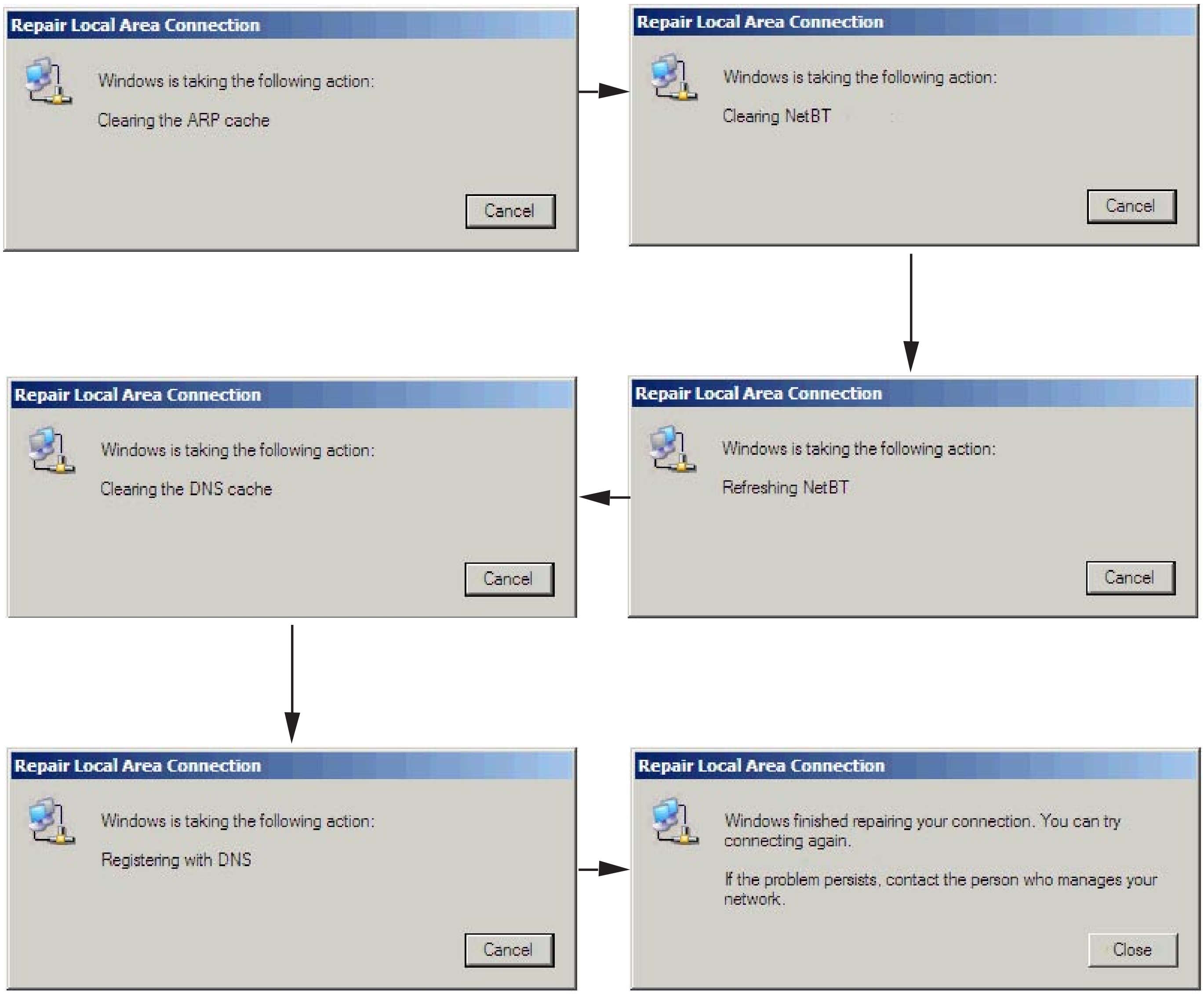
The ARP cache can be cleared manually at a command line, but it's much easier done using the built in tools within Windows. To do this, open the Windows Network Connections dialog:




Next, highlight the Local Area Connection and then right click it. A drop down menu similar to the following will appear:



One of the options in the drop down box is “Repair”. What this does is reset the PC’s Ethernet interface — in essence “refreshing” it. During this process the following six messages will go by:




This process usually completes in a few seconds and when it does the network connection should be fully functional. If it still does not work, then some friendly assistance from your local IT guru might be required.

# 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 Vorsis 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-8 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 <u>your</u> program format is <u>the</u> 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-8 contains our specialized Vorsis Bass Management System (VBMS).

VBMS 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-8 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-8 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 GUI, page 3-15); 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-8 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:

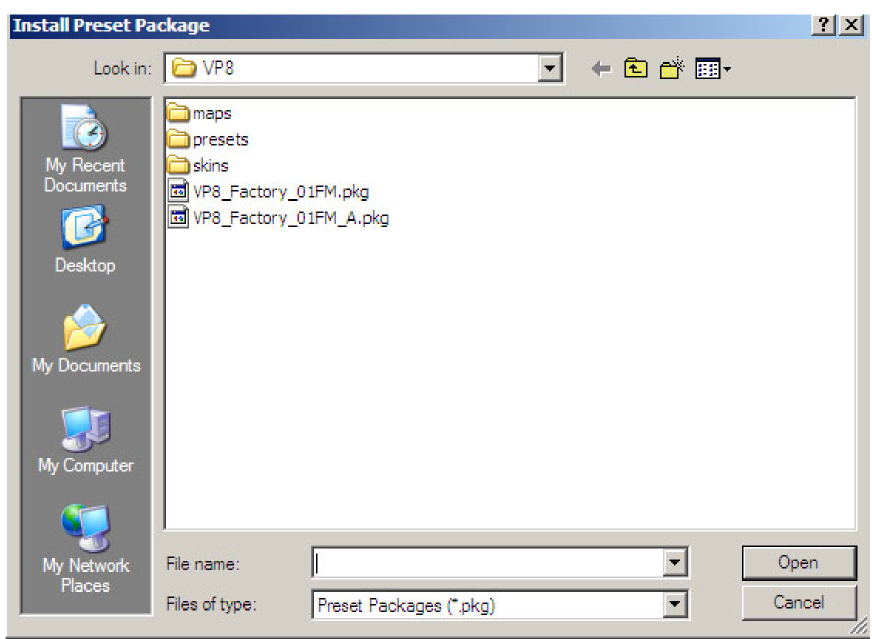
- Reduce the amount of reverb;
- Operate the AGC Backoff control at slightly less negative numbers;
- Reduce the overall drive to the AGC/Compressor section;
- Raise the Gate Threshold to less negative numbers and decrease the Gate Delay time to something under 100mSec.

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.

## Future Presets and Preset Updates

From time to time there will be new presets offered for download on the Vorsis website that were either created by the Vorsis development team or contributed by our loyal customers. These presets will be downloadable as "Preset Packages" which will have a ".pkg" file extension. Each preset package file will be named appropriately for the Vorsis product that they are compatible with.

Before downloading a preset package and installing it into your Vorsis audio processor, it is always recommended to consult the Vorsis website to ensure that the preset package that you wish to download is compatible with the current version of firmware in your Vorsis processor.



# Vorsis VP-8 GUI

## Chapter Contents

# Vorsis VP-8 GUI

## Architecture of the VP-8 GUI (Graphical User Interface)

The VP-8 Remote GUI 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-8 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.

## Network Configuration - Vorsis VP-8 GUI

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

## Getting Started

The VP-8 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 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 Vorsis Windows-based GUI's: "\Program Files\Vorsis\application folder name, where in this instance the application folder name is "VP8".

After installation, a "Vorsis VP8" icon will appear in the Programs list and clicking on it will start the GUI program.

*Note: Vorsis GUI programs DO NOT put entries in the Windows registry!*

To configure the VP-8 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-8's Windows-based GUI's could be a bit problematic.

# Connecting the VP-8 and the GUI

## Using a LAN Connection

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

The VP-8 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-8. Likewise, the 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-8's TCP/IP address from the factory default, the remote GUI can be used to accomplish this.

### Configuring the VP-8 IP Address*

Once the GUI has been installed you must configure it and the VP-8 so that they can communicate with each other. This requires configuration for both the VP-8 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-8'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:C2:23:xx:xx where xx:xx are the digits unique to your VP-8.

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

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


**Figure 1**

Next, carefully enter your VP-8'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-8 will never "hear" it.

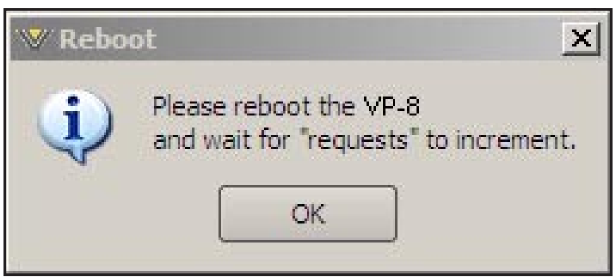
* *The VP-8 does not support DHCP (Dynamic Host Configuration Protocol) and therefore requires a static TCP/IP address on the network. Its presence on the network will not interfere with DHCP addressing of other network connected devices as long as the IP address that is configured for the VP-8 does 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-8, like "WABC VP8 2", etc.

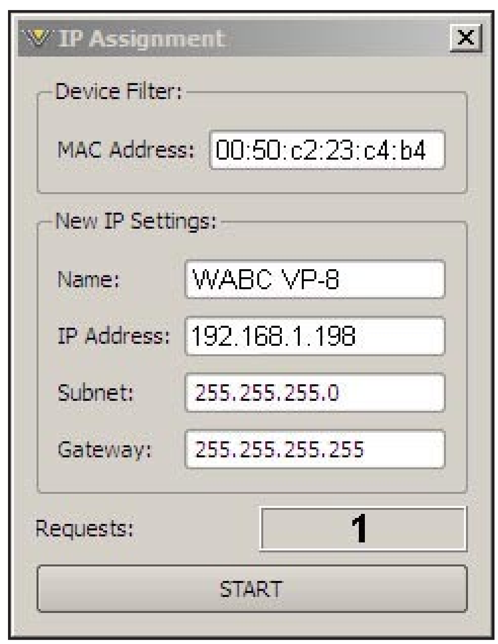
Then, enter the IP address that you wish the VP-8 to have, noting that this address must be unique if the VP-8 will be communicating over your network.

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-8 for a few seconds, and then reapply it. As the VP-8 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-8 has been programmed with its new IP address and is ready for use.

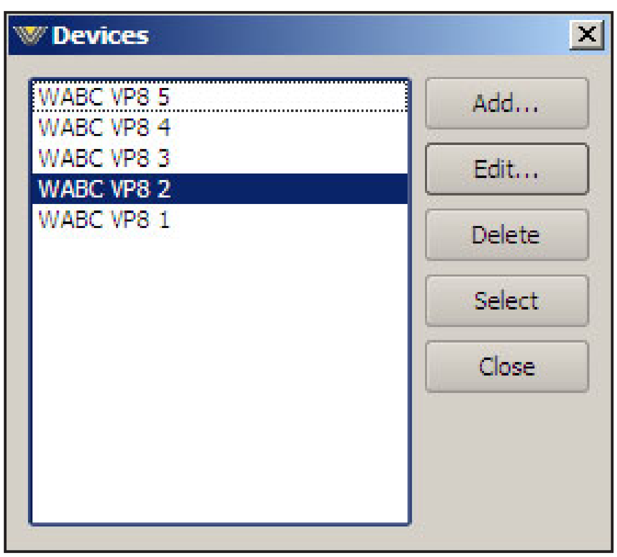


## Adding Devices

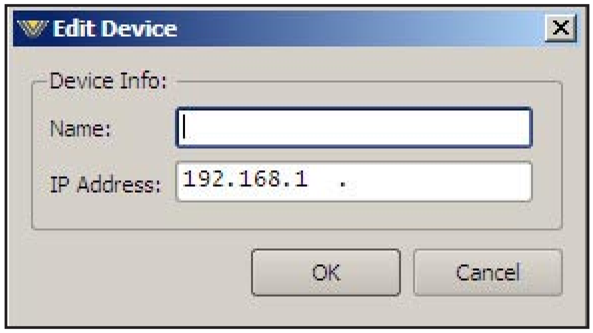
Now that the VP-8 has its own TCP/IP identity, we must configure the GUI so that it can talk to the VP-8. This is done by adding "devices" to the list of VP-8's that the GUI knows about. To do this, locate and click on the "Devices" button that is located along the right side of the GUI.


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-8.

Then enter the IP address that you configured the VP-8 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-8 device that you just added and click on "Select". This step tells the GUI specifically which VP-8 you want to connect to.

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

Online Status Indicator

Online/Offline Button


In the Status window you may see the message "Trying" as the GUI is handshaking with the VP-8. 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-8, 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-8 should be carefully examined to ensure that the destination TCP/IP address is consistent between the two.

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


**VP-8 GUI Status Bar**

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

## Connecting Directly Without a LAN

It is also possible to work without a LAN by connecting the VP-8 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-8 has auto-sensing ports. The controlling PC and the VP-8'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 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-8 while in a remote location such as their home, office, or car. This can be accomplished in a variety of ways.

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

We will leave the myriad other ways in which the VP-8 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-8 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-8 with VPN connections there is no technical reason why it would not work.

## The VP-8 and Internet Security Concerns

The VP-8 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-8 and use it as a pathway to the rest of the network to which it is connected.

## About DHCP and the VP-8

The VP-8 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-8

The VP-8 uses a variety of TCP and UDP ports in order to communicate with the remote GUI. The actual TCP port used depends on the Vorsis processor model. However, in each case the Vorsis processor is the "server" and the PC is the "client". Vorsis processors can accept up to 4 simultaneous GUI connections to host PC's.

The VP-8 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-8 to send meter data on that port. The port is typically 60001, 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 60001.

If that port cannot be opened, the GUI will then try port 60002, then 60003, etc., up to 60010. Whichever port successfully opens first, that port number will be sent to the VP-8 and the VP-8 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-8 then the VP-8 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-8 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 60001 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-8 and Host PC is blocking UDP traffic.

# Using the VP-8 GUI

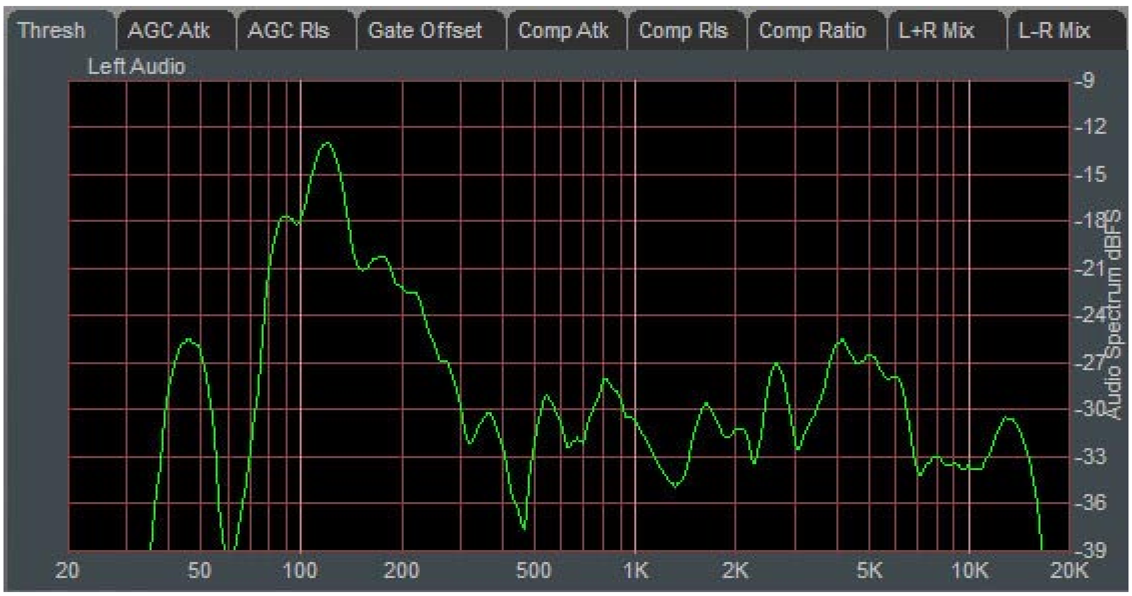
The GUI may be positioned on the host computer's screen by left-clicking-and-holding the "Vorsis" logo, and dragging the GUI to the desired position. In true "Windows" fashion, the GUI may be minimized on the desktop (taken off the screen) and/or closed by way of the familiar controls at the extreme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

Descriptions of the features and functions of the 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.



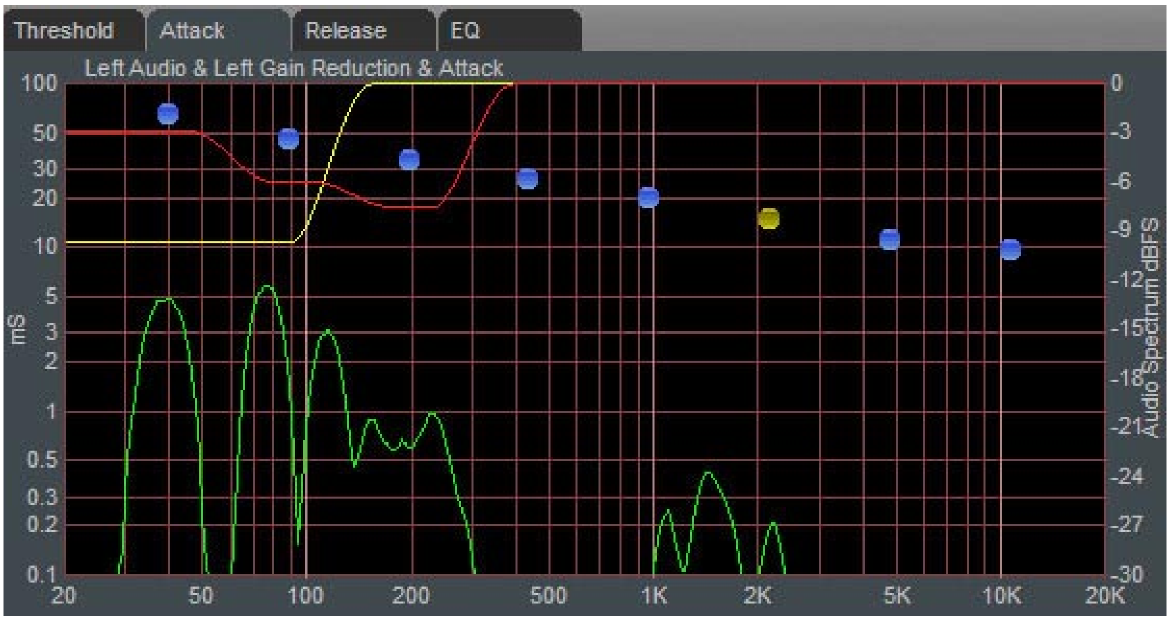

**VP-8 Audio Spectrum Display**

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-8 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.

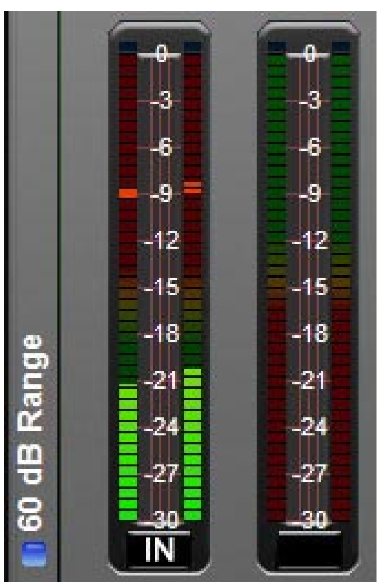
**Proportional Drag** – allows the dragging of dots to create slopes.

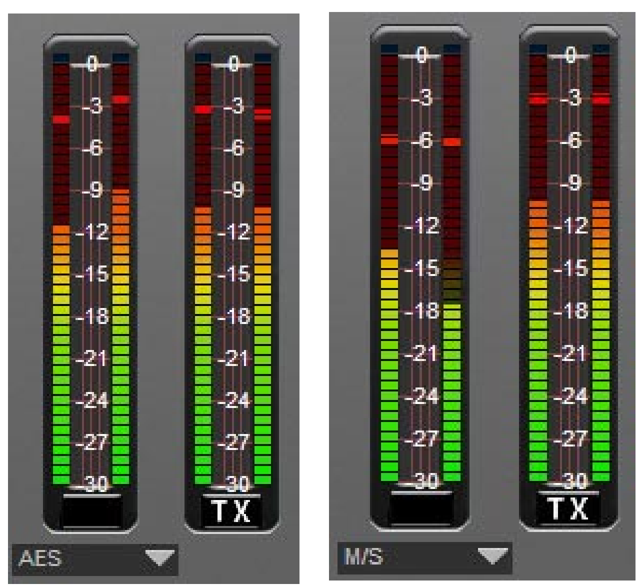
**Flat Drag** – allows the dragging of dots together in a “flat” orientation.

## 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 - Selectable - see below (page 3-17)
Right pair - TX Outputs

## Gain Reduction 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 / AGC/Compressor Output Meter (Right of Graph)

The dropdown list below the leftmost pair of the two vertical bargraph meters to the right of the graphical display window is used to select either the output level at the AES digital output (AES selection) or the output of the AGC/Compressor stage, which is always running in Sum and Difference Mode (M/S selection).

##  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-8, 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 checkboxes. 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 checkbox. 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.

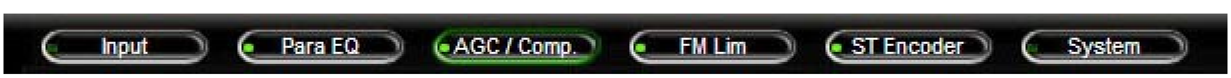

All of the knobs, faders, switches, and checkboxes affecting the audio processing itself are within this area. To operate them, either:

- 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 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 can be selected as either Analog, which is the analog Left/Right input, or AES, which is the AES3 digital audio input.

The currently selected input is shown by a red checkmark.

## Input Signal Presence

Below the source selection checkboxes 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.

## Input Failsafe

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

If the primary source was digital, the analog input will be selected immediately if the AES3 receiver chip detects invalid bits in the AES data stream or missing audio data. Also, the analog input will be selected after 30 seconds of a valid AES data stream having a signal level below -48dBFS.

If the primary source was analog, then a silence sense timeout of 30 seconds must elapse before the unit switches to the digital input in an attempt to restore audio. The audio failure sense threshold is fixed at -48dBFS, a value which is suitable for virtually all program types.

## 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 +/- 24dB 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. AES Gain can be adjusted over a +/- 24dB range.

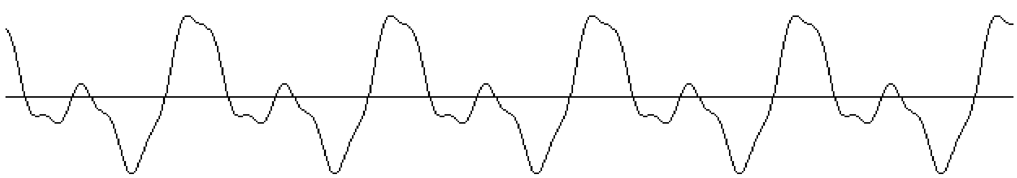
## Balance

Static level errors in Left/Right channel balance can be corrected by using the "Balance" knob. The Left/Right balance can be adjusted over a +/- 12dB range. The control is common to both the analog and AES digital inputs.

## Phase Rotator

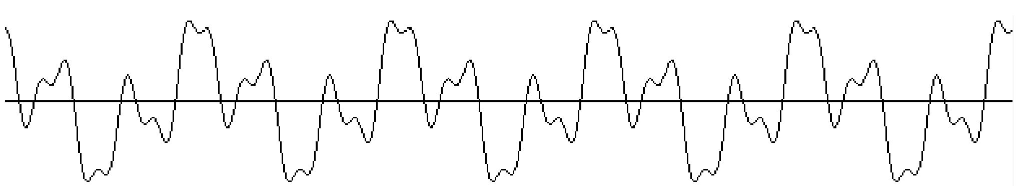
Human voice is asymmetrical in nature, which means that it is usually "peakier" in one polarity than the other. The algorithm fixes these asymmetrical peaks by phase shifting the signal harmonics relative to their fundamentals. This reduces the peak to average ratio and enables an increase in apparent 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-8 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 is also useful in removing subsonic energy that could upset the operation of equipment later in the audio chain, such as the AFC loop in an FM exciter.

The High Pass Filter has three operating modes:

**Off** – the High Pass Filter is not in operation.

**Stereo** – the High Pass Filter operating mode is stereo, and in this mode both left and right audio channels are treated equally by setting of the high pass filter.

**Sum/Diff** – the High Pass Filter is operating in Sum and Difference mode, which means that the sum (mono) information and difference (stereo) information may be processed by different values of high pass filter.

High Pass Filter Mode is selected as follows:

- If the Stereo checkbox is checked then the mode is Off if the High Pass Filter checkbox is not checked, or Stereo if High Pass Filter is checked.
- If the Stereo checkbox is not checked then the mode is Off if both High Pass Filter checkboxes are unchecked, or Sum/Diff if either (or both) High Pass Filter checkbox is checked.

**Why the Sum and Difference Option?**

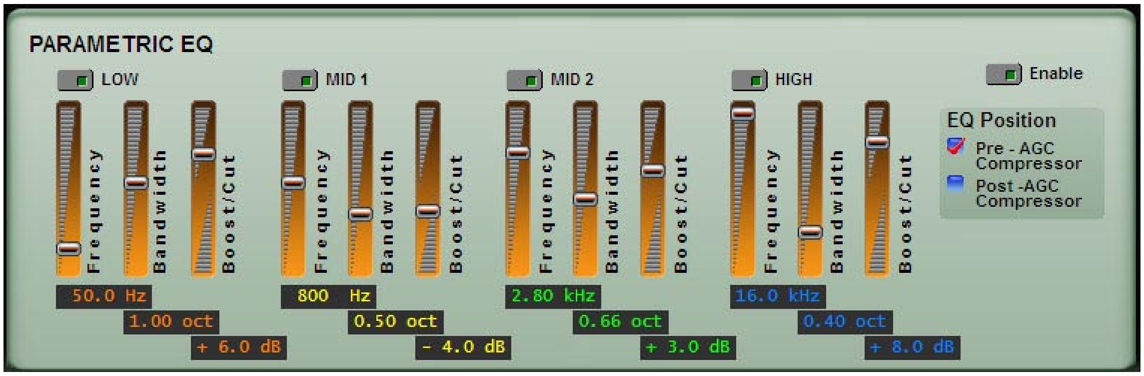
In most program material there is little very low frequency energy in the difference (L-R) signal. In fact, what energy is there is not typically correlated with the program material. Therefore the difference channel can operate with a much higher high pass filter cutoff frequency than the sum channel,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

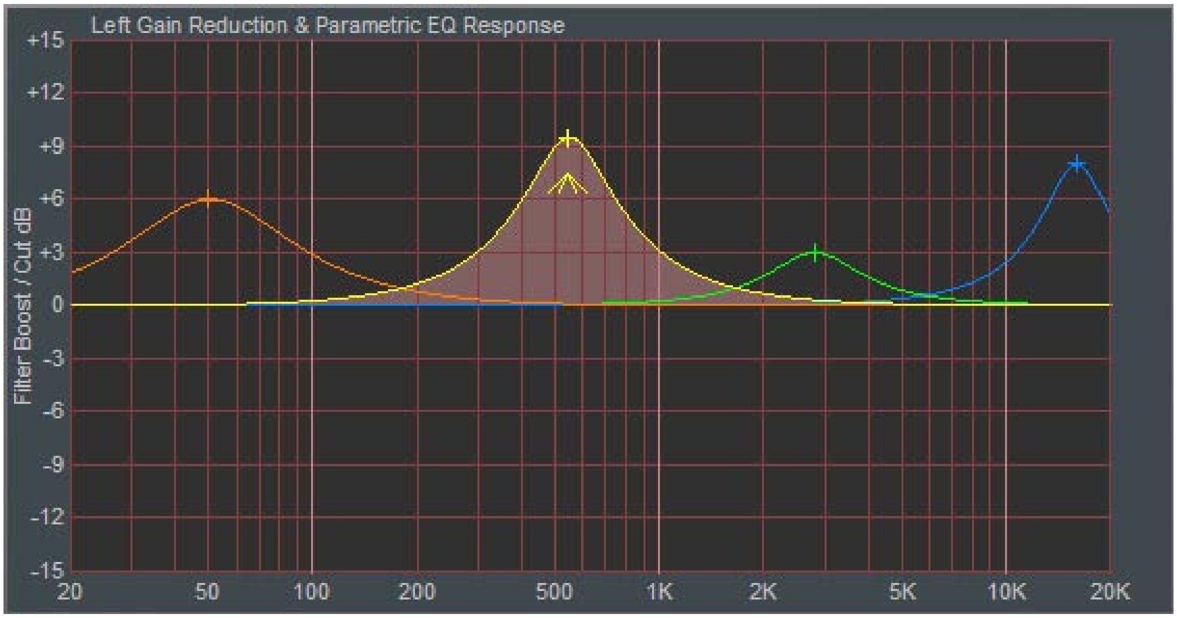
| | | |
|---|---|---|
| Center frequency | — | 20 Hz to 20 kHz. |
| Bandwidth | — | 0.2 to 3.0 octaves. |
| Boost/Cut | — | +/- 14.0dB. |

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.

## 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-8 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 and Compressor sections' 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 and 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.

### Drive

The 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.

**L-R** - The List box below the Drive control contains the trim adjustment for the Difference, or L-R channel, and may be adjusted +/- 6.0dB referenced to the Drive control. In other words, if the Drive control is set to +4.0dB and the L-R trim to -5.0dB, the difference channel drive is actually -1.0dB.

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

### Makeup

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 Makeup Gain control has such a wide range.

**L-R** - The list box below the Makeup control contains the trim adjustment for output of the difference, or L-R channel, and may be adjusted +/-6.0dB referenced to the Makeup setting.

Example: If the Makeup control is set to +30.0dB and L-R trim is set to -3.0dB, the difference channel is then scaled downward to +27.0dB, or 3dB below the sum channel.

## AGC/Comp Buttons

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

**AGC Button** - When illuminated by clicking on it, indicates that the output of the AGC section is in the audio path.

**Comp Button** - 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).

## Gate and AGC Backoff


**Gate and AGC Backoff Controls in the AGC Screen**

### Gate Threshold

The Gate 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 Hold and Ooze 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 Gate Threshold setting. This allows precise Gate thresholds to be assigned to each band as appropriate for the program format requirements.

### Gate Delay

Gate Delay is how long the gating circuit waits after the signal has fallen below the Gate Threshold before it actually modifies the release time according to the “Gated Mode” setting. The Gate 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 Gate 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 Gate threshold. This can be useful for maintaining a more natural drop in signal level during program fades.

### Hold

When “Hold” is selected and the audio falls below the Gate 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 Gate Threshold. This prevents a gain increase in the absence of audio which would likely increase background noise.

Note that the time between achieving the Gate threshold and the actual gain Hold is determined by the setting of the Gate Delay control.

### Ooze

When “Ooze” is selected and the audio input falls below the Gate 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 “Hold” setting may not be appropriate.

Note that the time between achieving the Gate threshold and the actual gain transition to Ooze is determined by the setting of the Gate 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.

## Band Coupling

The VP-8 utilizes inter-band 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 Band Coupling control, which ranges from -30.0dB to 0.0dB.

Example: Suppose that the Band 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 Band 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 Band Coupling control setting).

The Band 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.

With the 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 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 Backoff control allows the audio compression texture to be modified from a gentle, almost unobtrusive control (Backoff between “0” and about minus 3.0dB), to very obvious compression (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 crossover frequency is selected using the red list box. Frequencies between 50Hz and 198Hz may be selected.
- The Low Mid to High Mid crossover frequency is selected using the yellow list box. Frequencies between 250Hz and 1.50kHz may be selected.
- The High Mid to High band 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 range of crossover frequency settings is 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 <u>up</u> 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.

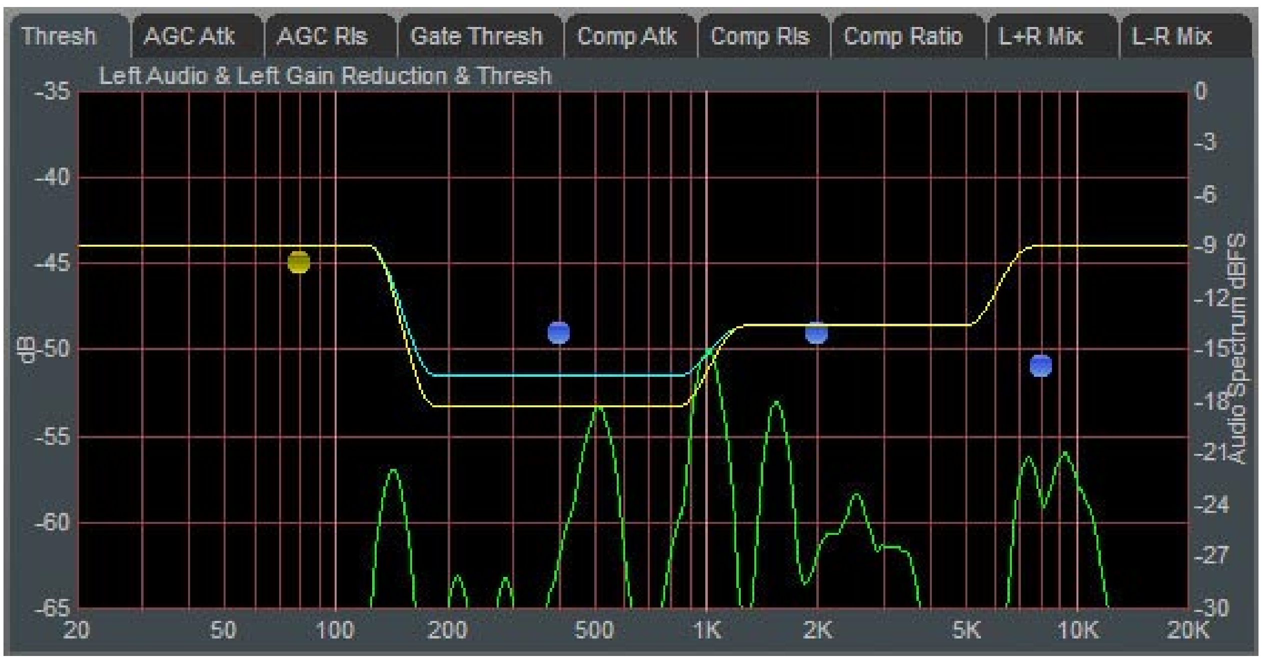
For more middle of the road music and for Classical and Jazz the Low to Low Mid crossover may be set between 120Hz and 150Hz, while the High Mid to High crossover may be set between 3.5Hz and 5.5Hz.

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 <u>very</u> general suggestions and your particular program format may sound better with other settings. Please feel free to experiment!

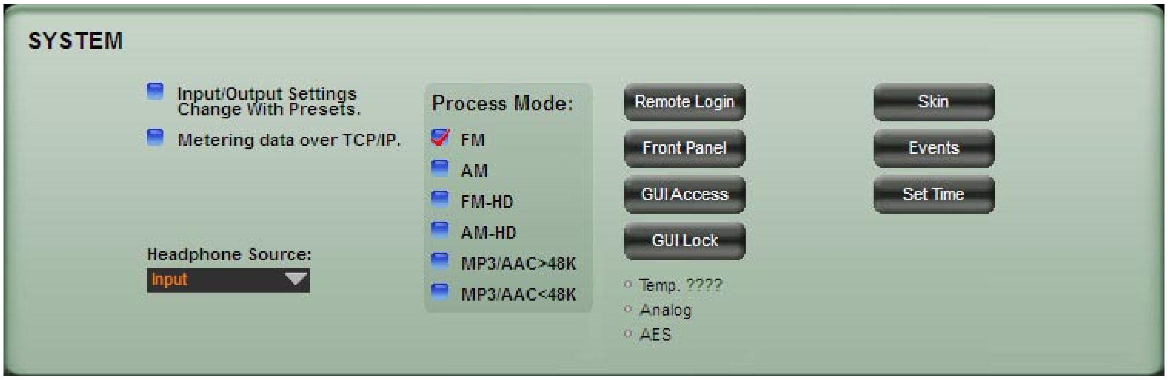
NOTE: Although the AGC and Compressor always operates in sum and difference mode, the crossover frequency settings track both signal chains, assuring a stable stereo image at all times.

## AGC/Compressor Gain Reduction




The curve above is broken into five segments representing the instantaneous gain control for each of the five 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 checkbox labeled:

### Input/Output Settings Change With Preset Takes

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-8 SYSTEM, or global, parameters, which are saved with presets but not restored when presets are "taken," unless the "Input/Output Settings Change With Preset Takes" checkbox 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 AES

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

**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

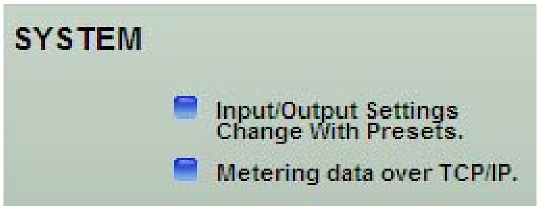
**From the MP3/ACC<48K Output Screen when in MP3/AAC<48k Mode:**

Digital Output

Analog Output

The option allows the VP-8 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 System Settings Change with Preset Takes checkbox is *unchecked*.

## Metering Data Over TCP/IP




The VP-8 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-8 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-8 and its GUI to using the TCP protocol for metering.

Switching the VP-8 to utilize TCP instead of UDP is done through the System menu. In the upper left corner of that screen is a checkbox “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.
**Line Input:** Audio appearing on the analog line input.
**AES Input:** Audio appearing on the AES3 digital input.
**Compressor:** Output of the multiband compressor.
**PreDelay:** Output of the FM limiter before the diversity delay.
**Final Output:** Final Output of the processing chain.

NOTE: The VP-8 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

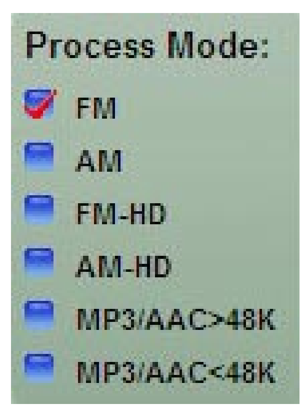
On the left side of the front panel of the VP-8 is a quarter inch female TRS jack for headphone use. The volume control for the headphone output is located on the right side of the front panel, and at the lower right of the remote GUI.

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.

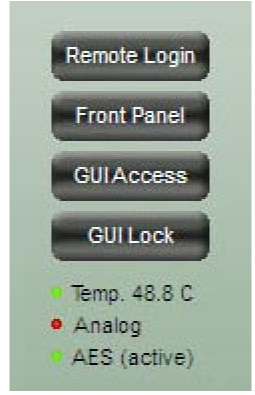
*Note: The headphone outputs of the VP-8 are driven by National Semiconductor LM675 power operational amplifiers capable of >3A into any reasonable load. Actual headphone impedance is not a concern; however headphone circuit current limiting is provided at 160mA in order to protect external devices and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

## Process Modes

The VP-8 has six built-in operating modes for FM, AM, FM-HD, AM-HD, MP3/AAC>48k, and MP3/AAC<48k, Changing operating modes is a simple as selecting as new mode from within the GUI; no reboot of the unit is required. For 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 GUI access to a VP-8 over Ethernet. When a GUI TCP connection session is initiated, the VP-8 accepts the connection and then asks the remote GUI for a password.

If the remote 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-8 will not send any of its current settings *to* the remote GUI nor accept any changes *from* the connecting GUI.

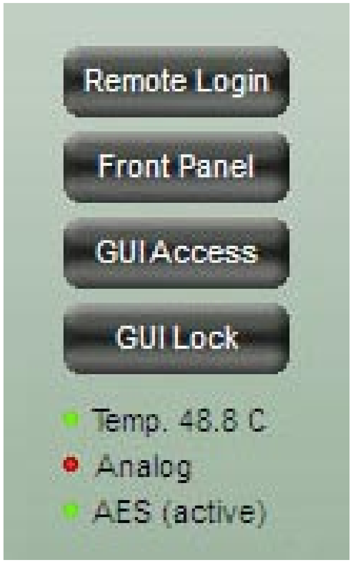
The dialog box allows you to change the login password if one has been set. You will need to know the current password beforehand, and also enter the new one twice for verification. The password can be any combination of characters, numbers, or symbols. If the "Remember this password" check box is checked, the 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.

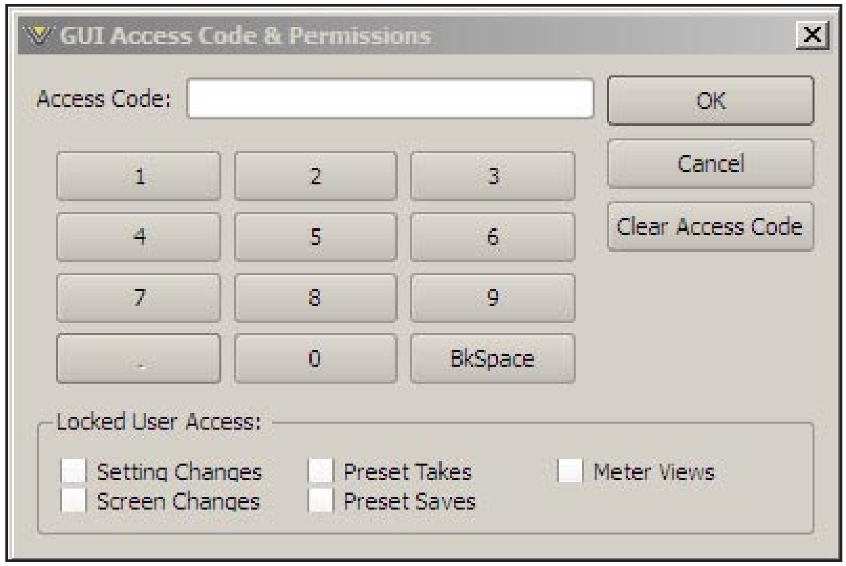
Occasionally you may see the word "pending" temporarily appear in the connection status box of the remote GUI. That means the VP-8 and Remote GUI are handshaking on the password string that is being sent.

## Front Panel

The Front Panel Password button allows the assignment of pass codes to the VP-8 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.

## 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-8. Privileges are granted by "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!

## GUI Lock

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

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

The following instructions must be followed exactly as presented!

1. Re-install the PC Remote GUI, but this time into a NEW folder on your computer. DO NOT start up the GUI yet!!

2. Make sure that the VP-8 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-8). DO NOT PROCEED if you do not have network connectivity to the VP-8!

3. Start the new remote GUI on your computer but DO NOT click on the Connect button at the top of the 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 GUI is open click on the Devices button on the right hand side of the 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-8 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 GUI.

11. Once the GUI is online click on the SYSTEM tab. The System screen will open.

12. Click on GUI Access button **(<u>DO NOT</u> CLICK ON GUI LOCK!!!)**. The GUI Access Code & Permissions box will open.

13. <u>Carefully</u> click <u>only</u> 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

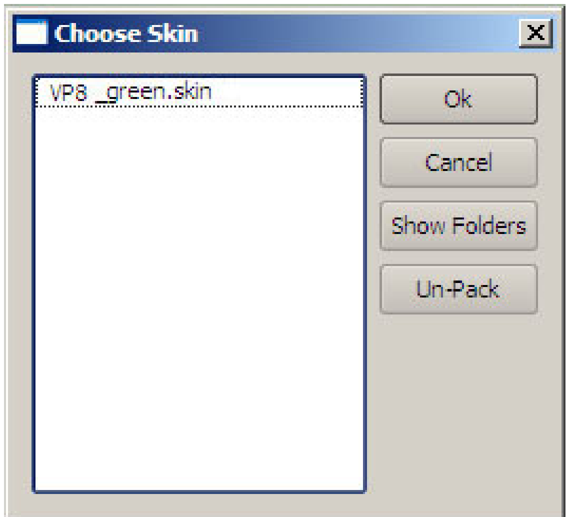
**Temp —** When the “Temp” indicator is green, the operating temperature inside the VP-8 is satisfactory. When it is red, the temperature is too high. The displayed value is the current operating temperature in degrees Celsius of the VP-8 motherboard. Normal temperatures should be in the range of 25ºC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

**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.

**AES —** 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.

## Skin

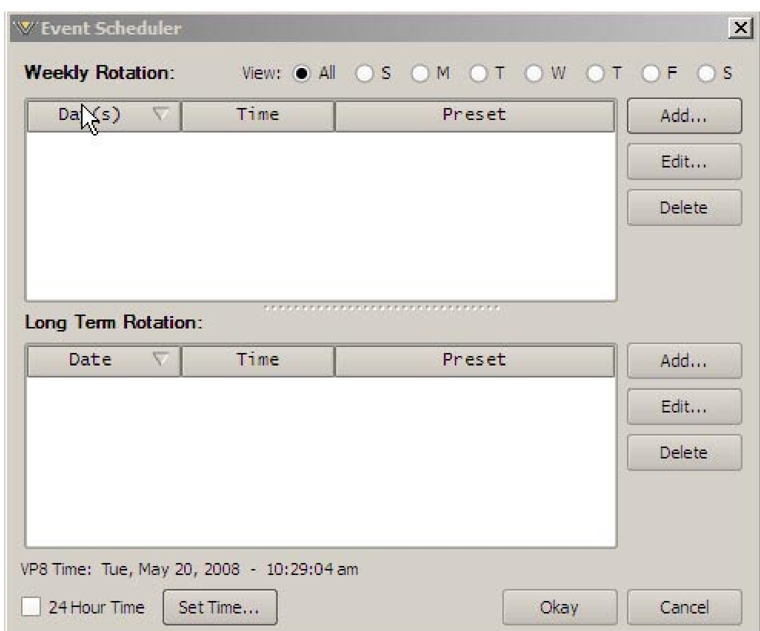

Allows the option to change the look of the GUI's skin. As new skins are released from Vorsis they will be available for download from the website.




## Events

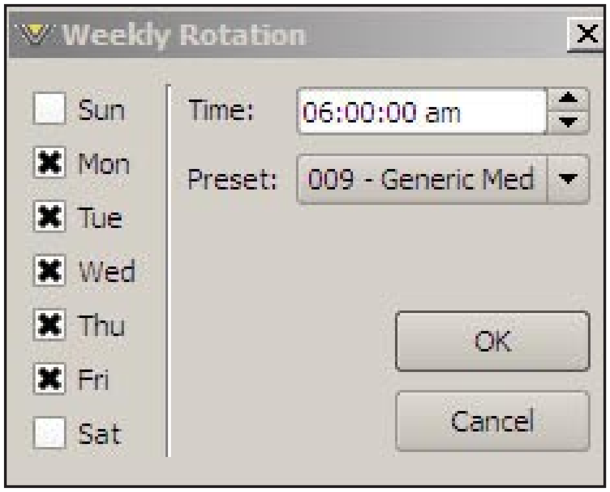
The VP-8 contains a comprehensive scheduling utility that allows presets to be automatically changed on desired dates and times. There is also a "Long Term Rotation" utility to allows one time preset changes to be scheduled for any time in the future. To create, edit, or view entries in the Event Scheduler 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 checkboxes may be chosen.

3. Next select the time that the preset should change.

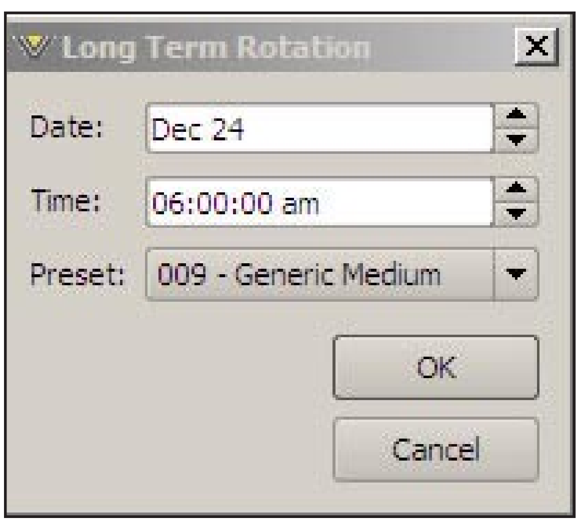
4. Next choose the preset that should be selected at the time programmed in the previous step. When finished editing, click OK.

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.

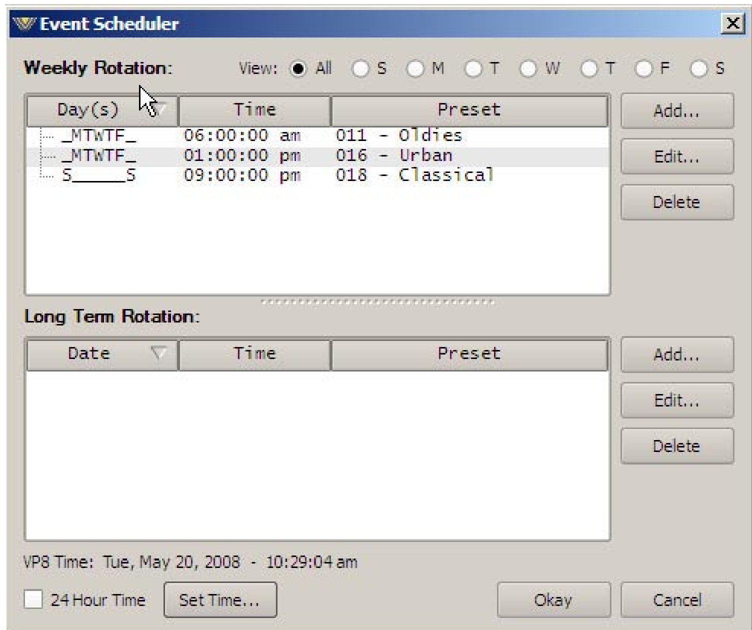
**Creating a Long Term Rotation Event**

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 Scheduler window by navigating to the System menu and clicking Events. The following window will open:



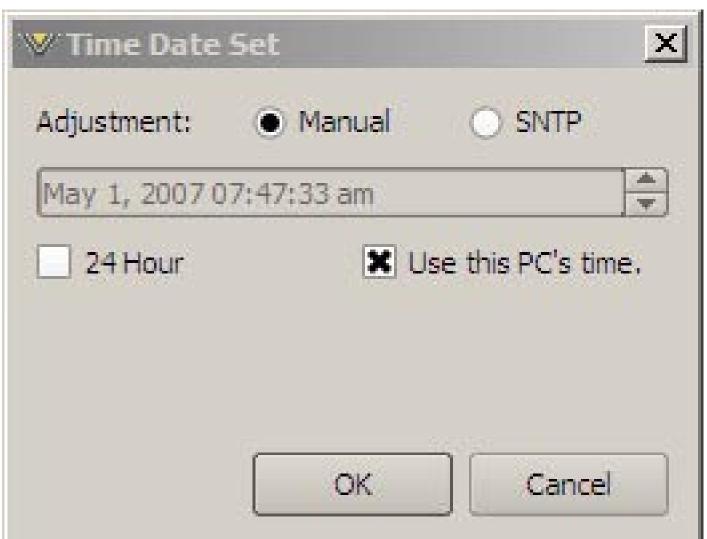

To delete a schedule in either the **Weekly** or **Long Term Rotation** schedules simply highlight it in the list and then click the appropriate **Delete** button on the right side of the Event Scheduler window. The highlighted scheduled event will be removed from the list.

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

## Set Time

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

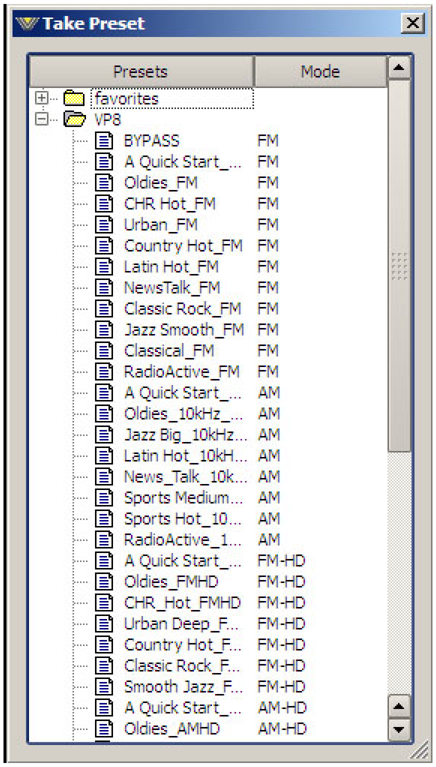
Alternately, time can be synchronized to the PC that is running the Windows-based GUI. In this instance time updates will occur upon successful connection to the GUI on the remote PC.



## Side Bar Region

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


### Preset




When left-clicked, a “Windows” 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 so) lists the presets in order. Double-clicking on a preset brings it immediately into use in the VP-8. 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-8 from this dialog box.

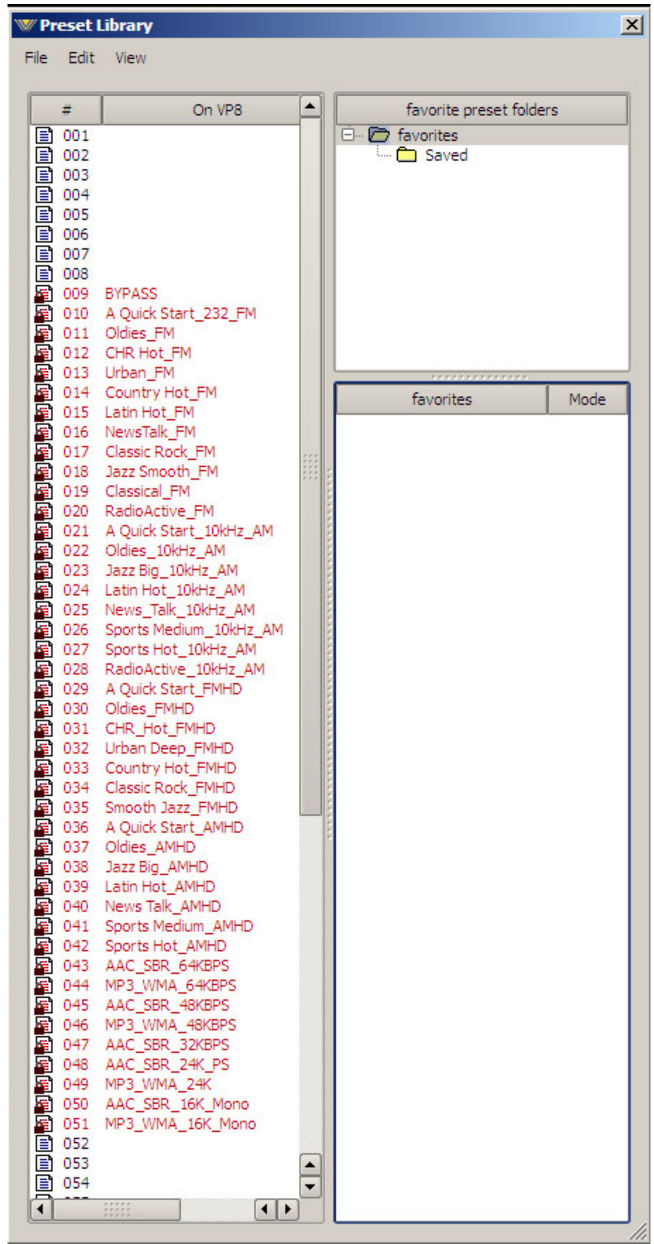
### Save

A small “Windows” dialog box appears, which prompts for a name under which to save the present settings of the VP-8. The preset will be saved into the next available empty slot. Alternatively, by nudging the preset number (“PRE #”) 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 "Windows" box appears. The large panel on the left has two columns, being the preset number and the VP-8 preset contents.

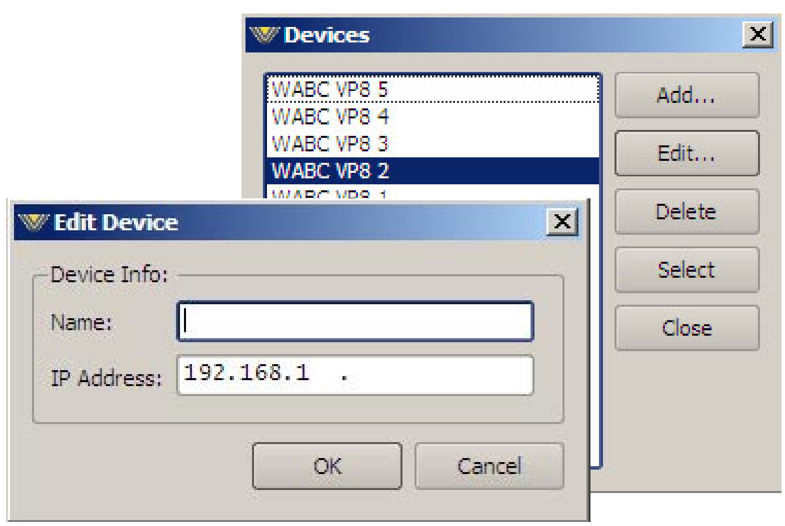
The upper and lower right hand panels give access to the GUI's presets library; the contents of folders in the upper panel are shown in the lower panel; drag-and-drop allows movement between the library folders and the VP-8 and its mirror archive in the GUI.



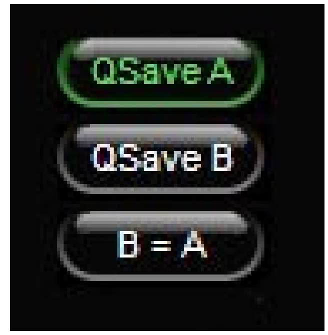

## Devices

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

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



## Quick Save (QSave)




The VP-8 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-8 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-8 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-8 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-8. 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-8 GUI screen (in line with the "Vorsis VP-8" product label to the left, and the Windows "About"/"Minimize"/"Exit" icons to the right) are indicators and controls for the management of devices and presets. The VP-8 GUI is capable of controlling multiple VP-8's ("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

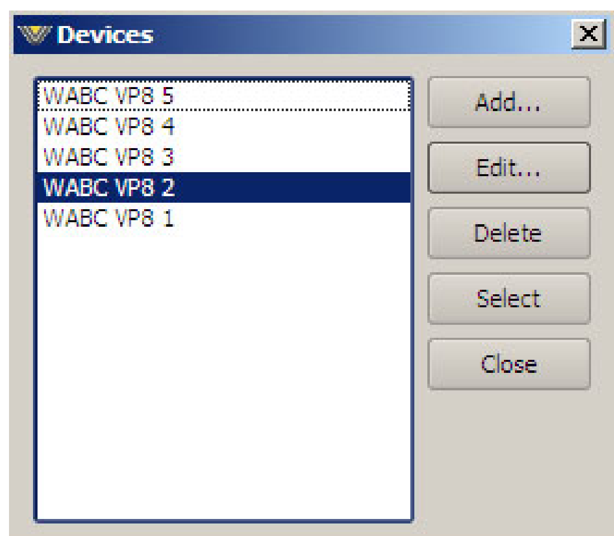
This indicates: "Online," meaning the GUI is in communication with and directly reflects a VP-8; "Offline," meaning the GUI and VP-8 are disconnected; and "Trying," meaning the GUI is attempting to find the VP-8 on the LAN and connect to it. An adjacent button toggles between "Online" and "Offline."

## Devices

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

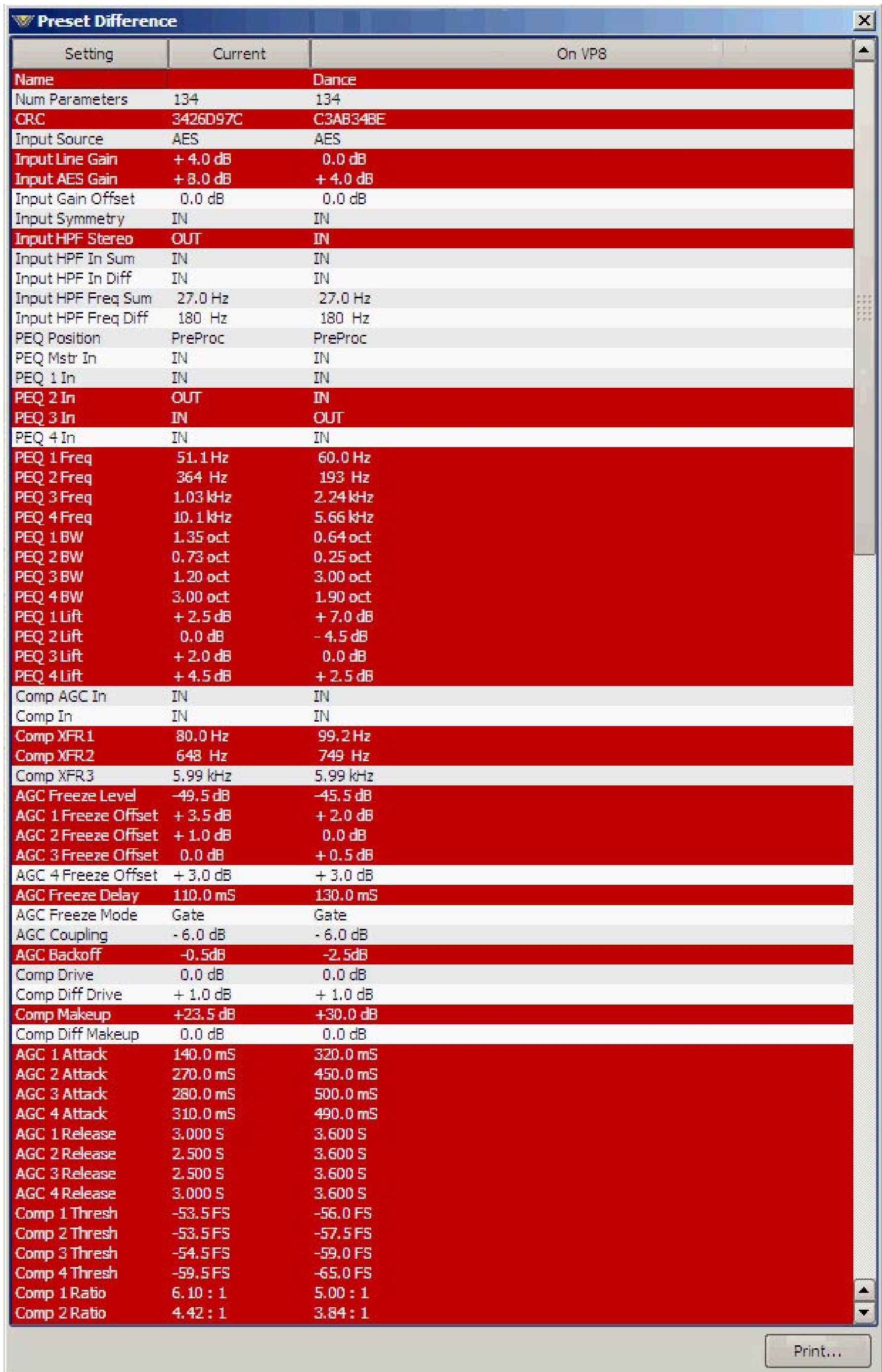
Double-clicking on the name area brings up the "Devices" Windows box. A list of VP-8 devices with which the GUI has had cause to become familiar is shown in a Windows box. Highlighting the desired VP-8's name and hitting "Select" causes the GUI to attempt to connect to it if you are Online.

In the event no VP-8's are listed, or one is adding a further device, or one for the first time, "Add" brings up a small 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-8's settings for this preset agree with those in the GUI's mirror archive; red indicates a discrepancy. Double-clicking on the name brings up a "Differences" — parameter-by-parameter, setting-by-setting — comparison list between the current VP-8 settings and the recalled preset.



Preset Difference

| Setting | Current | On VP8 |
|---|---|---|
| Name | | Dance |
| Num Parameters | 134 | 134 |
| CRC | 3426D97C | C3AB34BE |
| Input Source | AES | AES |
| Input Line Gain | + 4.0 dB | 0.0 dB |
| Input AES Gain | + 8.0 dB | + 4.0 dB |
| Input Gain Offset | 0.0 dB | 0.0 dB |
| Input Symmetry | IN | IN |
| Input HPF Stereo | OUT | IN |
| Input HPF In Sum | IN | IN |
| Input HPF In Diff | IN | IN |
| Input HPF Freq Sum | 27.0 Hz | 27.0 Hz |
| Input HPF Freq Diff | 180 Hz | 180 Hz |
| PEQ Position | PreProc | PreProc |
| PEQ Mstr In | IN | IN |
| PEQ 1 In | IN | IN |
| PEQ 2 In | OUT | IN |
| PEQ 3 In | IN | OUT |
| PEQ 4 In | IN | IN |
| PEQ 1 Freq | 51.1 Hz | 60.0 Hz |
| PEQ 2 Freq | 364 Hz | 193 Hz |
| PEQ 3 Freq | 1.03 kHz | 2.24 kHz |
| PEQ 4 Freq | 10.1 kHz | 5.66 kHz |
| PEQ 1 BW | 1.35 oct | 0.64 oct |
| PEQ 2 BW | 0.73 oct | 0.25 oct |
| PEQ 3 BW | 1.20 oct | 3.00 oct |
| PEQ 4 BW | 3.00 oct | 1.90 oct |
| PEQ 1 Lift | + 2.5 dB | + 7.0 dB |
| PEQ 2 Lift | 0.0 dB | - 4.5 dB |
| PEQ 3 Lift | + 2.0 dB | 0.0 dB |
| PEQ 4 Lift | + 4.5 dB | + 2.5 dB |
| Comp AGC In | IN | IN |
| Comp In | IN | IN |
| Comp XFR1 | 80.0 Hz | 99.2 Hz |
| Comp XFR2 | 648 Hz | 749 Hz |
| Comp XFR3 | 5.99 kHz | 5.99 kHz |
| AGC Freeze Level | -49.5 dB | -45.5 dB |
| AGC 1 Freeze Offset | + 3.5 dB | + 2.0 dB |
| AGC 2 Freeze Offset | + 1.0 dB | 0.0 dB |
| AGC 3 Freeze Offset | 0.0 dB | + 0.5 dB |
| AGC 4 Freeze Offset | + 3.0 dB | + 3.0 dB |
| AGC Freeze Delay | 110.0 mS | 130.0 mS |
| AGC Freeze Mode | Gate | Gate |
| AGC Coupling | - 6.0 dB | - 6.0 dB |
| AGC Backoff | -0.5dB | -2.5dB |
| Comp Drive | 0.0 dB | 0.0 dB |
| Comp Diff Drive | + 1.0 dB | + 1.0 dB |
| Comp Makeup | +23.5 dB | +30.0 dB |
| Comp Diff Makeup | 0.0 dB | 0.0 dB |
| AGC 1 Attack | 140.0 mS | 320.0 mS |
| AGC 2 Attack | 270.0 mS | 450.0 mS |
| AGC 3 Attack | 280.0 mS | 500.0 mS |
| AGC 4 Attack | 310.0 mS | 490.0 mS |
| AGC 1 Release | 3.000 S | 3.600 S |
| AGC 2 Release | 2.500 S | 3.600 S |
| AGC 3 Release | 2.500 S | 3.600 S |
| AGC 4 Release | 3.000 S | 3.600 S |
| Comp 1 Thresh | -53.5 FS | -56.0 FS |
| Comp 2 Thresh | -53.5 FS | -57.5 FS |
| Comp 3 Thresh | -54.5 FS | -59.0 FS |
| Comp 4 Thresh | -59.5 FS | -65.0 FS |
| Comp 1 Ratio | 6.10 : 1 | 5.00 : 1 |
| Comp 2 Ratio | 4.42 : 1 | 3.84 : 1 |

Print...

## Notes on “Online” and “Offline” Working

Most often operationally it will be required that the GUI act directly and instantaneously upon the VP-8 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-8 itself, which can continue to be processing away obliviously. (Preset list manipulation can be undertaken “Online,” too).

Importantly, any adjustments made on the GUI “Offline” will not take effect until the VP-8 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 GUI is in “Online” or “Offline” mode:

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

# Accessing Menu Options

Right clicking anywhere on the Vorsis VP-8 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 Vorsis VP-8 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-8 Control Panel.

*About...* — brings up the About box to indicate the GUI version.

*Center Window* — centers the main GUI window on the screen.

*Choose Skin...* — brings up the Choose Skin dialog box.

*Exit* - exits the GUI program.

## Hardware Menu Items

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

*Devices...* — opens the Devices dialog box. Allows the creation, editing, selection, and deleting of Vorsis 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.

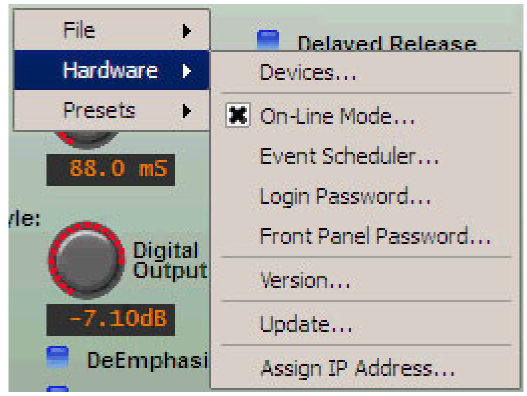
*Version...* — displays the current software versions running in your VP-8 hardware.

*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-8.

Future versions of the VP-8 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-8. Upon completion, you will be asked to restart the VP-8 (cycle power).

Only verified updates provided by Vorsis will work!

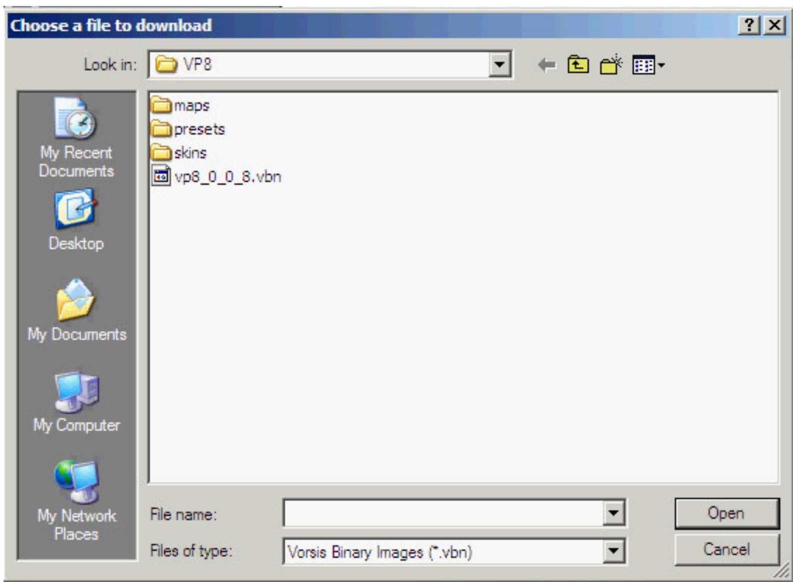



## VP-8 Hardware Update

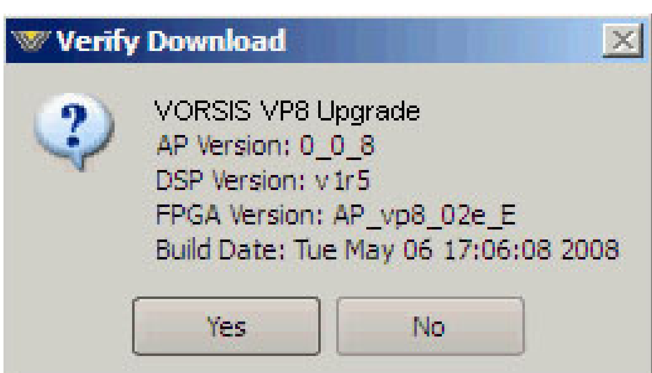
The VP-8 hardware is extremely easy to update whenever new firmware becomes available from Vorsis. There is only one file required for hardware updates, a file with the extension “VBN”.

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

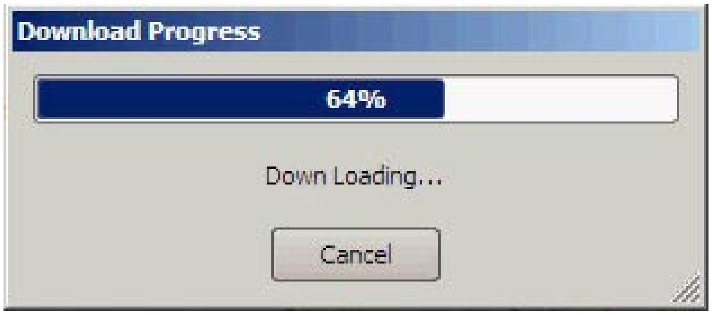
To update the VP-8, right click on any portion of the upper control area in any screen of the remote GUI to open the popup menu tree, and select the “Hardware -> Update...” option. The following screen will appear.

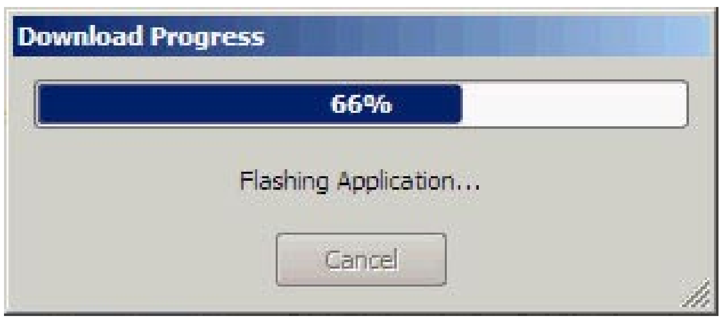



Navigate to the location of the VBN file that was downloaded from the Vorsis website or located on the CDROM that you received from Vorsis Technical Support. Verify that the version number is correct in that it matches the version number listed in the documentation about the firmware upgrade. If so, select the “Open” option in the “Choose a file to download” window. The following prompt will appear (actual text will vary):

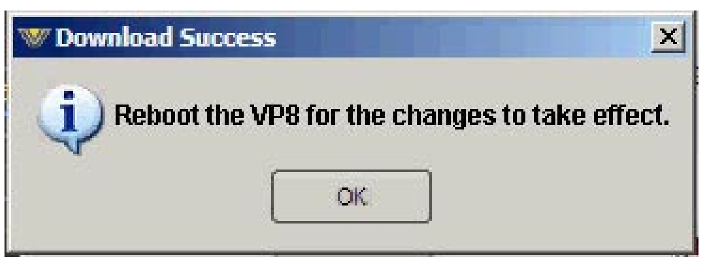


Check once again that the versions match the documentation! Then click “Yes”. Dialog boxes similar to the following will pop up as the download progress occurs:




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




When the VP-8 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-8 Control Panel.

*Preset...* - brings up the Take Preset dialog box.

*Save* - brings up the Save as Presets dialog box.

*Library...* - brings up the Preset Library dialog box.

# Vorsis VP-8 Process Modes

## Chapter Contents



FM

FM

AM

AM

FM-HD

FM-HD

AM-HD

MP3/ACC >48K

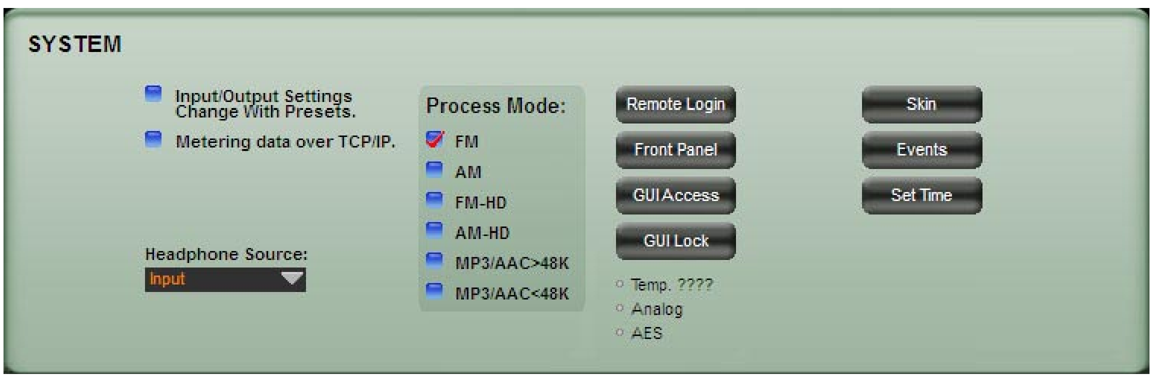


MP3/ACC <48K

# Vorsis VP-8 Process Modes

## Introduction

The VP-8 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-8 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.


### Mode Behavior Overview

Processing presets in the VP-8 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-8 is in its “FM” mode, that preset becomes an “FM” preset. Then if that preset is recalled later when the VP-8 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-8 GUI are the mode selection buttons, as well as a checkbox called “Discard Last Mode When Changing Presets” that allows presets to be “disconnected” from the mode they were created with. The purpose of this feature is to allow a user to create a family of presets that have the same settings up through the four-band AGC/Compressor without having to manually recreate the “front-end” settings for each preset in the family.

## 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.

## 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 preprocessing 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-8 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-8 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-8 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+.

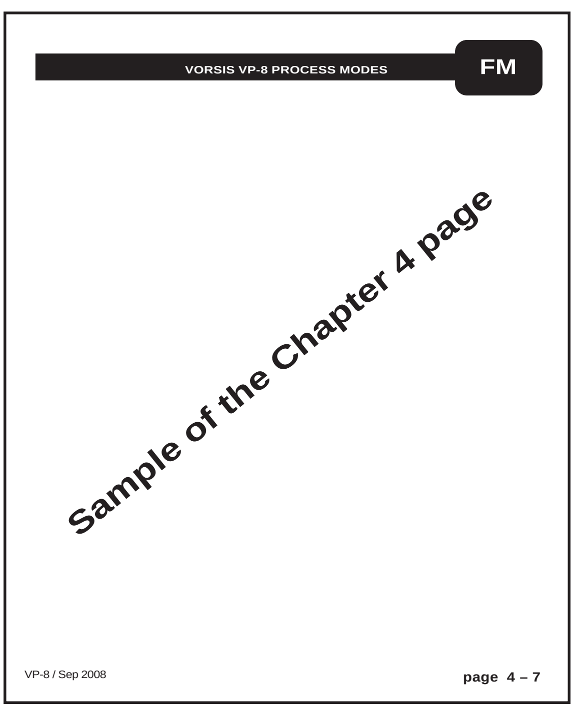
## Processing Common to All Modes

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 the Vorsis-exclusive eight band peak limiter. Operating modes other than the conventional analog AM and FM modes utilize an oversampled, lookahead 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 lookahead modes as desired.

Changing from one VP-8 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.



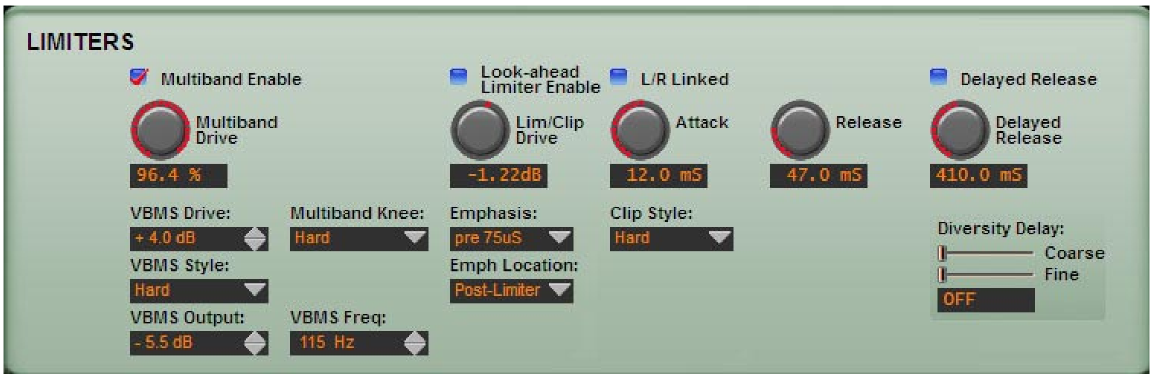

NOTE:
The balloon at the upper right on the following pages clearly indicates the processing mode described on that page.

# VP-8 FM Process Mode

The following discussion is unique to the Vorsis VP-8 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**

### Eight Band Limiter

The VP-8 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 longer 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 Management System (VBMS) to intelligently control bass program energy. In fact the VBMS is the reason why we can recommend longer limiter attack times at low frequencies - the VBMS will manage the (intentional) limiter overshoots caused by lengthening the bass attack times in an artistic way.

## Multiband Enable


This checkbox determines whether or not any processing is to be applied by the eight band multiband limiter and clipper. 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 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 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.

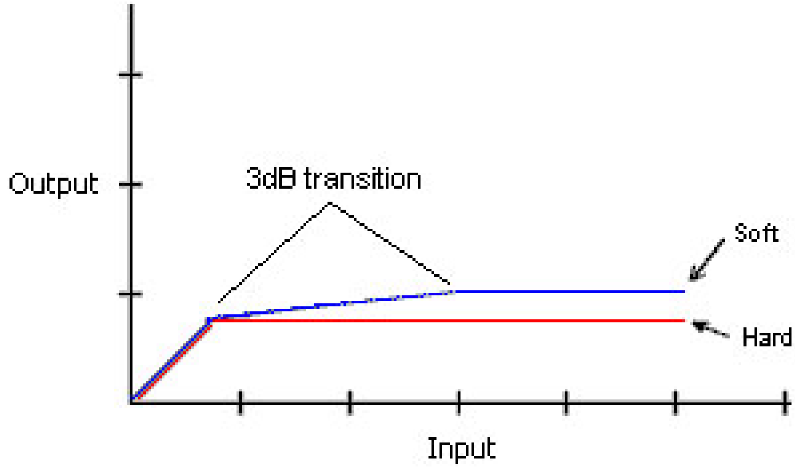
The Drive control adjusts the overall gain into the eight band limiter section, and its relative setting is dependent upon the setting of the Makeup control in the four band AGC. For this reason, the Drive control is not labeled in “dB” but instead is labeled in percent (%). With sane settings of the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

Higher Drive settings also cause deeper levels of clipping in the main clipper algorithm if the Lim/Clip Drive is not first reduced before the Multiband Drive is advanced. Therefore, if you feel that the Drive control is adding undue distortion, it means that the Lim/Clip Drive control setting should be reduced by several dB before the Multiband Drive is advanced. The Multiband Drive and 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.

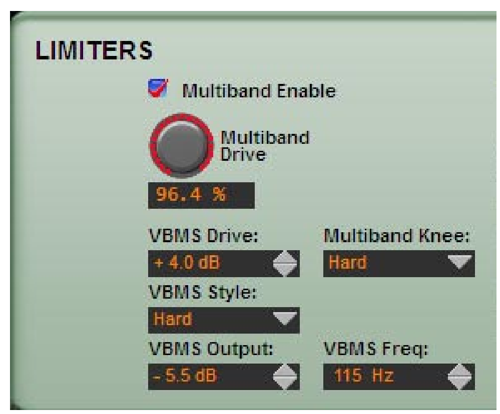



**Comparison of Hard and Soft Limiter Modes**

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.

## Vorsis Bass Management System (VBMS)

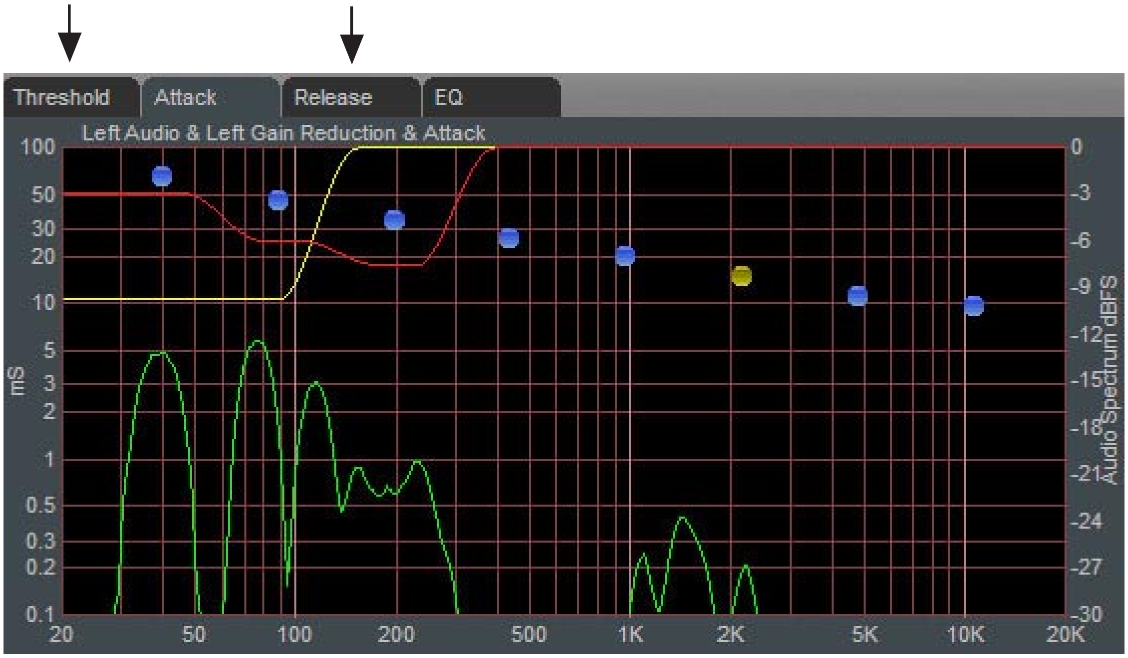



The Vorsis Bass Management System processes extremely low program frequencies in an artistic and natural program dependent manner without causing dynamic gain changes and undesirable intermodulation artifacts. Its primary task is two-fold: allow percussive instruments to sound natural while still being well controlled in peak amplitude, and prevent the generation of intermodulation between low frequency signals and those in the rest of the audio spectrum.

The Vorsis Bass Management system uses special DSP algorithms that interoperate with the multiband limiter's lower frequency bands to control low frequency program energy. Fully tunable, it allows bass punch to be precisely set for the type of program material and offers two operating modes plus an OFF setting.

The VBMS offers a visual indication of program induced bass control which can be viewed in the limiter graphic as a solid yellow line extending from the VBMS Frequency setting downward to 20Hz, similar to that shown below.

VBMS Activity Region




**VBMS Activity Display - 20Hz up to VBMS Frequency Setting**

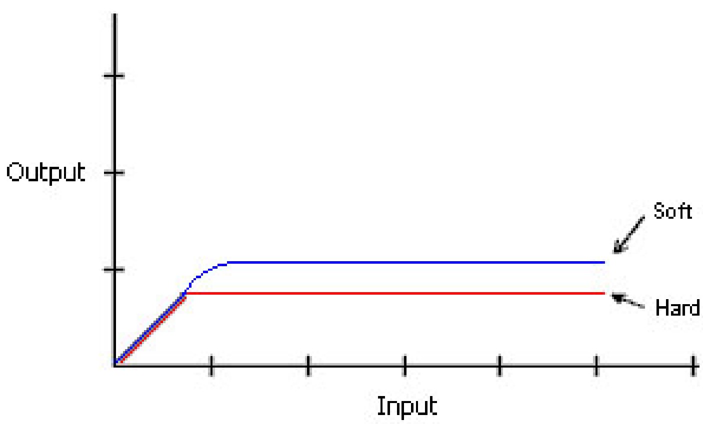
## VBMS Drive

Adjusts the amount of drive to the VBMS algorithm and is adjustable over a range of +/- 10dB. Higher drive settings cause bass energy within the "VBMS Freq" bandwidth to spend more time being controlled by the VBMS, which then increases the amount of low frequency energy loudness relative to higher frequency program material.

## VBMS Style

There are two VBMS operating styles available.

**Hard** - has a transfer function occupying approximately 0.25dB of transition amplitude. The transfer function is moderately abrupt and generates low order harmonics. Afterwards, and in a process that borrows from the technology of our Timbral mode, used in the AP-2000, we mathematically calculate and reduce significant harmonics above the fifth.

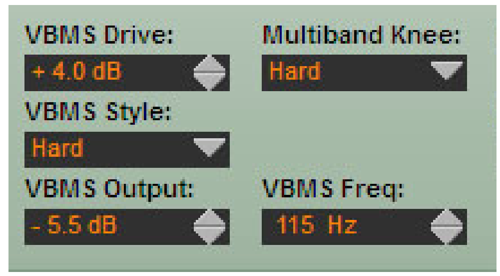



**Comparison of VBMS Operating Modes**

**Soft** - operates with a transfer function occupying approximately 1.00dB of transition amplitude. The transfer function is less abrupt than the hard setting, generating even fewer higher order harmonics, and in this mode we calculate and reduce energy above the third harmonic.

The two modes may appear to be nearly identical due to the small difference in transfer functions. Rest assured, however, that there is a *significant* difference in the way the two styles sound on low frequency program material.

### VBMS Output




Adjusts the bass contribution from the VBMS algorithm and may be thought of as the VBMS *output mix* control. The control is adjustable over a range of +/-10dB.

Generally speaking, the VBMS Drive may be adjusted to personal taste and format requirements. Care should be exercised in the setting of the VBMS Output control, though, because setting it higher than approximately +0dB might cause the peak amplitude of bass material to confuse the main clipper distortion masking scheme, unmasking undesirable distortion products in strange and non-intuitive ways.

The higher the VBMS Frequency is tuned the more energy is managed within the VBMS algorithm and therefore the more active the VBMS activity indication will be. There is no right or wrong amount of VBMS processing - whatever sounds best for the format and market is correct.

Note that overdriving the VBMS will not cause obnoxious forms of distortion. This is because the VBMS algorithm is intelligent in that it “knows” about material passing through itself relative to that occurring in the rest of the spectrum and adapts to minimize unintentional intermodulation.

### VBMS Frequency

The VBMS operating bandwidth is adjustable from 60Hz to 300Hz. Lower frequency settings produce a “heavier” and “tighter” sound, while higher settings result in less coloration while still favorably controlling bass energy excursions. This control should be adjusted to personal taste and program format with one caveat:

Bass energy that is NOT handled by the VBMS will have to be managed by the main clipper. What we mean by this is if the VBMS Freq is set to 80Hz then the advantages of VBMS will only extend from 80Hz downward, leaving bass energy above 80Hz to be managed by the main clipper. This may or may not be acceptable depending on the type of program material you are processing.

In formats where the program content contains very heavy low bass, such as in R&B, Rap, and Dance, the VBMS Frequency should be tuned to between 120 and 180Hz for best on-air results. Other formats might have the control set to higher frequencies. Classical and Jazz, for instance, might see the frequency set to 250Hz, or perhaps all the way up to 300Hz. A good tradeoff setting for many formats is 180Hz.

Although the VBMS Frequency control is adjustable down to 60Hz we have yet to find program material that requires it to be set to such a low setting.


## Emphasis Selection

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

## Emphasis 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.

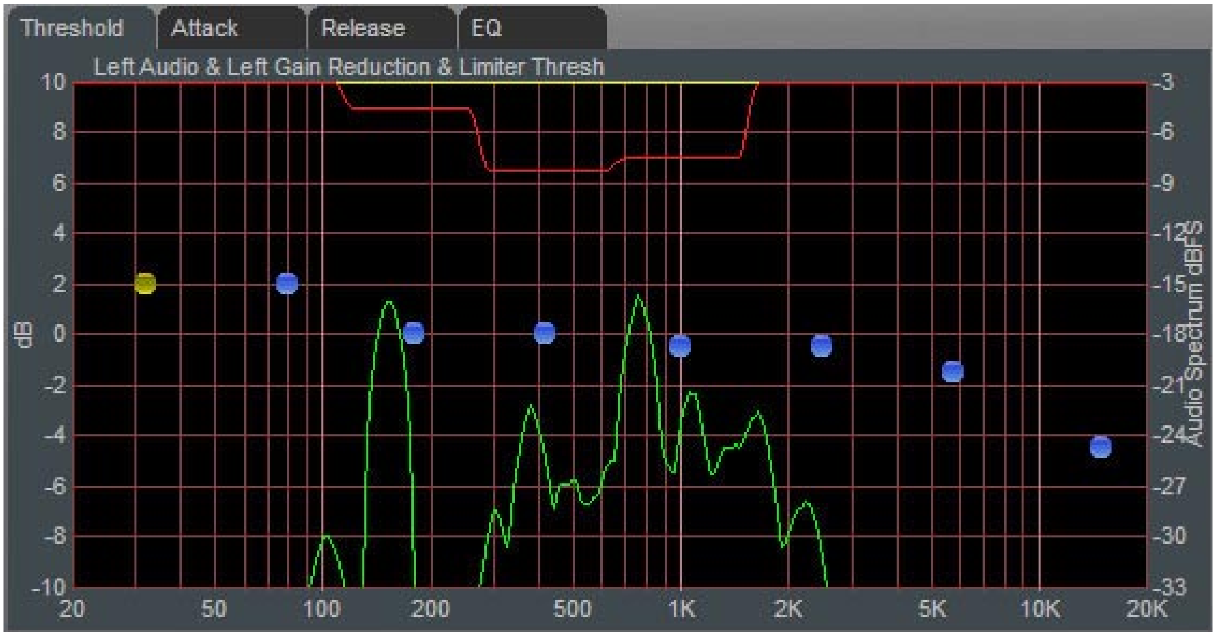
**Post-Limiter —** In this position the pre-emphasis is *after* the eight band section and Look Ahead limiter and *before* the main clipper.

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.

## Eight Band Limiter Dynamics Screens

### Threshold Screen

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




**Eight Band Limiter Threshold Screen**

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

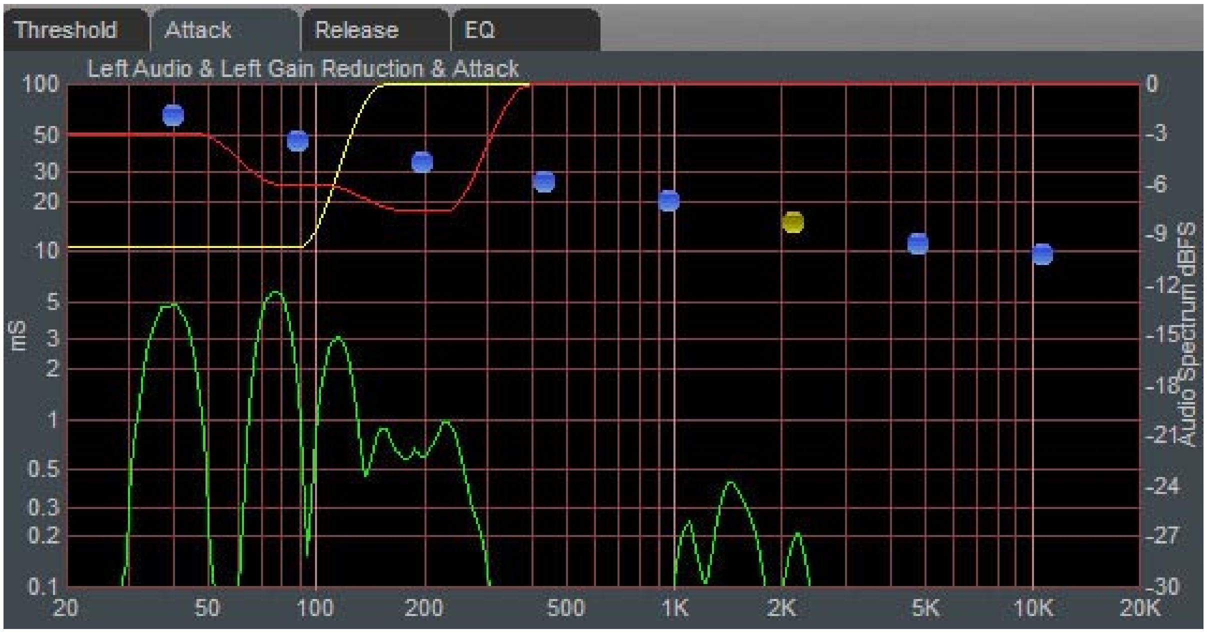
Each of the limiter thresholds can be adjusted separately or in groups according to the manner described in the earlier VP-8 Graphical Interface Operation section. Lower settings (more negative numbers) reduce the threshold, causing limiting to occur at lower audio levels. Conversely, higher thresholds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




**Eight Band 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.

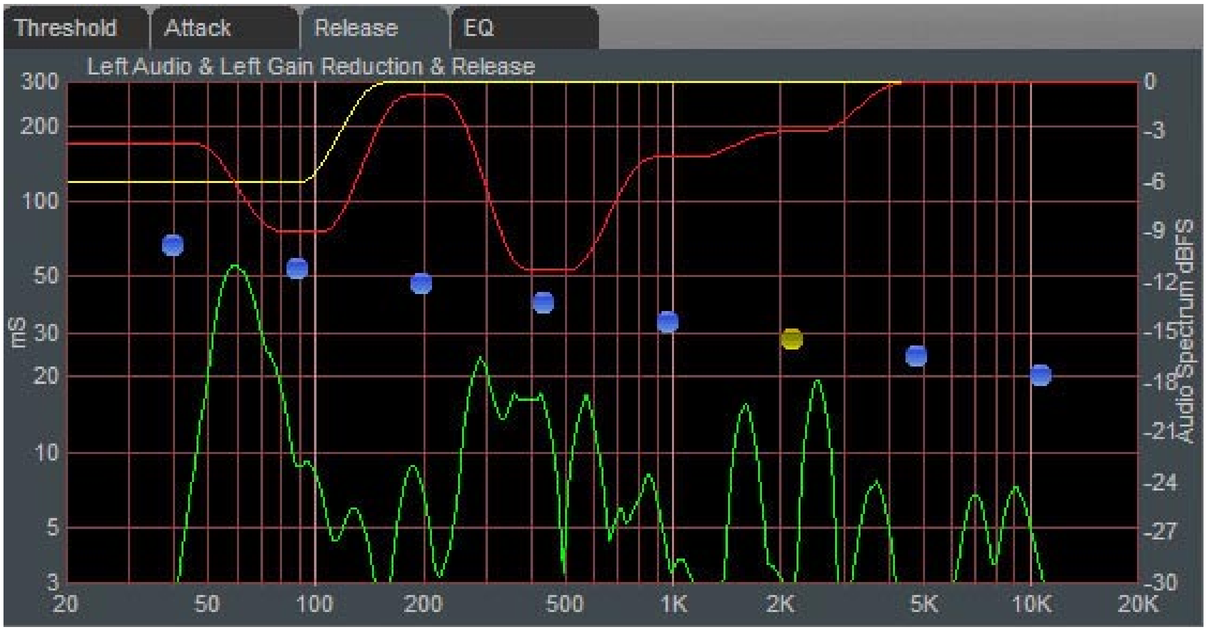
Attack times are generally adjusted to taste, keeping in mind that the slower the attack time the more work the final clipper or limiter must do, while conversely the faster the attack time the fewer short term dynamics the program material will have.

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.



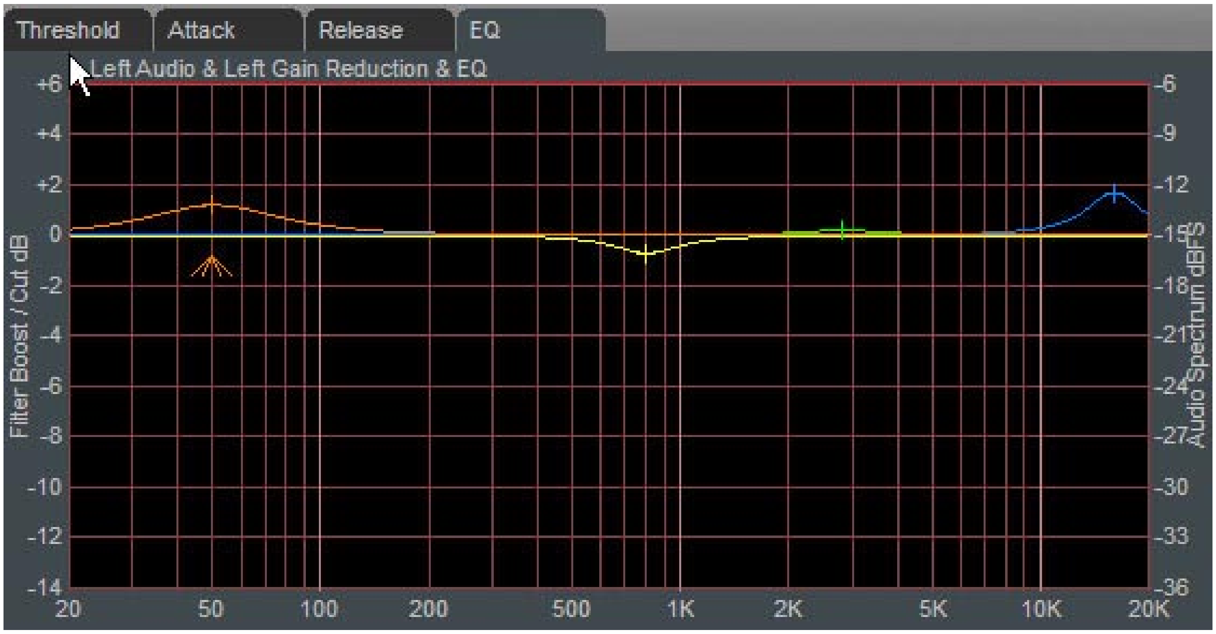

**Eight Band Limiter Release Time Screen**

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.




**Eight Band 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.

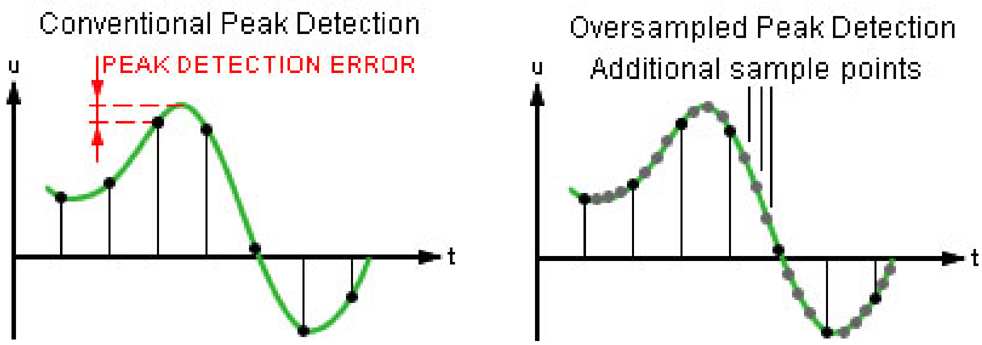
Small adjustments in EQ boost — on the order of 1dB or 2dB at most — are sufficient to alter the overall tonality of the eight band section.

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.


### Limiter Enable

When this checkbox 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 (lookahead) because the total "lookahead delay" is 0.5mS. Therefore a setting of 0.5mS creates a limiter with zero attack time (again, because the total "lookahead 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 Delayed Release checkbox, 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 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-8 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-8’s internal DSP-based precision stereo generator.

### Stereo Width 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 <u>mono</u> 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 Stereo Width 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 Stereo Width 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 Stereo Width 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 Vorsis approach is more intelligent in how it works; it does not *decrease* stereo separation on program material that does not *need* it.

## Operating the Vorsis BS-412 MPX Power Controller

In the upper left corner of the Stereo Encoder screen of the VP-8's FM Mode is the control associated with the BS-412 MPX Power Controller. The control has 15 possible settings, plus OFF: +8db, +7dB, +6db, +5dB, +4dB, +3dB, +2dB, +1.5dB, +1.0dB, +0.5dB, 0, -0.5dB, -1.0dB, -1.5dB, and -2.0dB.

The 0.5dB resolution near the control's zero setting 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. Though the MPX Power Controller is not easily fooled we provided a small amount of tweaking near the "0dB" reference level "just in case".

### Turning the BS-412 Controller On

**IMPORTANT!** *The MPX Power Controller's 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!***

## Lim/Clip (Limiter/Clipper) Drive


The Lim/Clip Drive control adjusts the drive level to whichever composite processor is selected by the Multiplex Process checkbox 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.

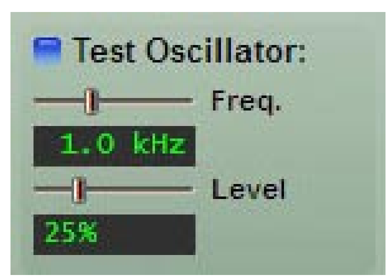
## DeEmphasis

When the DeEmphasis checkbox 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 checkbox 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.

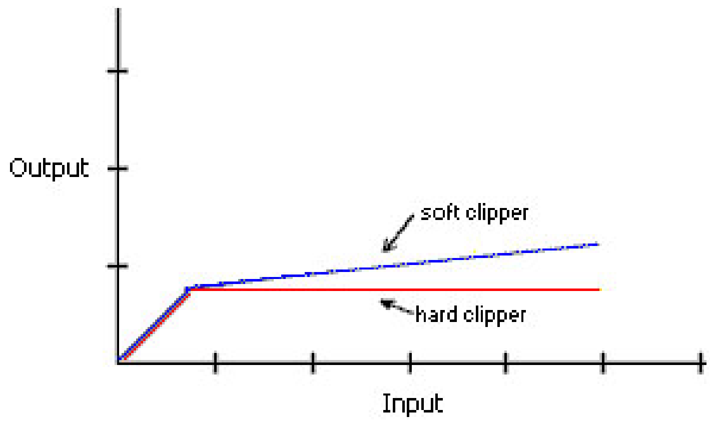
## Multiplex Process


The VP-8 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.

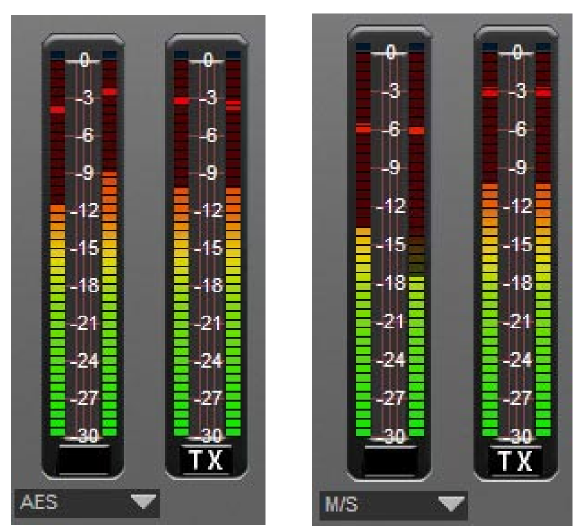



**Comparison of Hard and Soft**

**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.

## Output Metering

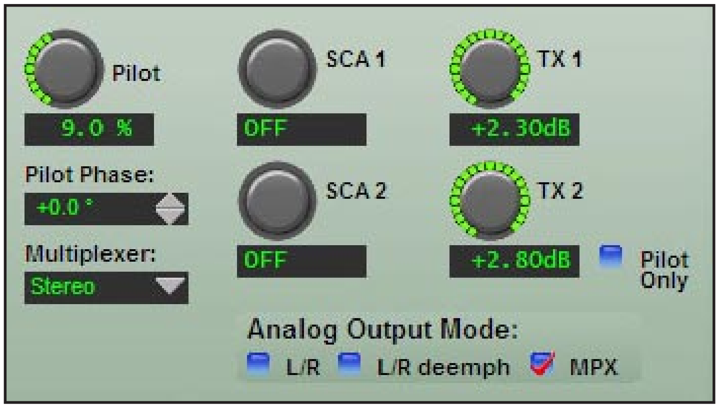
As indicated earlier (see Chapter 3, page 3-17), the metering bargraphs in the GUI can be switched to indicate operating levels of the Sum and Difference (M/S) levels or the peak output levels of the FM AES3 output (AES). This is done with the dropdown selector below the meter in the lower part of the GUI. The TX 1 and TX 2 outputs are always available on a second meter.




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

## Stereo Encoder

The VP-8 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

The pilot 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.

## 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-8 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.

### 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.

### Pilot Only

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

### 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-8 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.

### Multiplexer

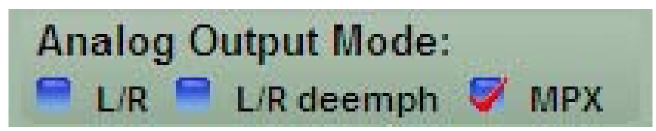
The stereo encoder can be operated in Stereo or Mono by selecting the desired operating mode with this switch. When in mono, the Stereo Pilot is turned off.

### Mask Filter

Regardless of which Multiplex Process algorithm is chosen, out of band harmonics above 53kHz are tightly controlled by a Mask Filter. The Mask Filter provides a protection notch for the stereo pilot as well as protection above 53kHz for the RDS and SCA regions. The Mask Filter cannot be defeated.

## 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-8 AM Process Mode

The following discussion is unique to the Vorsis VP-8 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-8 is in its AM mode.

Audio processing while in AM Mode takes place in the Sum and Difference domain, which means the VP-8 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-8 has perfect mono compatibility because in Mono mode the L-R channel is simply discarded.

Note: In ALL VP-8 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**

## Eight Band Limiter

The VP-8 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.

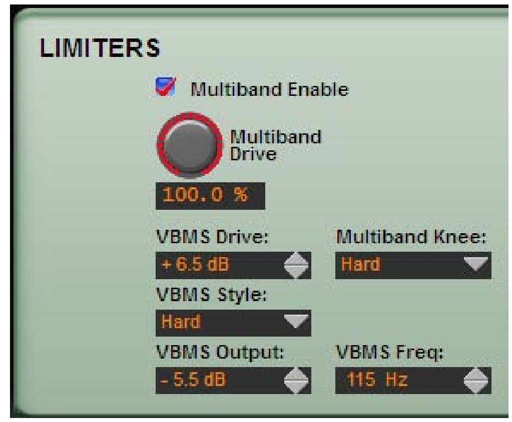
The eight band limiter's operating parameters are fully adjustable in order to accommodate a wide variety of program material and competitive needs. It also contains the embedded Vorsis Bass Management System (VBMS) which is tightly integrated within the limiter structure to intelligently control bass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

## Multiband Enable

This checkbox 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 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 drive settings that result in more than 12-15dB of continuous gain reduction are typically unnecessary with most AM formats.

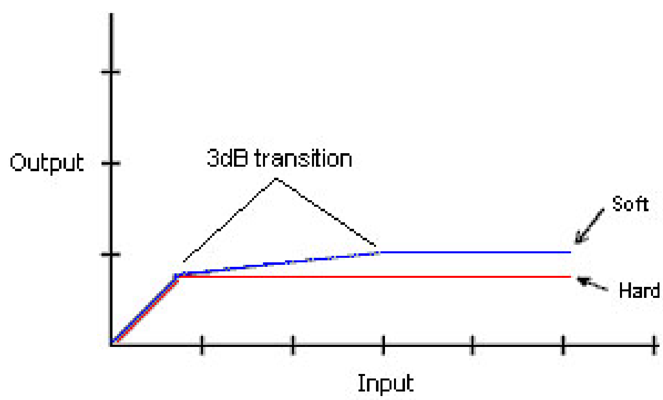



The drive control setting is also somewhat dependent upon the setting of the Makeup control in the four band AGC. For this reason, the control is not labeled in “dB Drive,” but instead is labeled in percent (%). With sane settings of the four band AGC's Makeup control the correct eight band limiter Drive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

Higher 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.



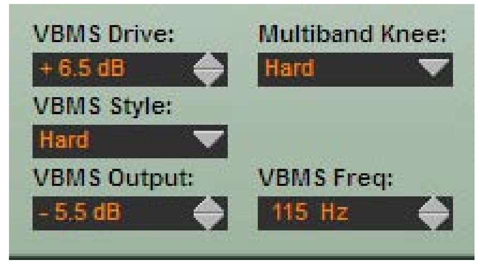

**Comparison of Hard and Soft Limiter Modes**

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.

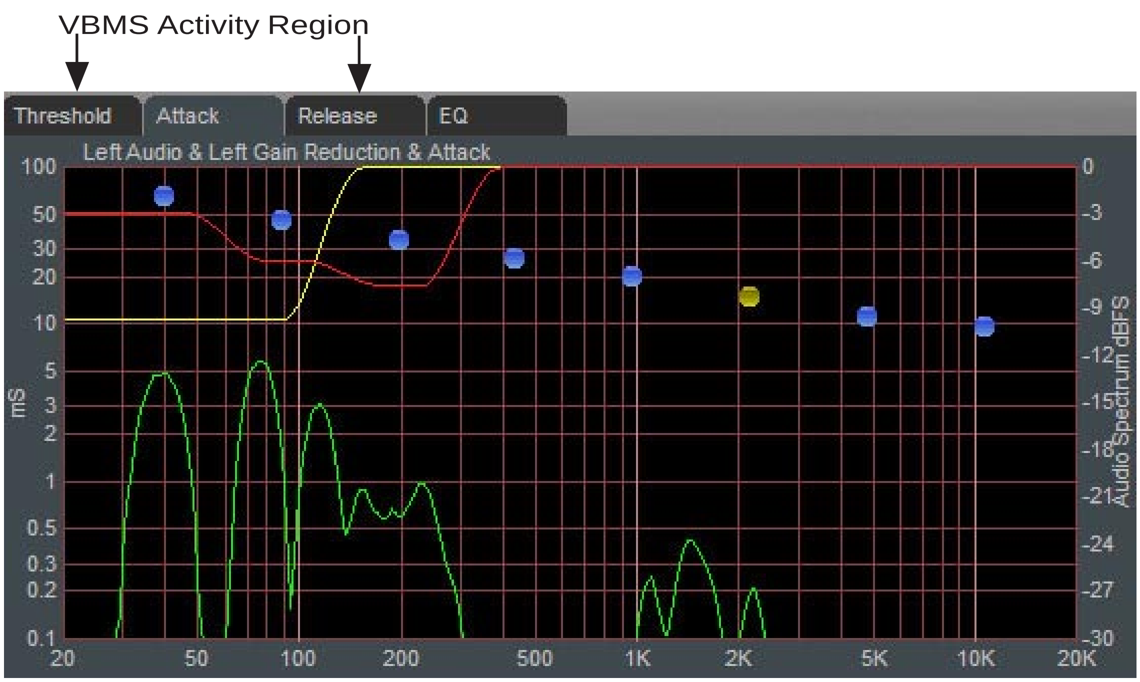
## Vorsis Bass Management System (VBMS)




The Vorsis Bass Management System processes extremely low program frequencies in an artistic and natural program dependent manner without causing dynamic gain changes and undesirable intermodulation artifacts. Its primary task is to allow percussive instruments to sound natural while still being well controlled in peak amplitude, while also controlling long-term bass energy to prevent intermodulation with higher frequency material within the main clipper.

The Vorsis Bass Management System uses special DSP algorithms that interoperate with the multiband limiter's lower frequency bands to control low frequency program energy. The algorithm is fully tunable; it allows bass punch to be *precisely* set for any type of program material and offers two operating modes plus an OFF setting.

The VBMS offers a visual indication of program induced bass control which can be viewed in the limiter graphic as a solid yellow line extending from the VBMS Frequency setting downward to 20Hz, similar to that shown below. Note that the lower frequency limit for VBMS operation is also a function of the setting of the High Pass Filter setting in the Input Screen!




**VBMS Activity Display - 20Hz up to VBMS Frequency Setting**

### VBMS Drive

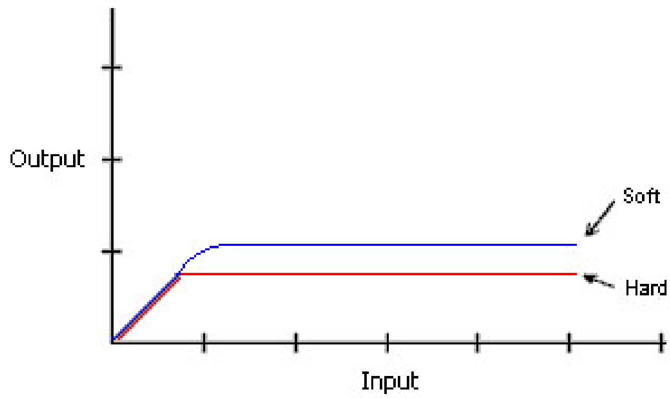
Adjusts the amount of drive to the VBMS algorithm and is adjustable over a range of +/- 10dB. Higher drive settings cause bass energy within the “VBMS Freq.” bandwidth to spend more time under control of the VBMS, which increases the amount of low frequency loudness relative to higher frequency program material.

### VBMS Style


There are two VBMS operating styles available.

**Hard** - has a transfer function occupying approximately 0.25dB of transition amplitude. The transfer function is moderately abrupt and generates low order harmonics. Afterwards, and in a process that borrows from the technology of the Timbral mode in our Vorsis flagship processor, the AP-2000, we mathematically manipulate harmonics above the fifth.

**Soft** - operates with a transfer function occupying approximately 1.00dB of transition amplitude. The transfer function is less abrupt than the hard setting, generating even fewer higher order harmonics, and in this mode we manipulate energy above the third harmonic.




**Comparison of VBMS Operating Modes**

The two VBMS Styles may appear to be nearly identical due to the small difference in transfer functions. Rest assured, however, that there is a *significant* difference in the way the two styles sound on low frequency program material.

### VBMS Output

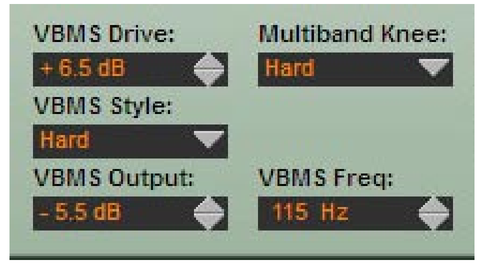
Adjusts the bass contribution from the VBMS algorithm and may be thought of as the VBMS *output mix* control. The control is adjustable over a range of +/-10dB.

Generally speaking, the VBMS Drive control may be adjusted to personal taste and format requirements. Care should be exercised in the setting of the VBMS Output control, though, because setting it higher than approximately -3dB *might* confuse the main clipper's distortion masking scheme and cause undesirable distortion products on some program material in quite strange and non-intuitive ways.

The higher the VBMS Frequency is tuned the more energy is managed by the VBMS algorithm and therefore the more active the VBMS activity indication will be. There is no right or wrong amount of VBMS processing - whatever sounds best for the format and market is correct. See "VBMS Frequency" below for some guidelines.

Note that overdriving the VBMS will not cause obnoxious forms of distortion. This is because the VBMS algorithm is intelligent in that in "knows" about material passing through itself relative to what is occurring in the higher frequency path, and continuously adapts to minimize unintentional intermodulation.

## VBMS Frequency



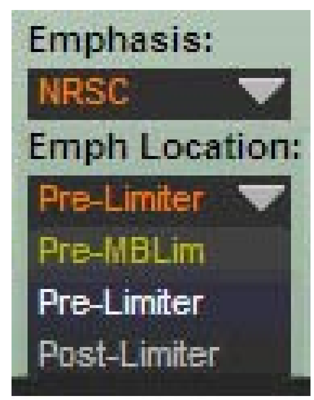

The VBMS operating bandwidth is adjustable from 60Hz to 300Hz. Lower frequency settings produce a “heavier” and “tighter” sound, while higher settings result in less coloration while still favorably controlling bass energy excursions. This control should be adjusted to personal taste and program format with one caveat:

Bass energy that is *not* handled by the VBMS *must* be managed by the main clipper. By way of example we mean that if the VBMS Freq is set to 80Hz then the advantages of VBMS will *only* extend from 80Hz downward. Bass energy *above* 80Hz will be (necessarily) managed by the main clipper. This may or may not be desirable depending on the type of program material you are processing.

In formats where the program content contains very heavy low bass, such as in R&B, Rap, and Dance, the VBMS Frequency should be tuned to between 120 and 180Hz for best on-air results. Other formats might sound best with the control set to higher frequencies. Classical and Jazz, for instance, might require the frequency to be set to 250Hz, or perhaps all the way up to 300Hz. A good starting setting for many music formats is 180Hz, while Talk formats might find that it sounds better with the VBMS Frequency set between 250Hz and the 300Hz maximum upper limit.

Note that although we provided a means to do so, we do not recommend operating the Multiband Limiter with the VBMS algorithms turned off!

## Emphasis Selection




Pre-emphasis is selectable as NRSC or Flat (OFF).

## Emphasis Location

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

**Pre-MBLim** — Pre-emphasis is placed *before* the eight band limiter/clipper section and the eight band section may be operated as a frequency conscious limiter tailored to the inverse of the selected pre-emphasis.

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

**Post-Limiter** — Pre-emphasis is *after* the eight band section and Look Ahead limiter and *before* the main clipper.

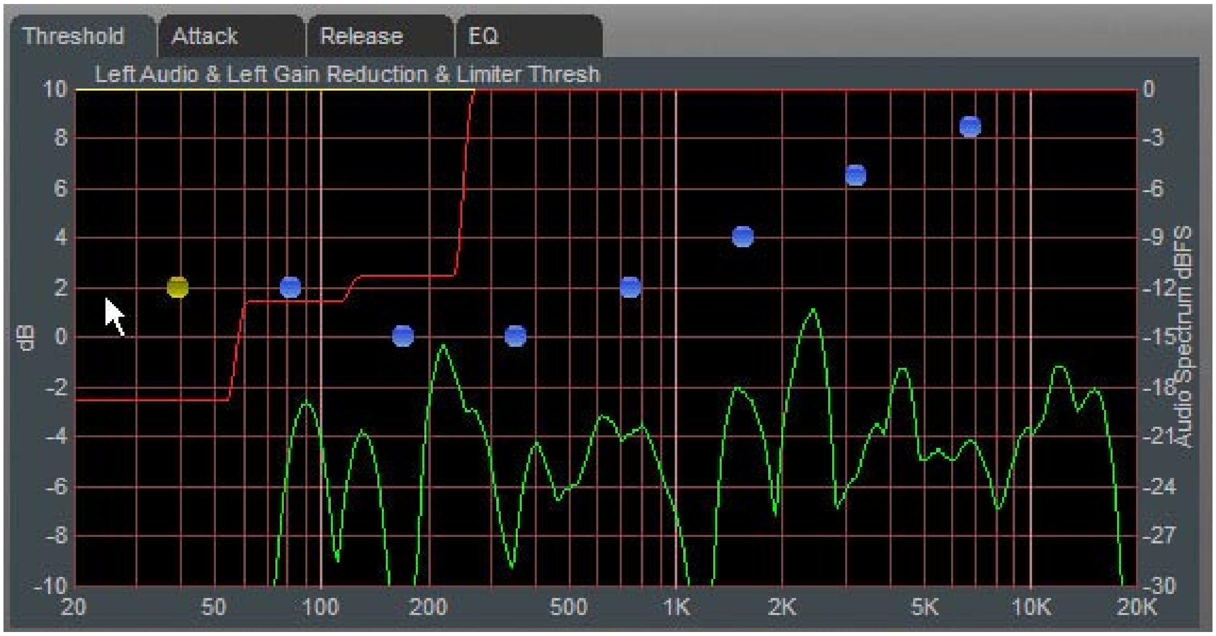
Entirely different sound and spectral energy textures may be created by placing the pre-emphasis in one location versus the other. There is no wrong setting unless that setting either does not produce the desired on air sound or creates undesirable artifacts.

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.

## Eight Band 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.




**Eight Band 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.

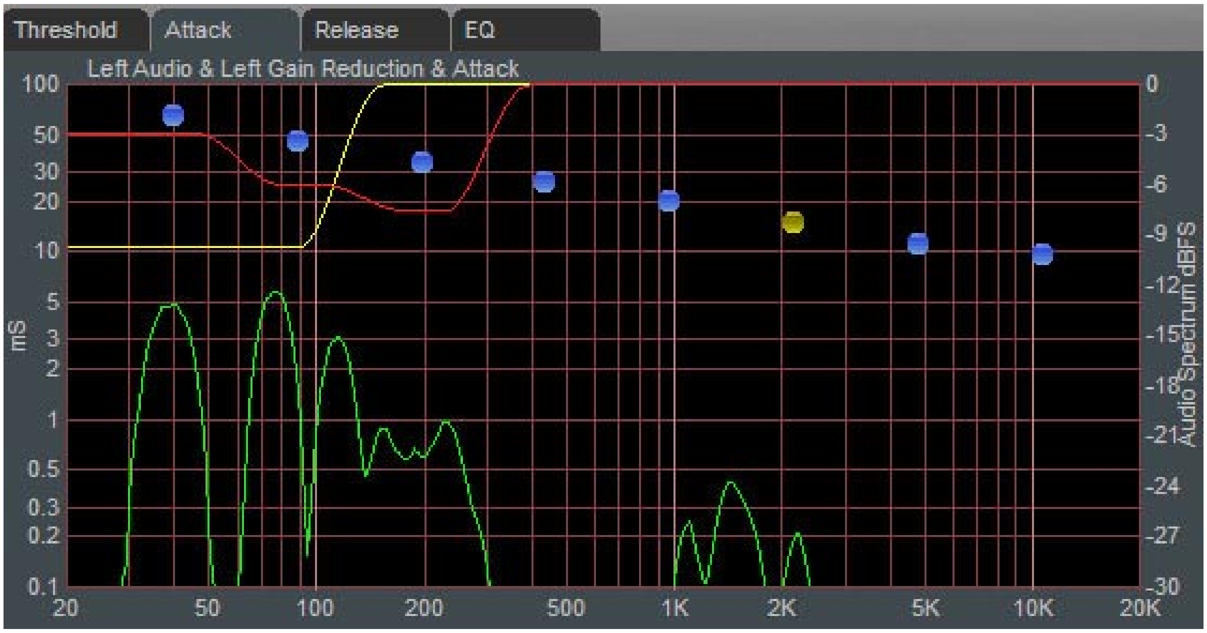
Each of the limiter thresholds can be adjusted separately or in groups according to the manner described in the earlier VP-8 Graphical Interface Operation section. Lower settings (more negative numbers) reduce the threshold, causing limiting to occur at lower audio levels. Conversely, higher thresholds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




**Eight Band 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.

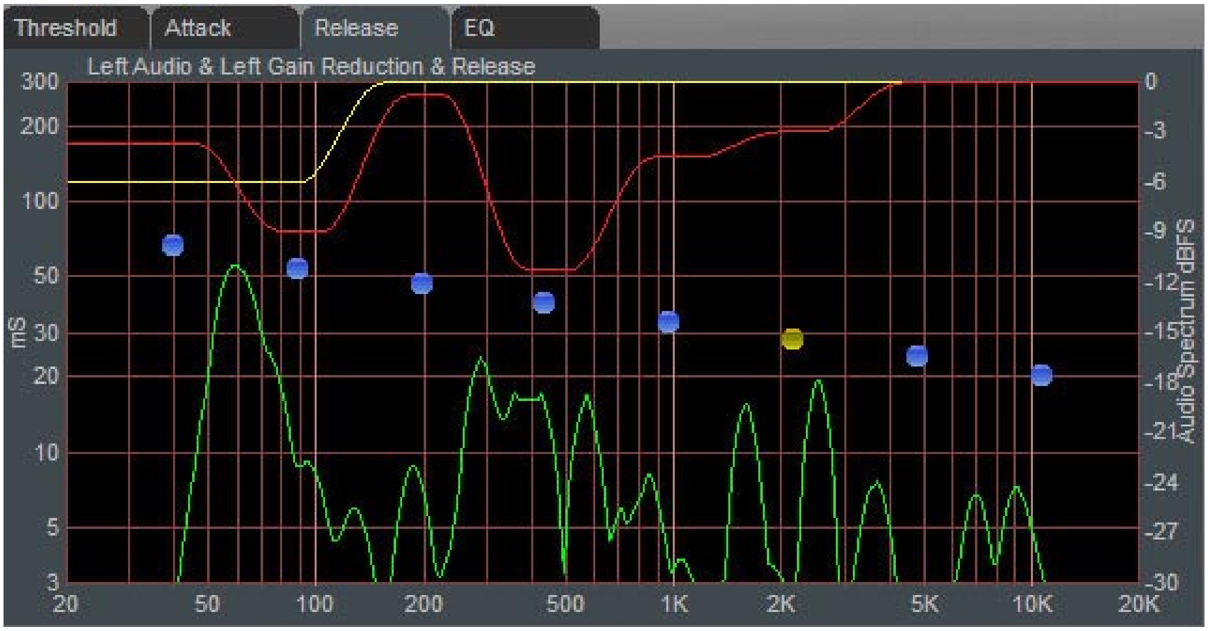
Attack times are generally adjusted to taste, keeping in mind that the slower the attack time the more work the final clipper or limiter must do, while conversely the faster the attack time the fewer short term dynamics the program material will have.

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.



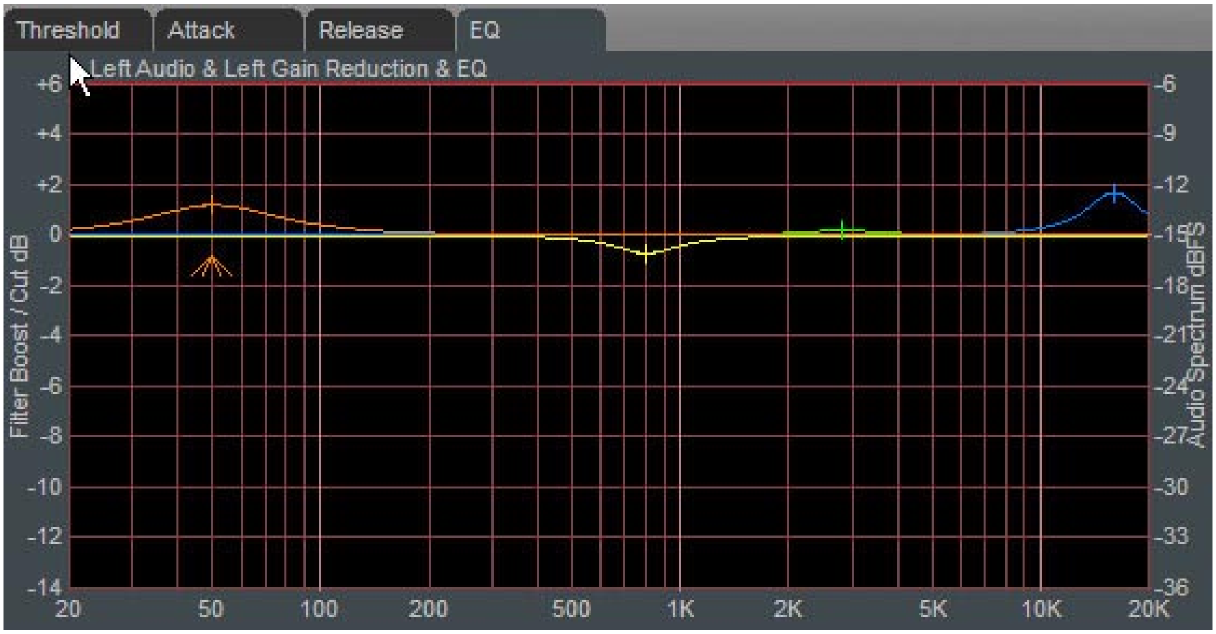

**Eight Band Limiter Release Time Screen**

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.




**Eight Band 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.

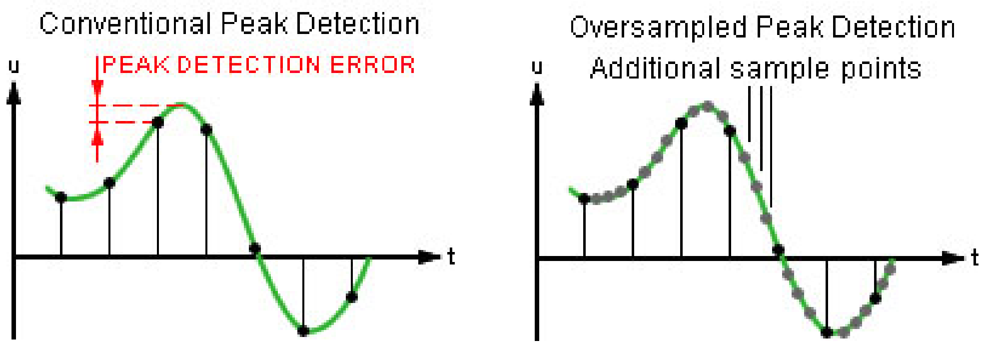
Only very small amounts of boost equalization are recommended once final limiting has been accomplished in the eight band section. In fact, the adjustment range has been limited to +3dB/-12dB in order to curtail the temptation to perform aggressive equalization where it is the most inappropriate, and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

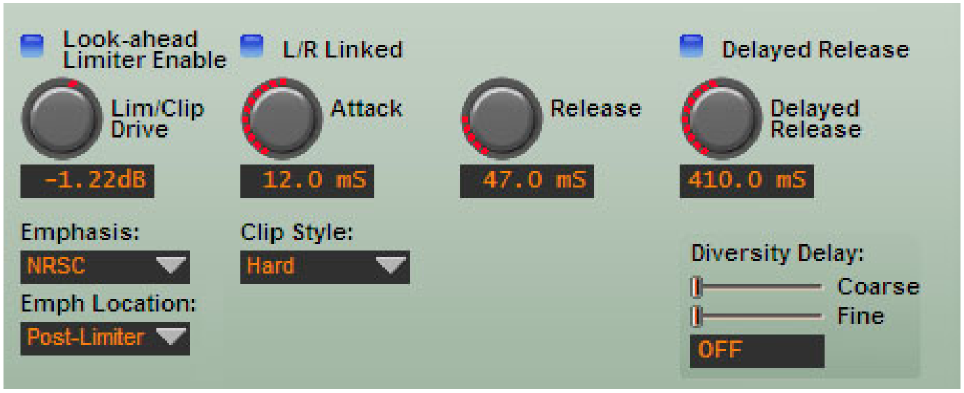
If desired, the AM output may be peak controlled by a highly oversampled feed forward limiter with a 0.5 millisecond lookahead time. The control path (not the audio path) is highly oversampled in order to achieve extremely precise peak control. The primary goal of oversampling the control sidechain is 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.




### Limiter Enable

When this checkbox is checked the look ahead limiter is enabled. Likewise, when Limiter Enable 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.

## 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 "lookahead" 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 "lookahead" 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 Delayed Release checkbox is checked (see below).

When Delayed Release is engaged via the Delayed Release checkbox, 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 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-8 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.

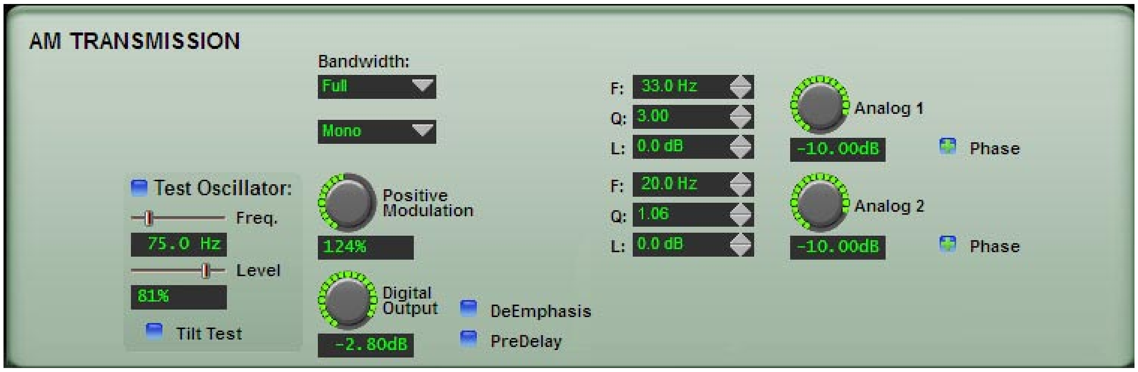
The **Fine** control covers a range of 50 milliseconds in 0.1 millisecond (100 microsecond) steps.

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-8, 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.

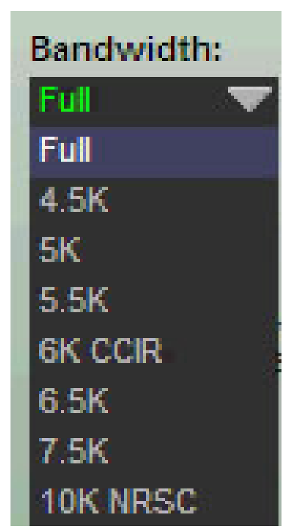
### Test Oscillator

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 checkbox 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-8's outputs in Stereo M/S mode, it can be accomplished by adjusting the "L-R" control located under the Drive control on the AGC/Compressor screen of the GUI. A +/-6dB adjustment range has been provided.

### 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-8 results in a +24dBu PEAK output level at its balanced output!

### Analog 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.

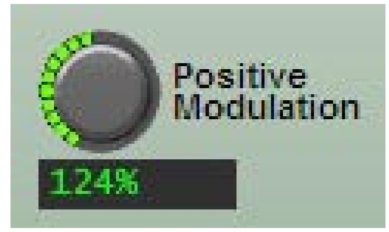
The output level control should be adjusted to provide the desired negative modulation percentage. Positive modulation (asymmetry) may be set using the Positive Modulation control.

### Analog 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.

## Positive Modulation




This control adjusts the asymmetry of the VP-8's final clipper, which adjusts the amount of positive modulation compared to a fixed negative value. The VP-8’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-8’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-8 and the transmitter is mismatched! You can change the polarity by either reversing the analog audio pair in the cable between the VP-8 and the transmitter, or more simply, flip the audio phase by clicking the Phase button next to the Analog 1 or Analog 2 output level control (if in a Stereo mode).

## 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 35.90 dBFS to 0 dBFS.

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'.

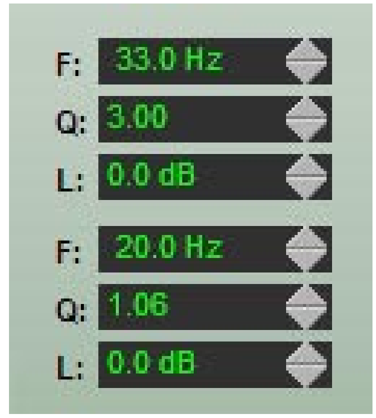
## DeEmphasis

When checked, applies complimentary de-emphasis to the AES3 digital output.

## PreDelay

When checked, the AES Output signal is taken *before* the Diversity Delay.

## 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

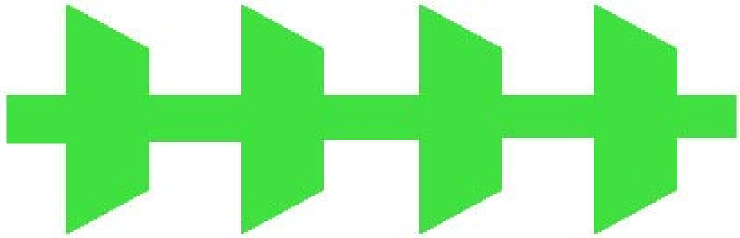
Adjusting the AM Tilt Corrector is a simple process. The first step is to determine if the transmitter will require Tilt Correction in order to modulate well at low frequencies and the following procedure will determine this:

1. Connect the VP-8 to the audio input of the transmitter to be tested.

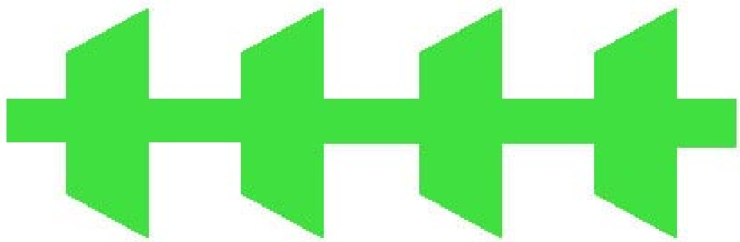
2. Connect the transmitter to either a dummy load of rated power or to the antenna system.

3. Turn on the VP-8'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.

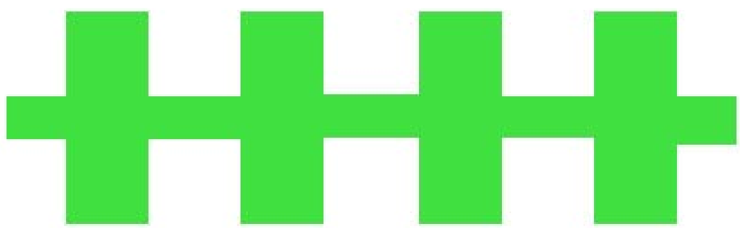
4. Turn on the VP-8 Test Oscillator “Tilt Test” by clicking its checkbox. Using an oscilloscope connected to a low level RF sample of the transmitter output observe the resulting RF waveform. It should appear like waveform A or waveform C below, or somewhere in between the two extremes, depending on 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).*

# VP-8 FM-HD Process Mode

The following discussion is unique to the Vorsis VP-8 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-8. These are the Limiters and FM-HD screens, the latter being visible and selectable via the FM-HD tab when VP-8 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**

### Eight Band Limiter

The VP-8 contains a Vorsis 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:

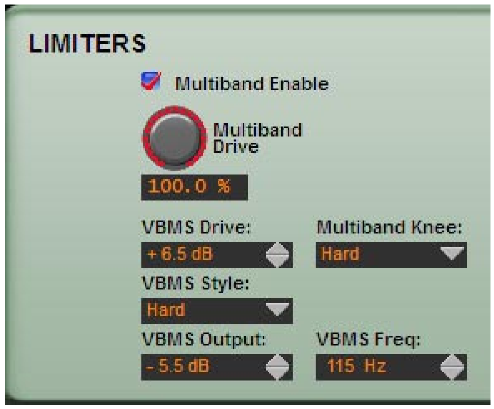
1. Extremely short attack times, while affording virtually perfect peak control, remove so much transient energy as to make the sound too homogenized and lifeless.

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 Management System (VBMS) which is tightly integrated within the limiter structure to intelligently control bass program energy.

## Multiband Enable




This checkbox 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.

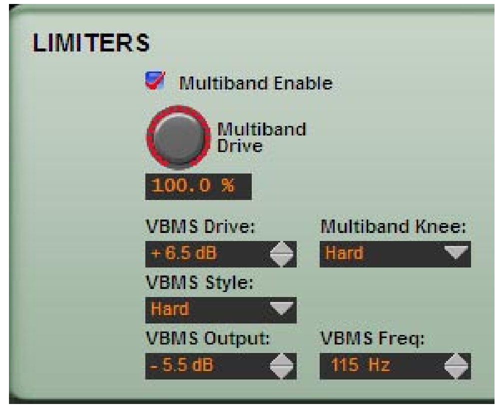
## Multiband Drive

This control determines the amount of input signal applied to the eight band limiter and, therefore, how hard it is being driven. Lower Drive settings create a more gentle sound while still controlling peak energy, while higher 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 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 Drive control setting will usually be between 85% and 95%. The full adjustment range is 0% to 100%.

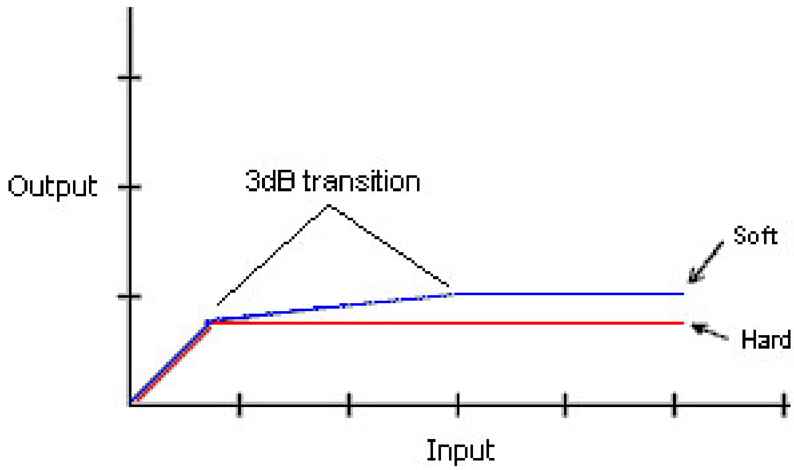
Higher Drive settings cause deeper gain reduction in the eight band limiter (as well as higher levels of limiting in the final look ahead limiter if the Lim/Clip Drive is not first reduced before the Multiband Drive is advanced). Therefore, Multiband Drive and Lim/Clip Drive are the pair of controls 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.

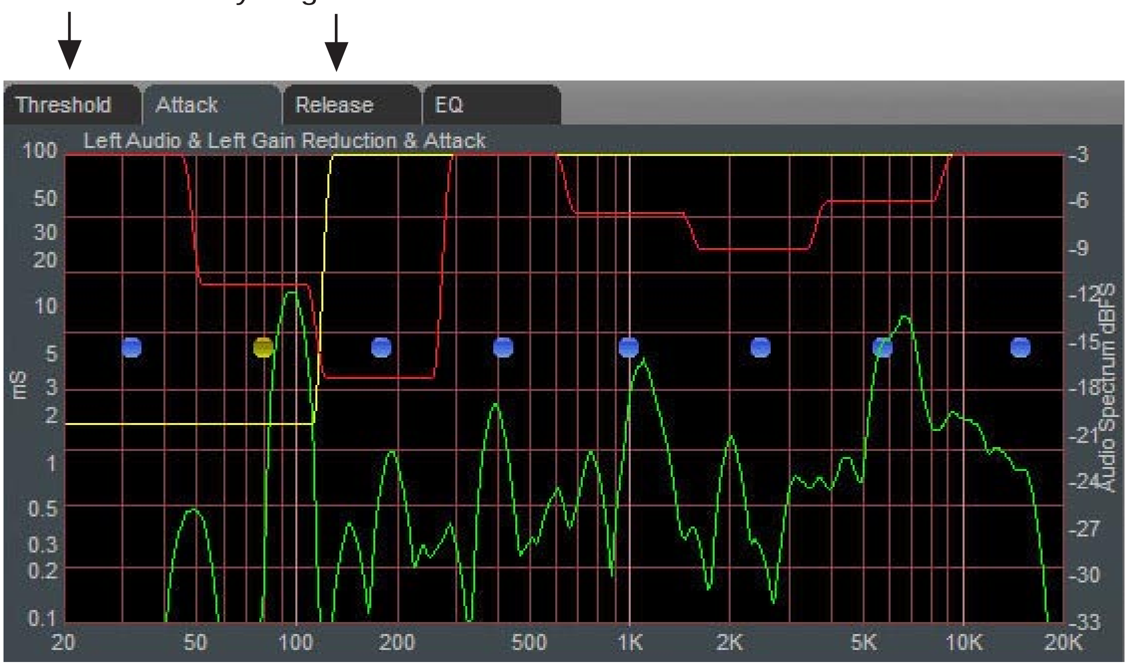
## Vorsis Bass Management System (VBMS)

The Vorsis Bass Management System processes low program frequencies in an artistic and natural program dependent manner without causing dynamic gain changes and undesirable intermodulation artifacts. Its sole task is to allow percussive instruments to sound natural while being well controlled in peak amplitude, while simultaneously controlling long-term bass energy to prevent intermodulation with higher frequency material within the main clipper.

The Vorsis Bass Management system uses special DSP algorithms that interoperate with several of the eight band limiter's lower frequency bands to control low frequency program energy. The algorithm is fully tunable, allows bass punch to be *precisely* set for any type of program material, and offers two operating modes plus an OFF setting, noting that we do not recommend turning the VBMS off!.

The VBMS offers a visual indication of when it is active and can be seen in the limiter graphic as a solid yellow line extending from the VBMS Frequency setting downward to 20Hz, similar to that shown below.

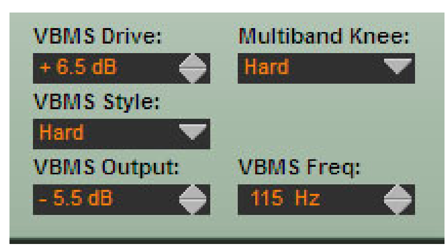
VBMS Activity Region




**VBMS Activity Display - 20Hz up to VBMS Frequency Setting**

Note that the lower frequency limit for VBMS operation is also a function of the High Pass Filter setting in the Input Screen!

### VBMS Drive



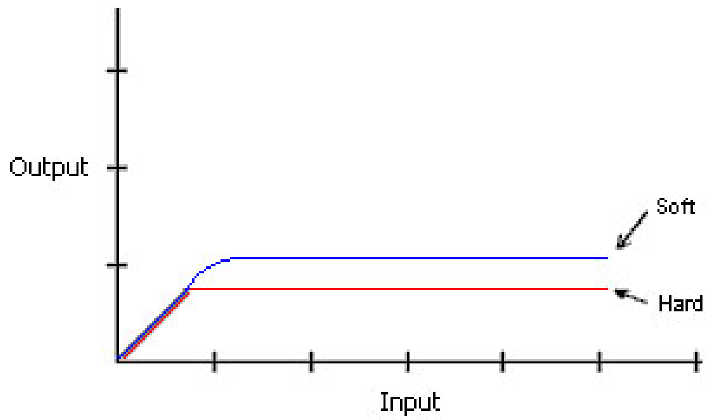

Adjusts the amount of drive to the VBMS algorithm and is adjustable over a range of +/- 10dB. Higher drive settings cause bass energy within the “VBMS Freq” bandwidth to spend more time under control of the VBMS, which increases the amount of low frequency loudness relative to higher frequency program material.

### VBMS Style

There are two VBMS operating styles available.

**Hard** - This transfer function is moderately abrupt and generates low order harmonics. Afterwards, and in a process that borrows from the technology of the Timbral mode in our Vorsis flagship processor, the AP-2000, we mathematically calculate and reduce significant harmonics above the fifth.

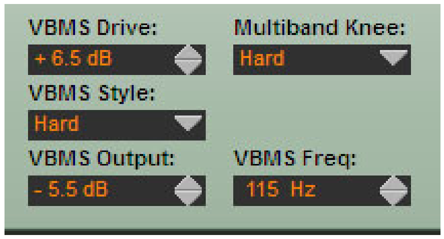
**Soft** - This transfer function is less abrupt than the hard setting, generating fewer higher order harmonics, and in this mode we calculate and reduce energy above the third harmonic.




**Comparison of VBMS Operating Modes**

The two VBMS styles may appear to be nearly identical due to the small difference in transfer functions. Rest assured, however, that there is a *significant* difference in the way the two styles sound on low frequency program material.

## VBMS Output




Adjusts the bass contribution from the output of the VBMS algorithm and is adjustable over a range of +/-10dB.

Generally speaking, the VBMS Drive and Output controls may be adjusted to personal taste and format requirements. Care should be exercised, though, in the setting of the VBMS Output control, because setting it higher than approximately -3dB *might* interfere with the main clipper distortion masking scheme when very heavy bass energy is present. Such interference could unmask undesirable distortion products on certain program material in quite peculiar and non-intuitive ways.

## VBMS Frequency

The VBMS operating bandwidth is adjustable from 60Hz to 300Hz. Lower frequency settings produce a “heavier” and “tighter” sound, while higher settings result in less coloration while still favorably controlling bass energy excursions. This control should be adjusted to personal taste and program format with one caveat:

Bass energy that is *not* handled by the VBMS *must* be managed by the final limiter. By this we mean that if the VBMS Freq is set to 80Hz then the advantages of VBMS will *only* extend below 80Hz! Bass energy *above* 80Hz will be (necessarily) managed by the look ahead limiter. This may or may not be desirable, depending on the type of program material you are processing, because under certain circumstances it may cause audible pumping.

The higher the VBMS Frequency setting the more energy is managed by the VBMS algorithm and the more active the VBMS activity indication will be. There is no right or wrong amount of VBMS processing — whatever sounds best for your format and market is correct.

Overdriving the VBMS will not cause obnoxious forms of distortion. This is because the VBMS algorithm is intelligent and “knows” about energy that is within the non-VBMS path, and it continuously adapts to minimize unintentional intermodulation.

Note also that there are up to *three* bands of multiband limiting operating within the VBMS bandwidth when it is set to 300Hz. Even at a “typical” VBMS setting of 180-200Hz there are still two full bands of the eight band limiter working on bass energy. No other on-air processor in this product class has such high precision control of low frequency energy.

In formats where the program content contains very heavy low bass, such as in R&B, Rap, and Dance, the VBMS Frequency should be tuned to between 120 and 200Hz for best on-air results (a good starting setting for most music formats is 180Hz). Other formats might sound best with the control set to higher frequencies. Classical and Jazz, for instance, might sound best when the VBMS frequency is set to 250Hz or higher. Talk formats might find that it sounds best with the VBMS Frequency set between 250Hz and the 300Hz maximum upper limit.

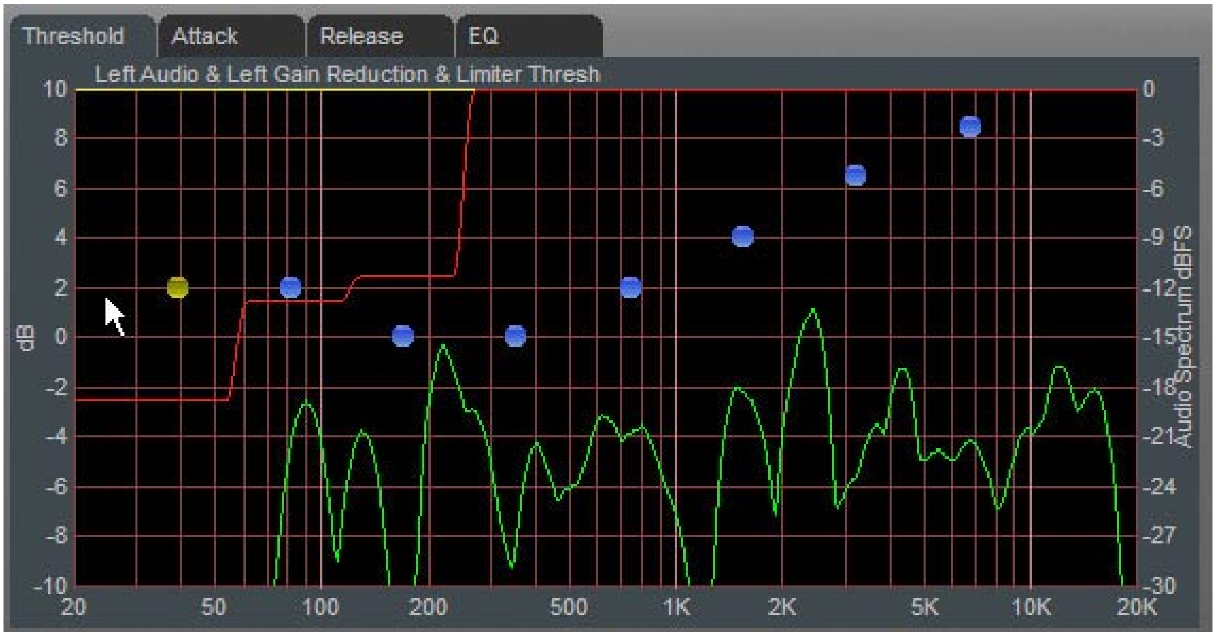
Because the VBMS is not a simplistic “bass clipper” like those found in other products it does not generate the kind of signals that would not interact well with a perceptual codec. Therefore the VBMS can be used to create bass impact in coded applications that would simply be impossible to do otherwise and without creating any codec-induced artifacts.

Note that although we provided a means to do so, we do not recommend operating the Multiband Limiter with the VBMS algorithms turned off!

## Eight Band 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.




**Eight Band Limiter Threshold Screen**

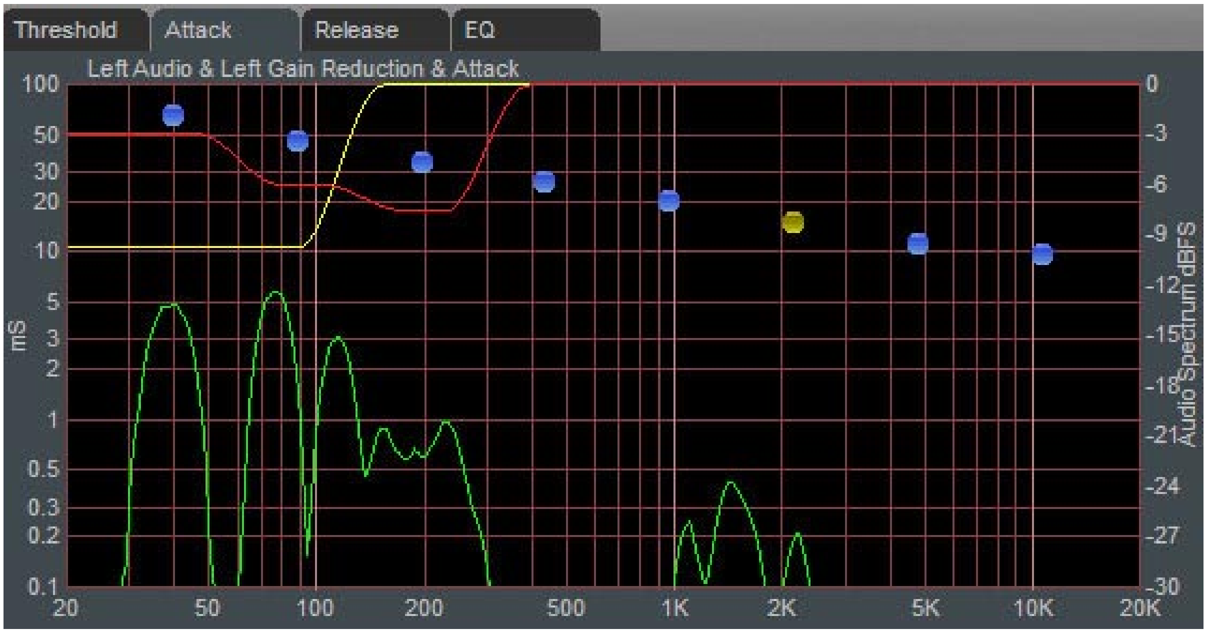
The limiter thresholds are adjustable over a +/-10dB range. This wide range permits frequency conscious limiting if needed.

Each of the limiter thresholds can be adjusted separately or in groups according to the manner described in the earlier VP-8 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.




**Eight Band 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.

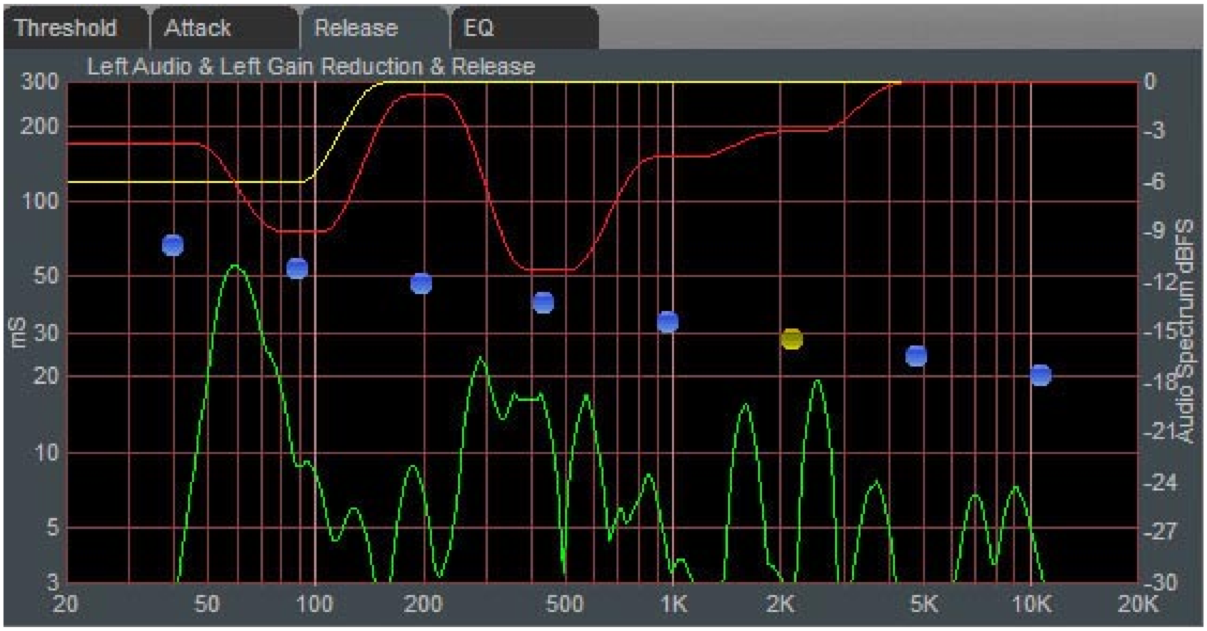
Generally, lower frequencies require slower attack times and higher frequencies require faster attack times for a given amount of control.

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.



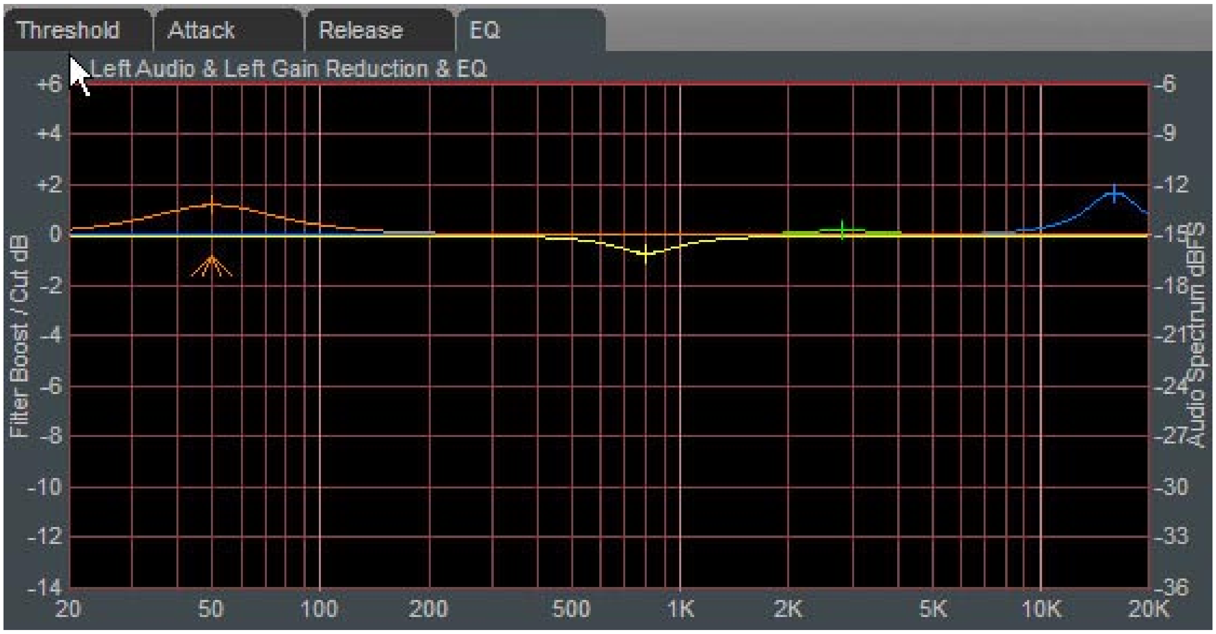

**Eight Band Limiter Release Time Screen**

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.




**Eight Band 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 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.

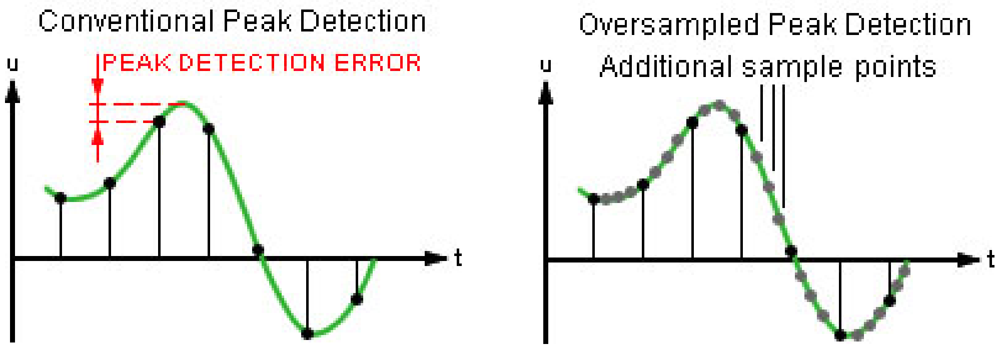
## FM-HD MENU


**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.

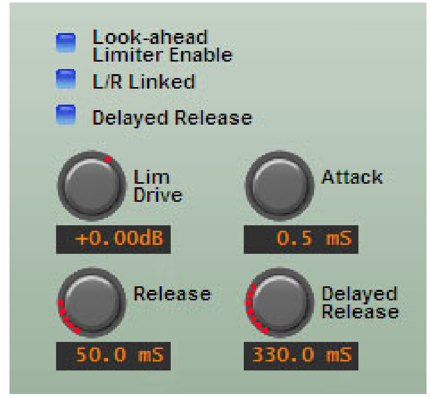
### FM-HD Look Ahead Limiter

The FM-HD output is accurately peak controlled by a highly oversampled feed forward limiter with a 0.5 millisecond (500 microseconds) lookahead 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.

### Limiter Enable

When this checkbox is checked the look ahead 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 Drive level adjustments.

### Attack

This control adjusts the attack time of the look ahead limiter. Because the "lookahead" 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 lookahead 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

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 checkbox is checked (see below).

When Delayed Release is engaged via the Delayed Release checkbox 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

When this checkbox is checked the electrical phase of the left and right signals are flipped 180 degrees.

### Left Right Reverse

When this checkbox 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-8, 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 Digital Output control can be set to OFF or adjusted over the range of minus 35.95dBFS to 0.00dBFS.

# VP-8 AM-HD Process Mode

The following discussion is unique to the Vorsis VP-8 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 listeners 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-8. These are the Limiters and AM-HD screens, the latter being visible and selectable via the AM-HD tab when VP-8 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**

### Eight Band Limiter

The VP-8 contains a Vorsis 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:

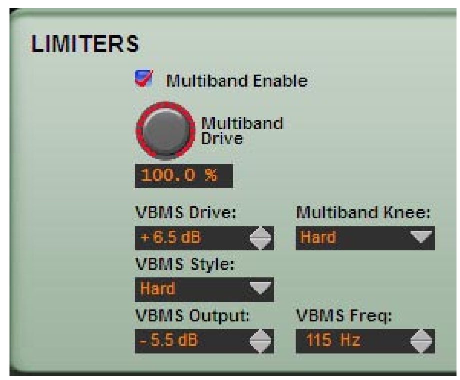
1. Extremely short attack times, while affording virtually perfect peak control, remove so much transient energy as to make the sound too homogenized and lifeless.

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.

Please open several of our Factory AM-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 Management System (VBMS) which is tightly integrated within the limiter's algorithms to intelligently control bass program energy.

## Multiband Enable




This checkbox 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.

## 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 Drive settings create a more gentle sound while still controlling peak energy, while higher 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.

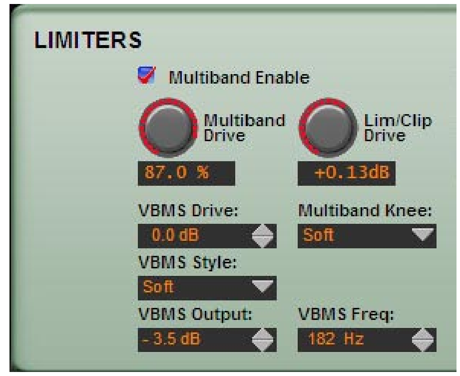
The absolute Drive control setting is dependent upon the setting of the Makeup control in the four band AGC's control screen. For this reason, the control is not labeled in "dB Drive," but instead is labeled in percent (%). With typical settings of the four band Makeup control (+24dB to +30dB) the correct eight band limiter Drive control setting will usually be between 85% and 95%. The full adjustment range is 0% to 100%.

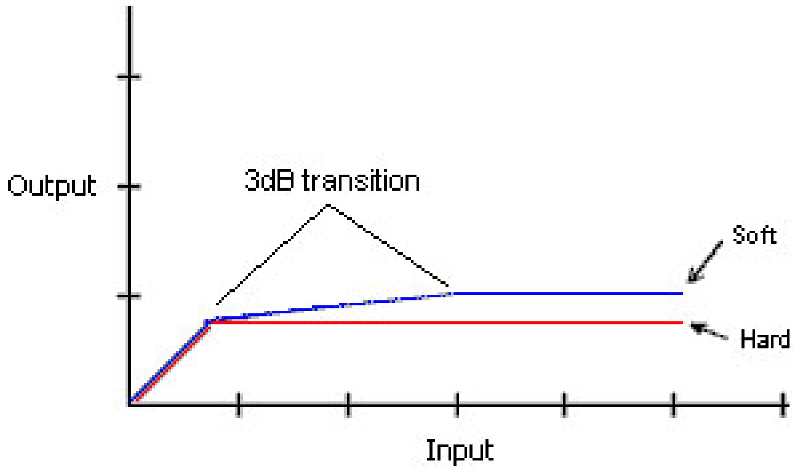
Higher Drive settings cause deeper gain reduction in the eight band limiter (as well as higher levels of limiting in the final look ahead limiter if the Lim/Clip Drive is not first reduced before the Multiband Drive is advanced). Therefore, Multiband Drive control is one of a pair of controls most useful 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 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 its behavior, the sound of the "Soft" function is a bit more obvious when in operation. Until the ultimate limit threshold (the setting of the band threshold) is finally reached it is also less precise as a true limiter.

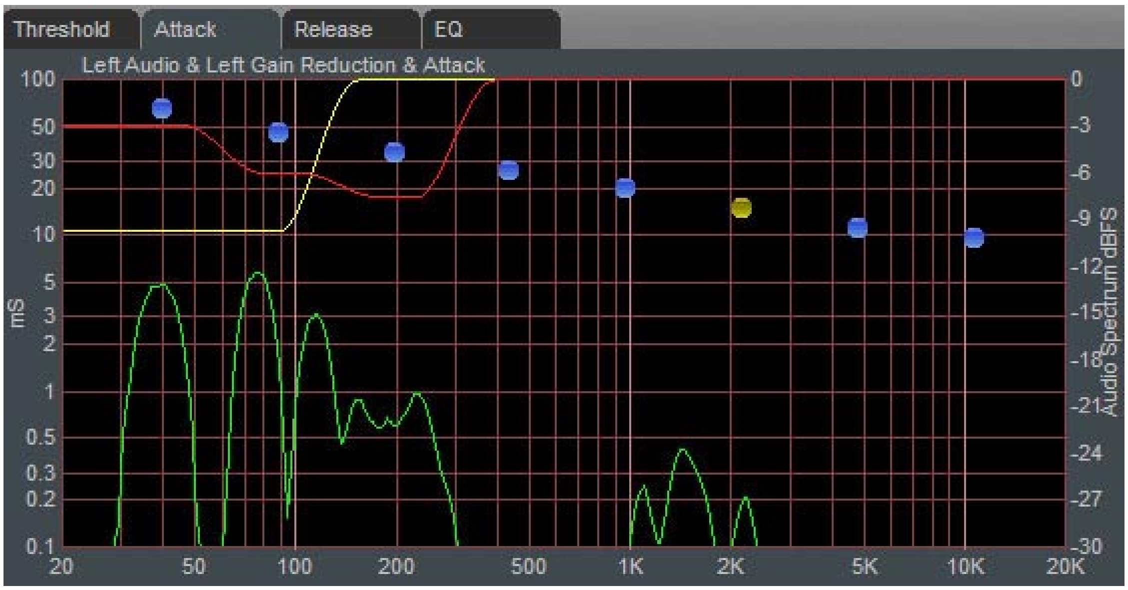
## Vorsis Bass Management System (VBMS)

The Vorsis Bass Management System processes low program frequencies in an artistic and natural program dependent manner without causing dynamic gain changes and undesirable intermodulation artifacts. Its primary task is to allow percussive instruments to sound natural while being well controlled in peak amplitude, while also controlling long-term bass energy to prevent intermodulation with higher frequency material within the main clipper.

The Vorsis Bass Management system uses special DSP algorithms that interoperate with the multiband limiter's lower frequency bands to control low frequency program energy. The algorithm is fully tunable, allows bass punch to be *precisely* set for any type of program material, and offers two operating modes plus an OFF setting, noting that we do not recommend turning the VBMS off!.

The VBMS offers a visual indication of program induced bass control which can be viewed in the limiter graphic as a solid yellow line extending from the VBMS Frequency setting downward to 20Hz, similar to that shown below.

VBMS Activity Region

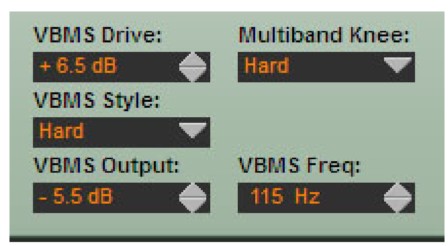



**VBMS Activity Display - 20Hz up to VBMS Frequency Setting**

Note that the lower frequency limit for VBMS operation is also a function of the High Pass Filter setting in the Input Screen!

### VBMS Drive

Adjusts the amount of drive to the VBMS algorithm and is adjustable over a range of +/- 10dB. Higher drive settings cause bass energy within the "VBMS Freq" bandwidth to spend more time under control of the VBMS, which increases the amount of low frequency loudness relative to higher frequency program material.

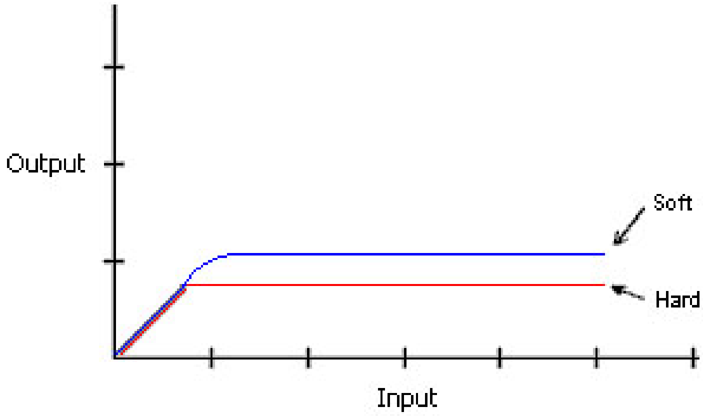



### VBMS Style

There are two VBMS operating styles available.

**Hard** - This transfer function is moderately abrupt and generates low order harmonics. Afterwards, and in a process that borrows from the technology of the Timbral mode in our Vorsis flagship processor, the AP-2000, we mathematically calculate and reduce significant harmonics above the fifth.

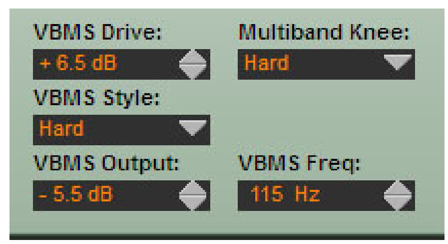
**Soft** - This transfer function is less abrupt than the hard setting, generating fewer higher order harmonics, and in this mode we calculate and reduce energy above the third harmonic.




**Comparison of VBMS Operating Modes**

The two VBMS styles may appear to be nearly identical due to the small difference in transfer functions. Rest assured, however, that there is a *significant* difference in the way the two styles sound on low frequency program material.

### VBMS Output




Adjusts the bass contribution from the output of the VBMS algorithm and is adjustable over a range of +/-10dB.

Generally speaking, the VBMS Drive and Output controls may be adjusted to personal taste and format requirements. Care should be exercised, though, in the setting of the VBMS Output control, because setting it higher than approximately -3dB *might* interfere with the main clipper distortion masking scheme when very heavy bass energy is present. Such interference could unmask undesirable distortion products on certain program material in quite strange and non-intuitive ways.

### VBMS Frequency

The VBMS operating bandwidth is adjustable from 60Hz to 300Hz. Lower frequency settings produce a "heavier" and "tighter" sound, while higher settings result in less coloration while still favorably controlling bass energy excursions. This control should be adjusted to personal taste and program format with one caveat:

Bass energy that is *not* handled by the VBMS *must* be managed by the final limiter. By this we mean that if the VBMS Freq is set to 80Hz then the advantages of VBMS will *only* extend below 80Hz! Bass energy *above* 80Hz will be (necessarily) managed by the look ahead limiter. This may or may not be desirable, depending on the type of program material you are processing, because under certain circumstances it may cause audible pumping.

The higher the VBMS Frequency is tuned the more energy is managed by the VBMS algorithm and therefore the more active the VBMS activity indication will be. There is no right or wrong amount of VBMS processing — whatever sounds best for the format and market is correct.

Overdriving the VBMS will not cause obnoxious forms of distortion. This is because the VBMS algorithm is intelligent and "knows" about energy that is within the non-VBMS path, and it continuously adapts to minimize unintentional intermodulation.

Note also that there are up to *three* bands of multiband limiting operating within the VBMS bandwidth when it is set to 300Hz. Even at a "typical" VBMS setting of 180-200Hz there are still two full bands of the eight band limiter working on bass energy. No other on-air processor has this level of precision on controlling low frequency energy.

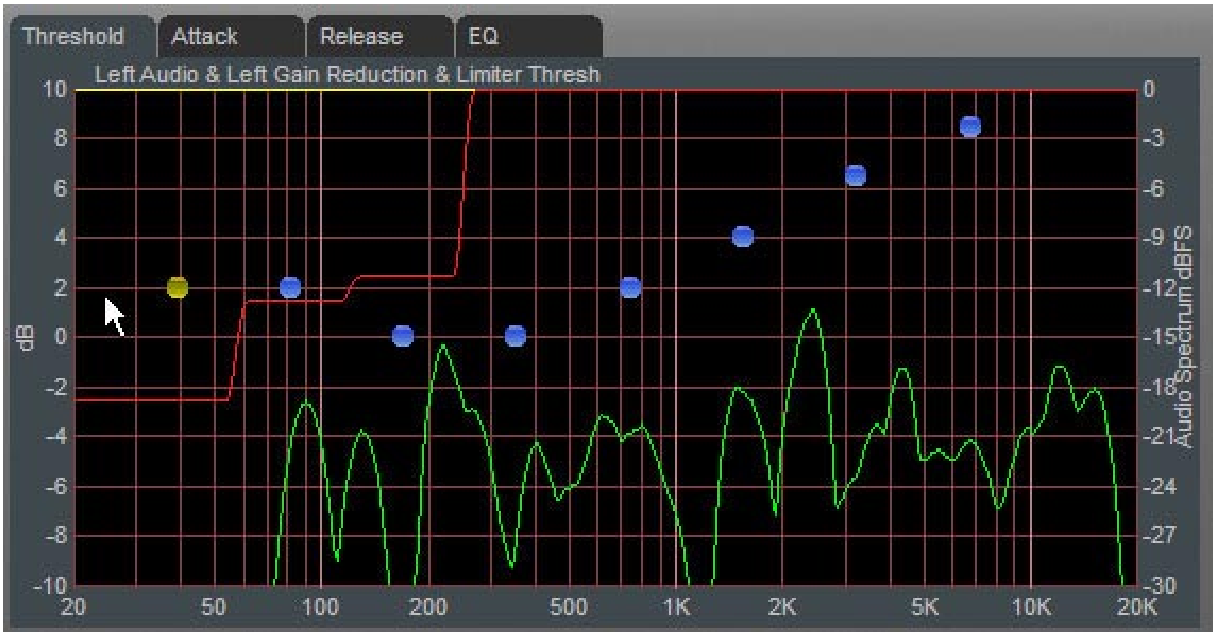
In formats where the program content contains very heavy low bass, such as in R&B, Rap, and Dance, the VBMS Frequency should be tuned to between 120 and 200Hz for best on-air results (a good starting setting for most music formats is 180Hz). Other formats might sound best with the control set to higher frequencies. Classical and Jazz, for instance, might sound best when the VBMS frequency is set to 250Hz or higher. Talk formats might find that it sounds best with the VBMS Frequency set between 250Hz and the 300Hz maximum upper limit.

Note that although we provided a means to do so, we do not recommend operating the Multiband Limiter with the VBMS algorithms turned off!

## Eight Band 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.




**Eight Band Limiter Threshold Screen**

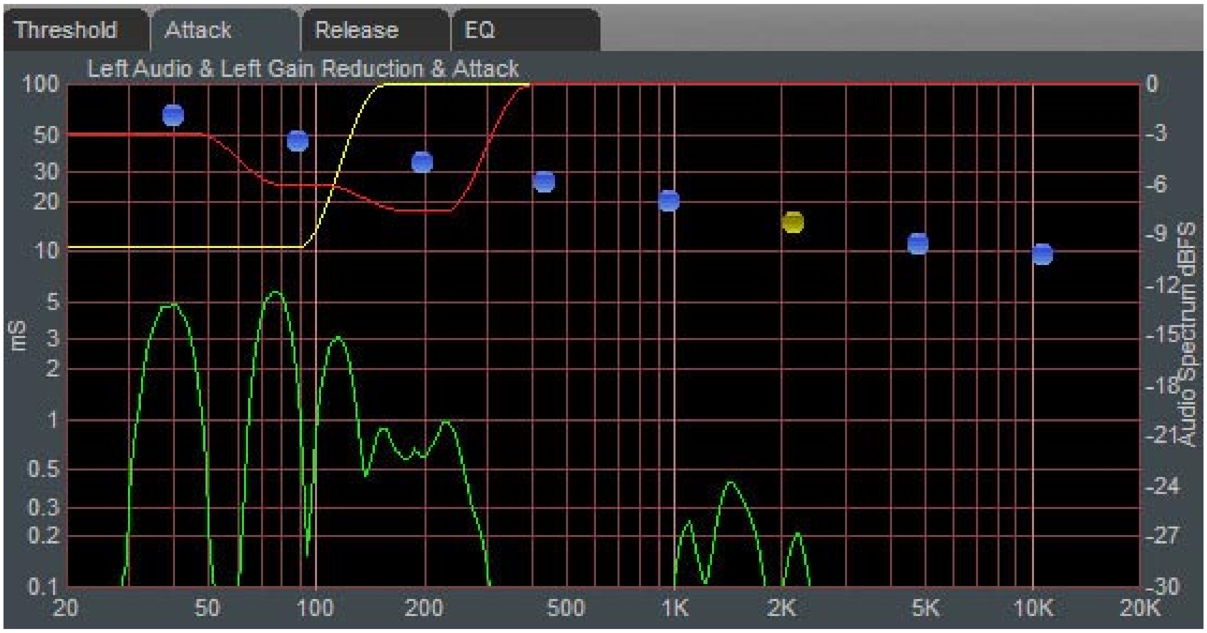
The limiter thresholds are adjustable over a +/-10dB range. This wide range permits frequency conscious limiting if needed.

Each of the limiter thresholds can be adjusted separately or in groups according to the manner described in the earlier VP-8 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.




**Eight Band 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.

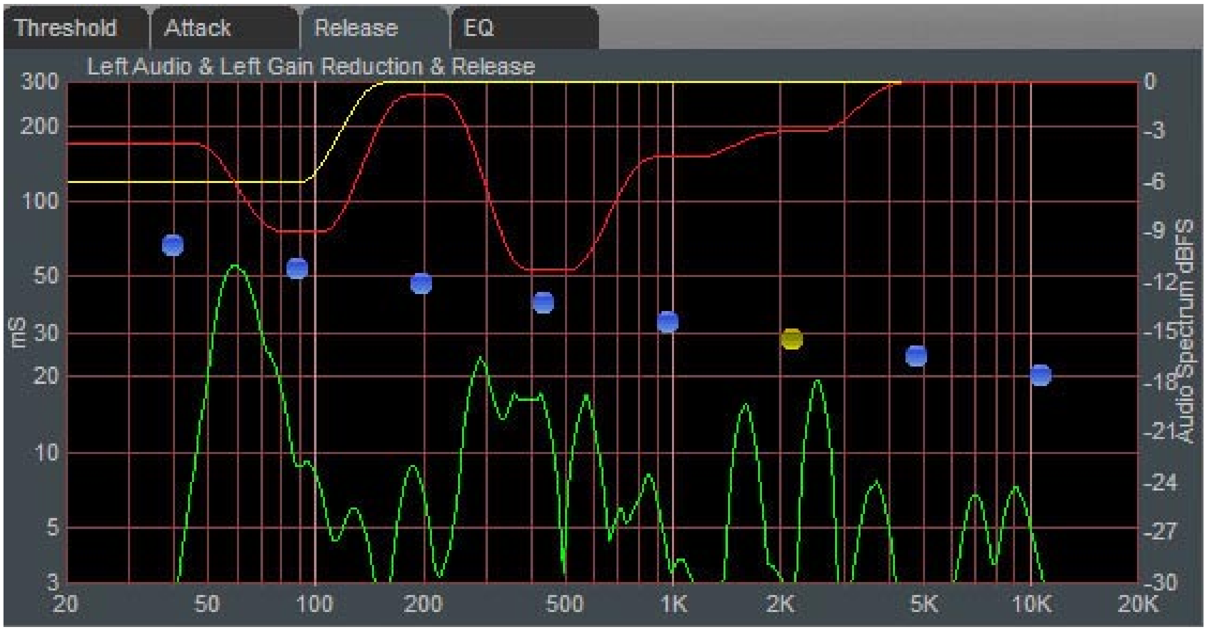
Attack times are generally adjusted to taste, keeping in mind that the slower the attack time the more work the limiter must do, while conversely the faster the attack time the fewer short term dynamics the program material will have.

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.



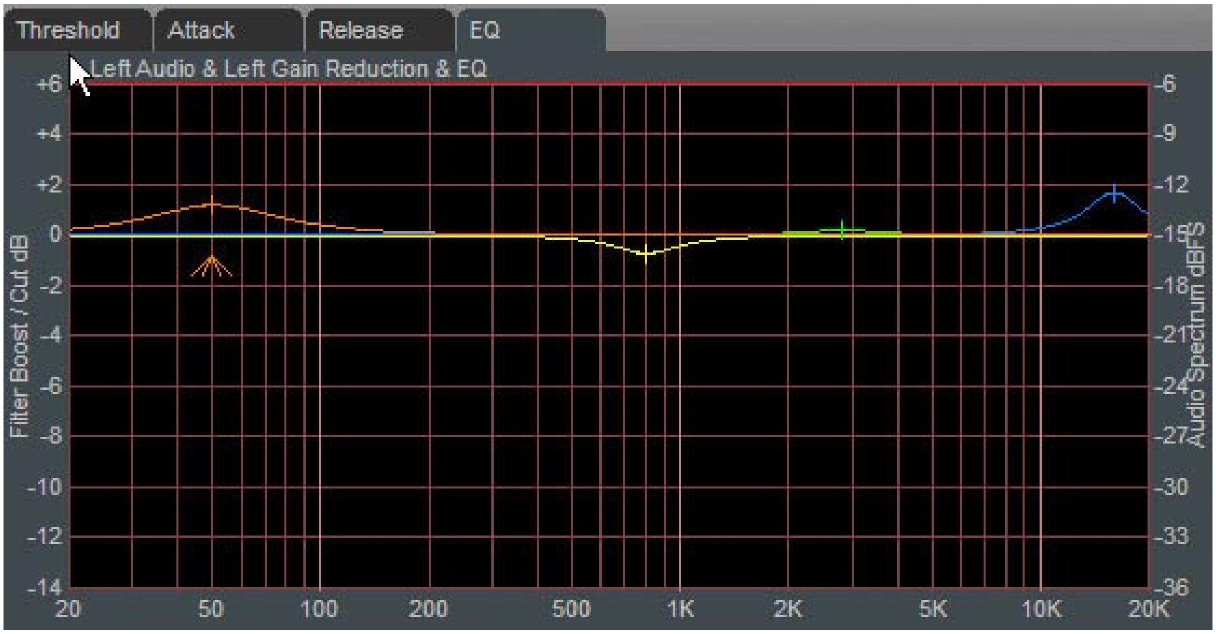

**Eight Band Limiter Release Time Screen**

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.




**Eight Band 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**

### 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.

### HF Protect Checkbox

When HF Protect is selected the VP-8 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 Freq. control adjusts the frequency where control action begins. This control works in concert with the Ceiling control (below).

### 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.

## 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 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 Checkbox

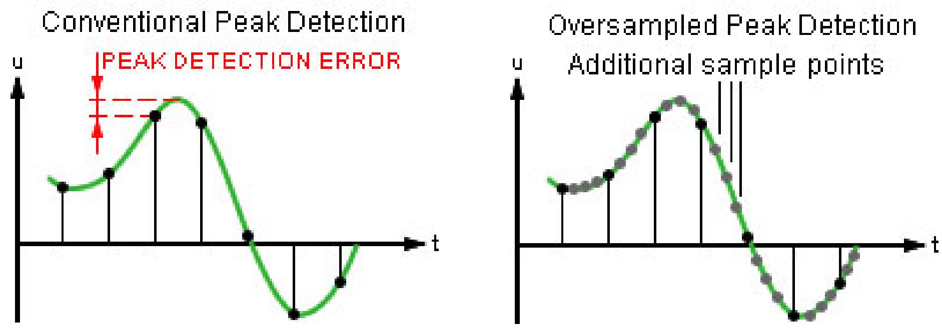
When this box is checked, weak or incorrect L-R information will be removed from the input audio, leaving a true monophonic signal where the left and right channels are absolutely identical.

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

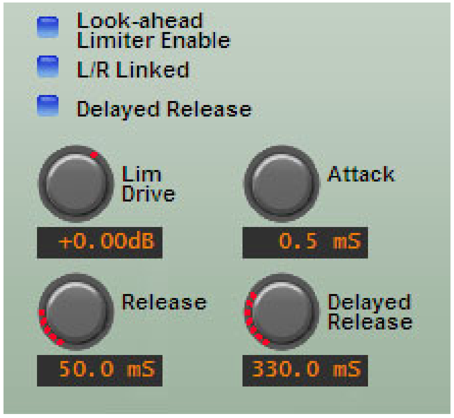
The AM-HD output is accurately peak controlled by a highly oversampled feed forward limiter with a 0.5 millisecond (500 microseconds) lookahead time. The control path (not the audio path) is highly oversampled in order to achieve extremely precise peak control. The primary goal of oversampling in the 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.




### Limiter Enable

When this checkbox 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

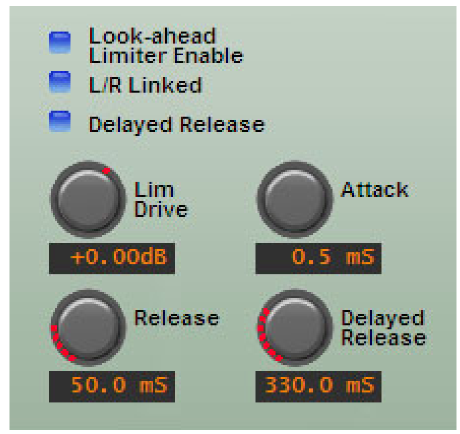
This controls the amount of drive to the look ahead limiter. The gain range of this control is +12dB/-18dB. The resolution of this control is 0.01dB, fine enough for the most exacting output level adjustments.

### Attack

This control adjusts the attack time of the look ahead limiter. Because the “lookahead” 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 “lookahead” 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 lookahead 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 checkbox is checked (see below).

When Delayed Release is engaged via the Delayed Release checkbox 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


When this checkbox 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-8 to that of a parallel, external signal path.

## Left Right Reverse

When this checkbox 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-8, 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 Digital 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-8'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.

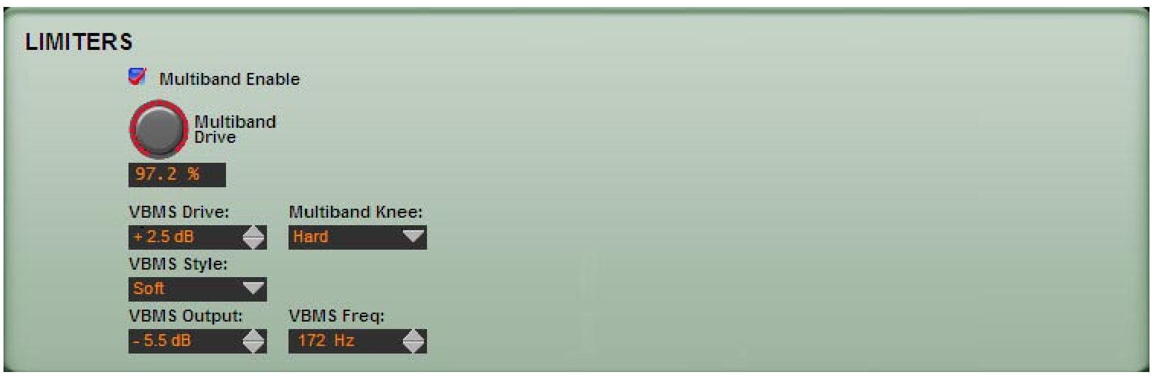
# VP-8 MP3/AAC>48K Process Mode

The following discussion is unique to the Vorsis VP-8 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-8 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**

### Eight Band Limiter

The VP-8 contains a Vorsis 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

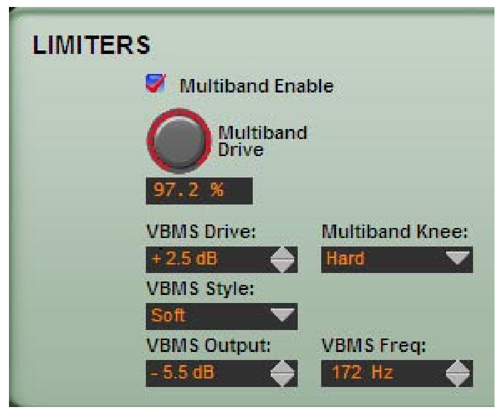
signal frequency and, not knowing that it is not "audio," code it along with the audio and reveal it as an unmasked artifact.

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 Management System (VBMS) which is tightly integrated within the limiter structure to intelligently control bass program energy.

## Multiband Enable

This checkbox 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 Drive settings create a more gentle sound while still controlling program energy, while higher 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 unnecessary.

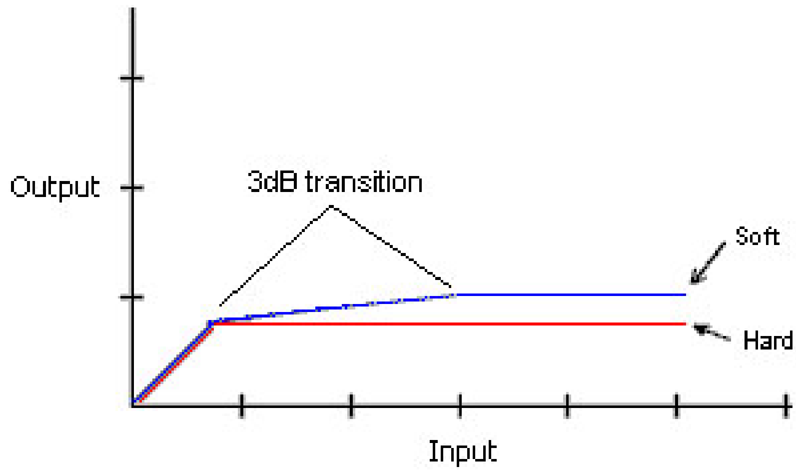
The absolute Drive control setting is also somewhat dependent upon the setting of the Makeup control in the four band AGC's control screen. For this reason, the control is not labeled in "dB Drive," but instead is labeled in percent (%). With typical settings of the four band AGC's Makeup control (+24dB to +30dB) the correct eight band limiter Drive control setting will usually be between 85% and 95%. The full adjustment range is 0% to 100%.

Higher 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.

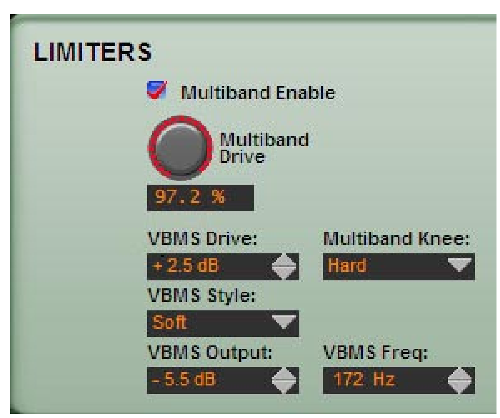



**Comparison of Hard and Soft Limiter Modes**

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.

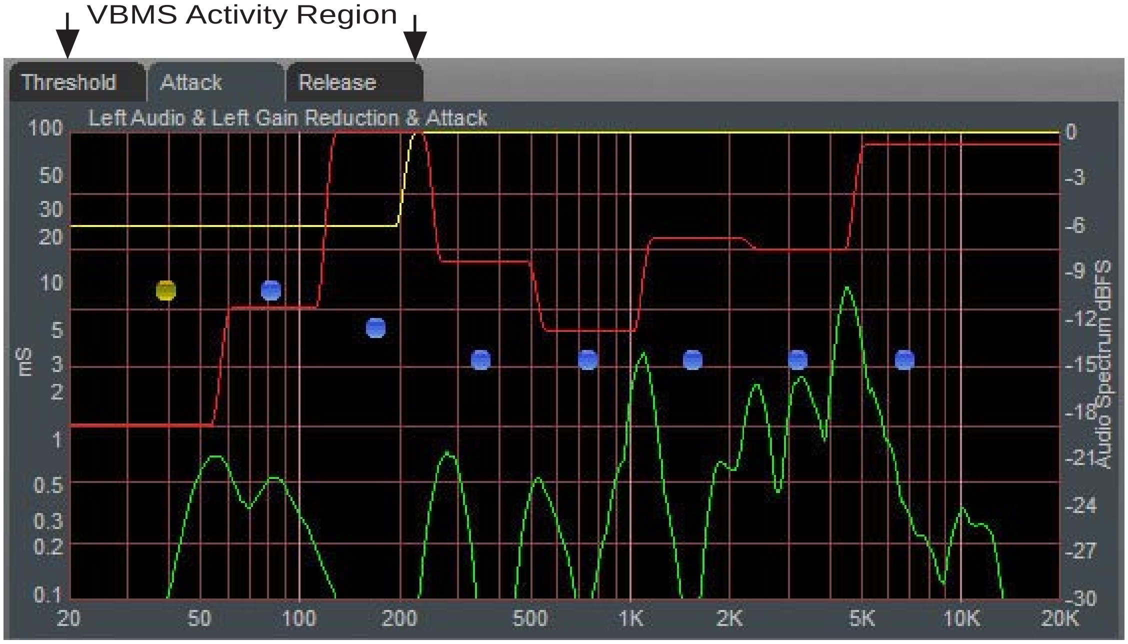
## Vorsis Bass Management System (VBMS)




The Vorsis Bass Management System processes low program frequencies in an artistic and natural program dependent manner without causing dynamic gain changes and undesirable intermodulation artifacts. Its sole task is to allow percussive instruments to sound natural while being controlled in peak amplitude, while simultaneously controlling long-term bass energy to prevent intermodulation with higher frequency material within the look ahead limiter.

The Vorsis Bass Management system uses special DSP algorithms that interoperate with several of the eight band limiter’s lower frequency bands to control low frequency program energy. The algorithm is fully tunable, allows bass punch to be *precisely* set for any type of program material, and offers two operating modes plus an OFF setting, noting that we generally do not recommend turning the VBMS off!

The VBMS algorithm offers a visual indication of when it is active and can be seen in the limiter graphic as a solid yellow line extending from the VBMS Frequency setting downward to 20Hz, similar to that shown below.

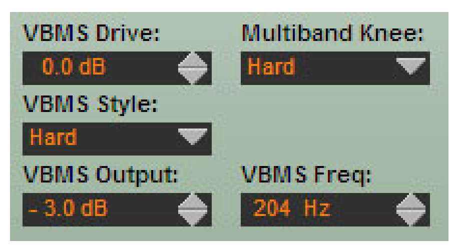



**VBMS Activity Display - 20Hz up to VBMS Frequency Setting**

Note that the lower frequency limit for VBMS operation is also a function of the High Pass Filter setting in the Input Screen! This is because the two acting in concert simulate a very low frequency bandpass filter with asymmetrical skirts.

### VBMS Drive

Adjusts the amount of drive to the VBMS algorithm and is adjustable over a range of +/- 10dB. Higher drive settings cause bass energy that is within the “VBMS Freq” bandwidth to spend more time under control of the VBMS. This increases low frequency loudness relative to higher frequency program material.




### VBMS Style

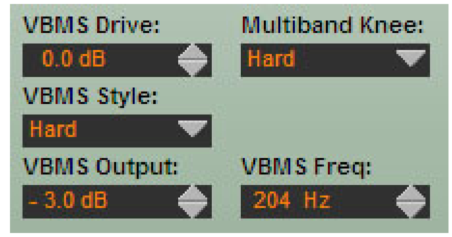
There are two operating styles available.

**Hard** - This transfer function is moderately abrupt and generates low order harmonics. Afterwards, and in a process that borrows from the technology of the Timbral mode in our Vorsis AP-2000 flagship processor, we mathematically calculate and reduce significant harmonics above the fifth.

**Soft** - This transfer function is less abrupt than the hard setting, generating fewer higher order harmonics, and in this mode we calculate and reduce energy above the third harmonic.

The two VBMS styles may appear to be nearly identical due to the small difference in transfer functions. Rest assured that there is a *significant* difference in the way the two styles sound on low frequency program material.

### VBMS Output




Adjusts the bass contribution from the output of the VBMS algorithm and is adjustable over a range of +/-10dB.

Generally speaking, the VBMS Drive and Output controls may be adjusted to personal taste and format requirements. Care should be exercised in the setting the VBMS Output control, because setting it higher than approximately -5dB *might* cause bass energy to interfere with the main limiter's distortion marking scheme when very heavy bass energy is present. Such interference could unmask undesirable distortion products on certain program material in quite peculiar and non-intuitive ways.

### VBMS Frequency

The VBMS operating bandwidth is adjustable from 60Hz to 300Hz. Lower frequency settings produce a "heavier" and "tighter" sound, while higher settings result in less bass coloration while still favorably controlling bass energy excursions. This control should be adjusted to personal taste and program format with one caveat:

Bass energy that is *not* handled by the VBMS *must* be managed by the final limiter. By this we mean that if the VBMS Freq is set to 80Hz then the advantages of VBMS will *only* extend below 80Hz! Bass energy *above* 80Hz will be (necessarily) managed by the look ahead limiter. This may or may not be desirable, depending on the type of program material you are processing, because under certain circumstances it may cause audible pumping.

The higher the VBMS Frequency setting the more energy is managed by the VBMS algorithm and the more active the VBMS activity indication will be. There is no right or wrong amount of VBMS processing — whatever sounds best for your format and market is correct.

Overdriving the VBMS will not cause obnoxious forms of distortion. This is because the VBMS algorithm is intelligent and "knows" about energy that is within the non-VBMS path, and it continuously adapts to minimize unintentional intermodulation.

Note also that there are up to *three* bands of multiband limiting operating within the VBMS bandwidth when it is set to 300Hz. Even at a "typical" VBMS setting of 180-200Hz there are still two full bands of the eight band limiter working on bass energy. No other processor in this product class has such high precision control of low frequency energy.

In formats where the program content contains very heavy low bass, such as in R&B, Rap, and Dance, the VBMS Frequency should be tuned to between 118 and 200Hz for best on-air results (a good starting setting for most music formats is 180Hz). Other formats sound best with the control set to higher frequencies. Classical and Jazz, for instance, sound best when the VBMS frequency is set to 250Hz or higher. Talk formats sounds best with the VBMS Frequency set between 250Hz and the 300Hz maximum upper limit.

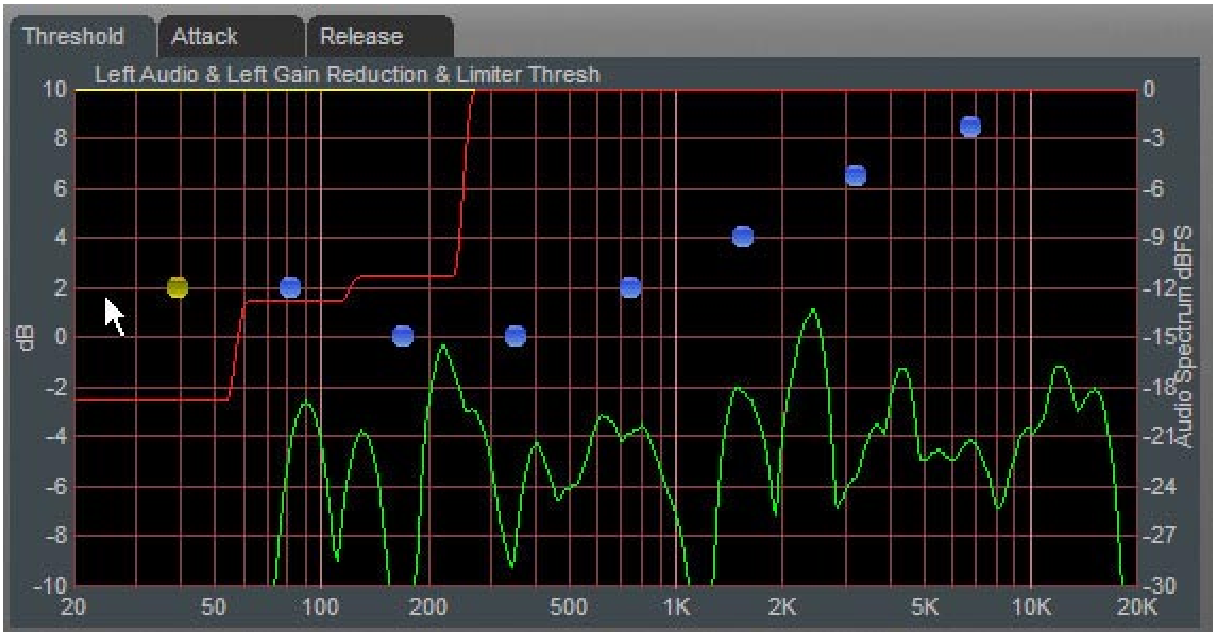
Because the VBMS is not a simplistic "bass clipper" it does not generate signals that would cause a perceptual codec to misbehave. Therefore the VBMS can be used to create bass impact in coded applications that would simply be impossible to do otherwise and without creating any codec-induced artifacts.

Note that although we provided a means to do so, we do not recommend operating the Multiband Limiter with the VBMS algorithms turned off!

## Eight Band 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.




**Eight Band Limiter Threshold Screen**

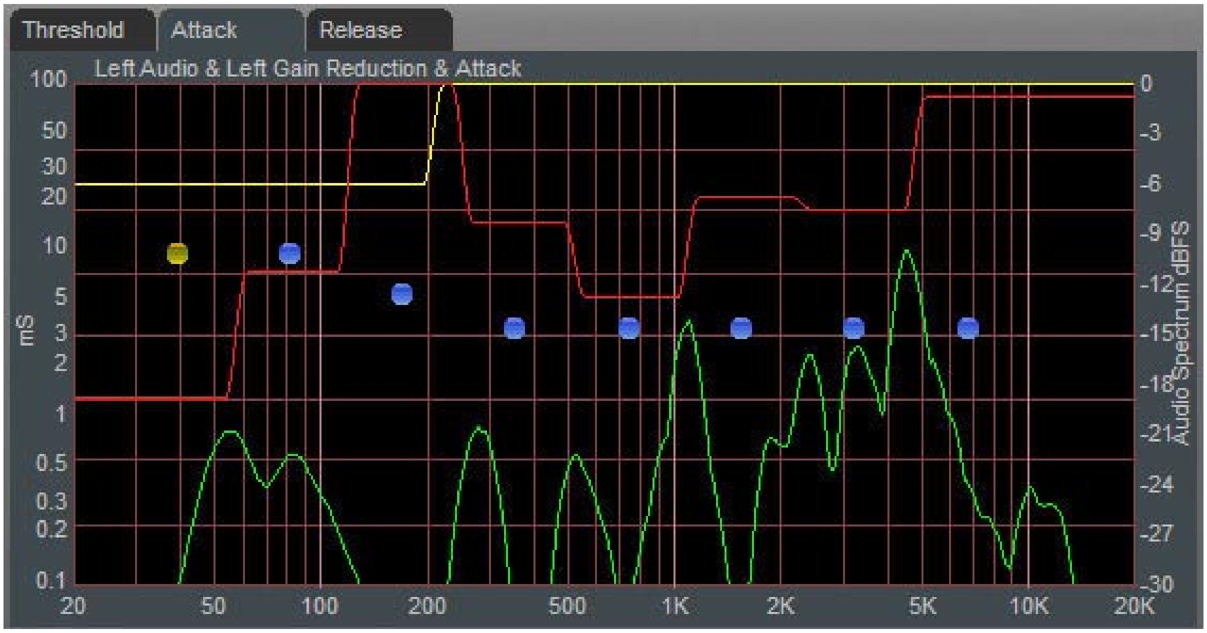
The limiter thresholds are adjustable over a +/-10dB range. The thresholds are adjustable over a wide range to permit frequency conscious limiting to be created if needed.

Each of the limiter thresholds can be adjusted separately or in groups according to the manner described in the earlier VP-8 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.




**Eight Band 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.

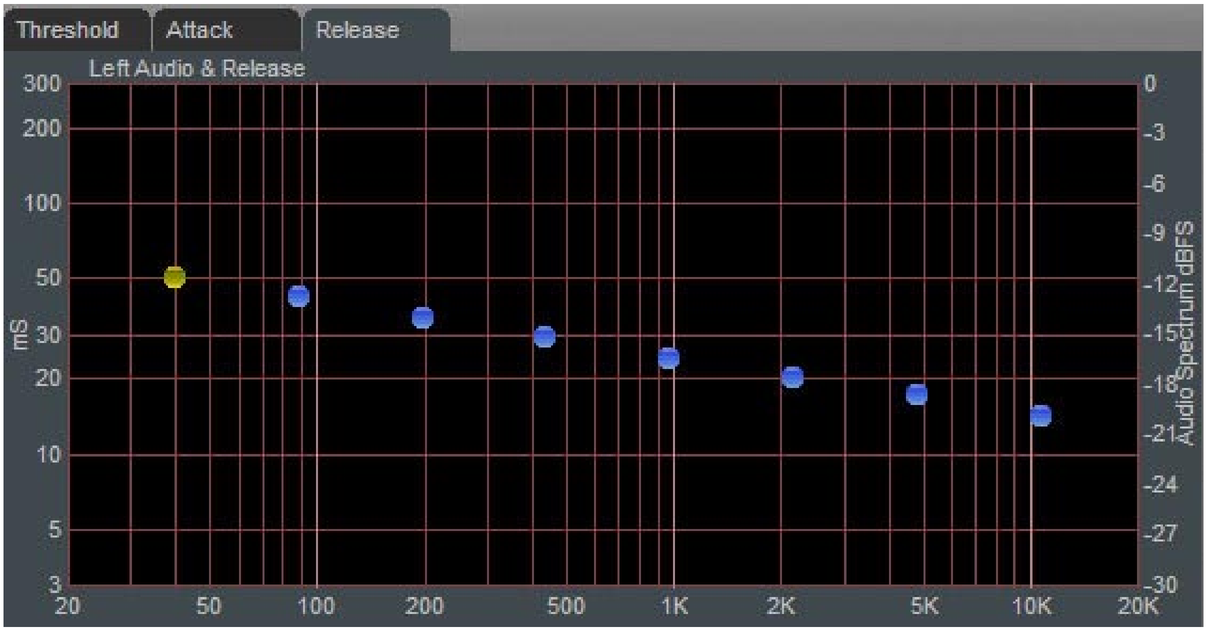
Attack times are generally adjusted to taste, keeping in mind that the slower the attack time the more work the final limiter must do. Conversely, the faster the attack time the fewer short term dynamics the program material will have.

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 3, page 3-15).

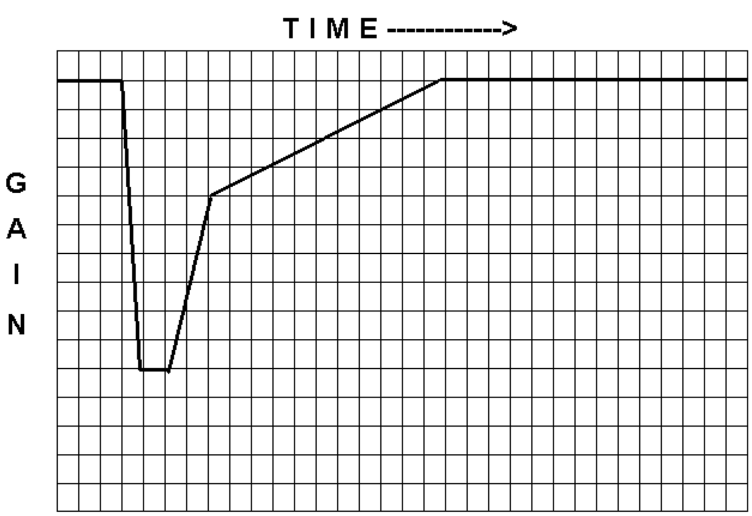
## Release Time Screen

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




**Eight Band Limiter Release Time Screen**

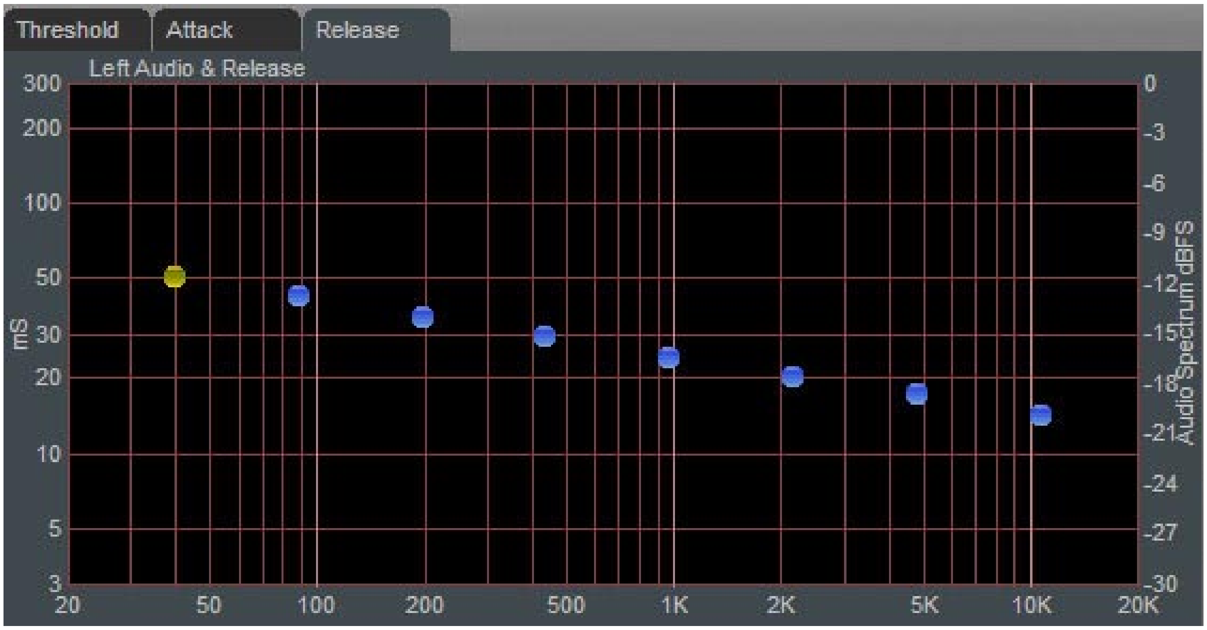
The release time of a limiter determines how fast it recovers from an audio peak once the peak has caused a momentary decrease in gain. Shorter (faster) release times create a denser sound, and excessively short release times might increase intermodulation distortion (the mixing of low and high frequency energy — or a muddying of the sound). The multiband limiters in the VP-8 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 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.




**<u>Release Time Note:</u>**

The eight band limiter's Release times may be adjusted all the way down to 3 milliseconds, which is far *<u>too fast</u>* 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-8'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 <u>create an effect</u>, it must be done with great care, and it is therefore <u>not</u> 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-8 is equipped with a specially designed Vorsis algorithm that accomplishes exactly this.

The VP-8'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-8'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-8 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-8 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-8+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-8'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-8's lower bitrate processing mode requires.

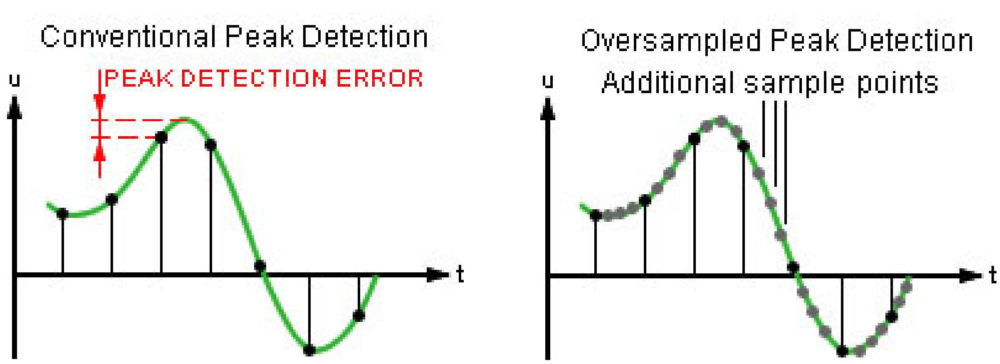
The MP3/AAC>48K screen is available for adjusting the characteristics of the Look Ahead limiter and Output signals,

### 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.

### Limiter Enable

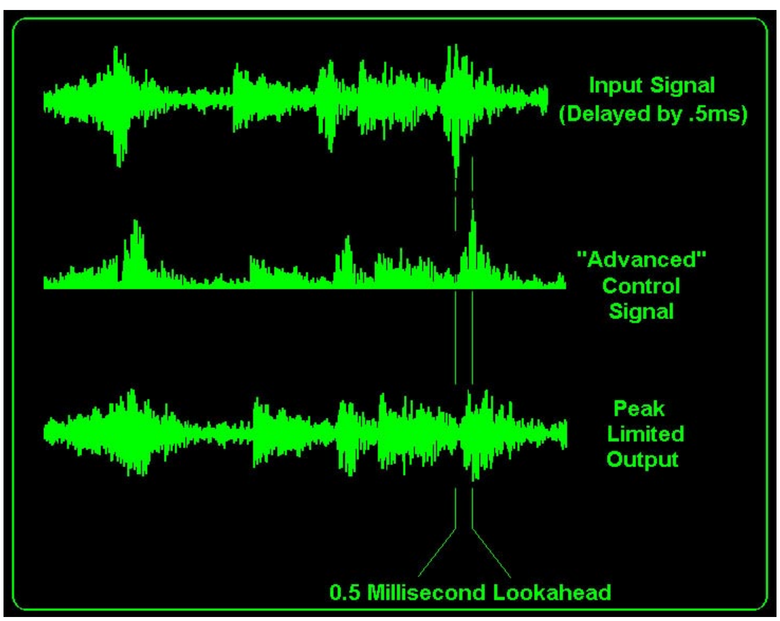
When this checkbox is checked the look head limiter is enabled. When the box is not checked no peak limiting is performed and the peak output levels will be uncontrolled and therefore undefined!

### Lim (Limiter) Drive

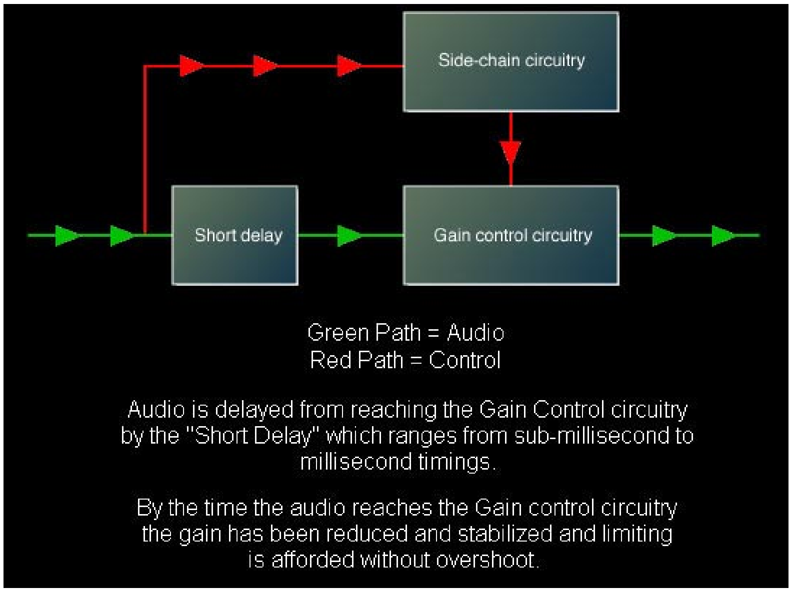
This controls the amount of drive to the look ahead limiter, noting that, unlike other VP-8 operating modes there are no clippers available - all limiting must be done by the look ahead limiter. The gain range of this control is -18dB to+12dB. The resolution of this control is 0.05dB, fine enough for 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-8 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-8.
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-8 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-8 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 checkbox 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-8 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 checkbox the electrical phase of the left and right signals are in phase with the signal applied to the input of the VP-8. 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-8. To change this setting merely click on it and its status will toggle.

## Left Right Reverse


When this checkbox 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-8, 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 35.95 dBFS to 0 dBFS.

**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.

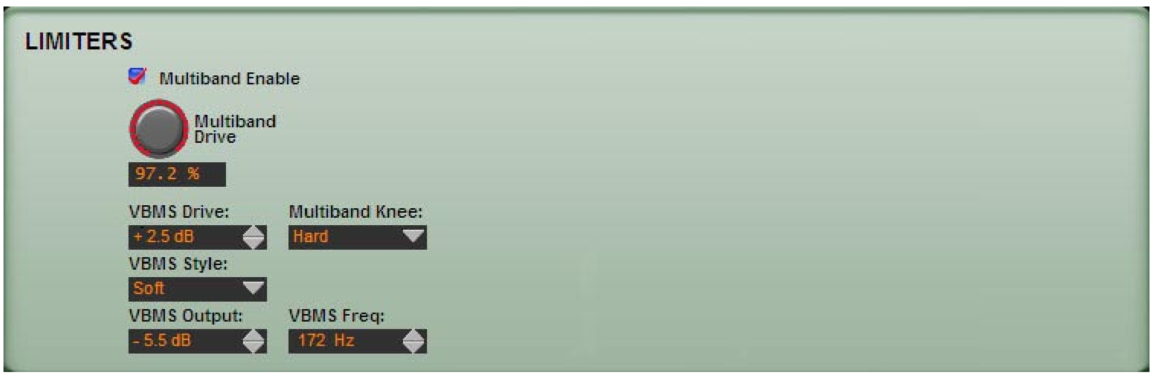
# VP-8 MP3/AAC<48K Process Mode

The following discussion is unique to the Vorsis VP-8 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-8 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**

### Eight Band Limiter

The VP-8 contains a Vorsis 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:

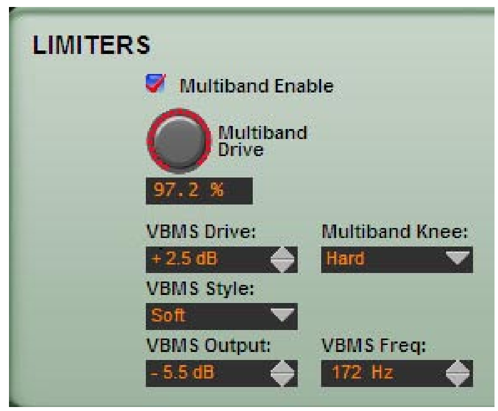
1. Extremely short attack times, while affording virtually perfect energy control, remove so much transient energy that it makes the sound too homogenized and lifeless.
2. Even though our limiters utilize a delayed release algorithm, extremely short release times may increase certain coding artifacts. 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 Management System (VBMS), which is tightly integrated within the limiter structure to intelligently control bass program energy. Please see the section of the VP-8 manual dedicated to GUI operation for how to accomplish the adjustment of the individual operating parameters of the eight-band section.

## Multiband Enable




This checkbox 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 Drive settings create a more gentle sound while still controlling program energy, while higher 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.

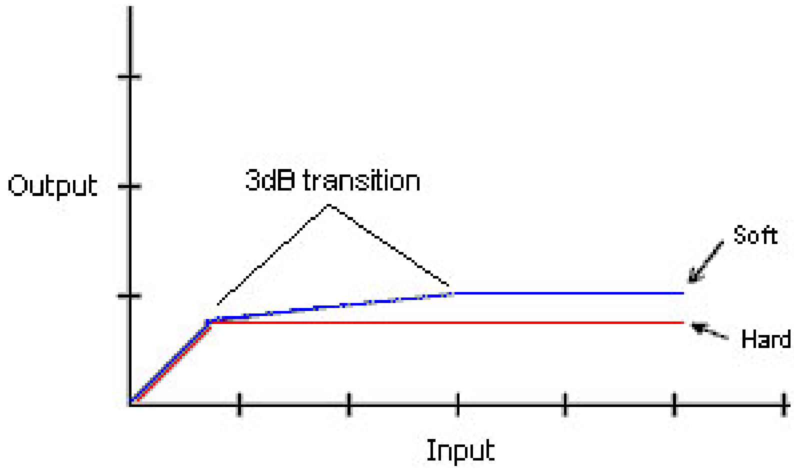
The Multiband Drive control setting is also somewhat dependent upon the setting of the Makeup control in the four band AGC's control screen. For this reason, the Multiband Drive control is not labeled in "dB Drive," but instead is labeled in percent (%). With typical settings of the four band AGC's Makeup control (+20dB to +30dB) the correct eight band limiter 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**

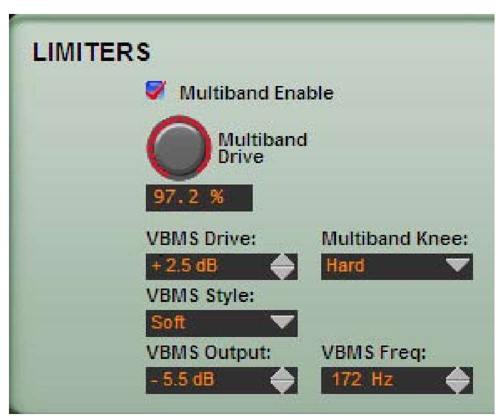
In the "**Soft**" mode, limiting starts to occur at a level that is 3dB below the user-set limiter threshold and the transfer function is relatively soft. That is, the audio waveform is not abruptly limited, but instead is allowed to gradually increase over a 3dB window until the ultimate limit threshold 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.

## Vorsis Bass Management System (VBMS)

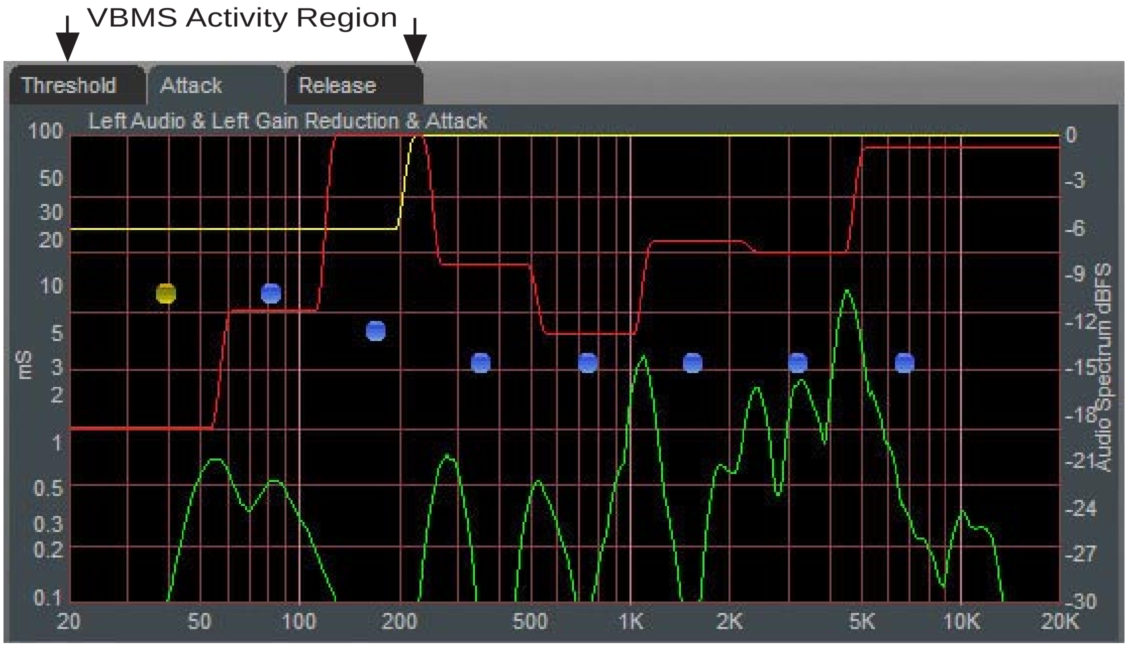
The Vorsis Bass Management System processes low program frequencies in an artistic and natural program dependent manner without causing dynamic gain changes and undesirable intermodulation artifacts. Its two tasks are to:

- allow percussive instruments to sound natural while being controlled in peak amplitude, and
- control long-term bass energy in a manner that serves to prevent intermodulation with higher frequency material within the look ahead limiter.




The Vorsis Bass Management system uses special DSP algorithms that interoperate with several of the eight band limiter's lower frequency bands to control low frequency program energy. The algorithm is fully tunable, and allows bass punch to be *precisely* set for any type of program material. It offers two operating modes plus an OFF setting, noting that we generally do not recommend turning the VBMS off!

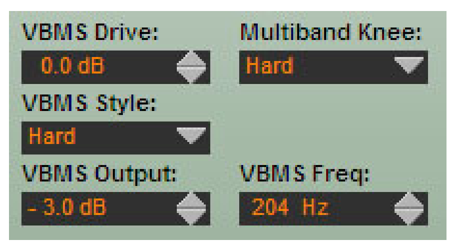
The VBMS algorithm offers a visual indication of when it is active and can be seen in the limiter graphic as a solid yellow line extending from the VBMS Frequency setting downward to 20Hz, similar to that shown below.




**VBMS Activity Display - 20Hz up to VBMS Frequency Setting**

Note that the lower frequency limit for VBMS operation is also a function of the High Pass Filter setting in the Input Screen because the two combine to create a very low frequency bandpass filter with asymmetrical skirts.

### VBMS Drive




Adjusts the amount of drive to the VBMS algorithm and is adjustable over a range of +/- 10dB. Higher drive settings cause bass energy within the "VBMS Freq" bandwidth to spend more time under control of the VBMS. This increases the amount of low frequency loudness relative to higher frequency program material.

### VBMS Style

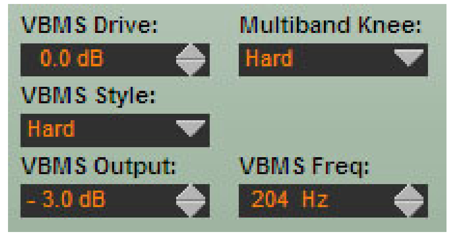
There are two operating styles available.

**Hard** - This transfer function is moderately abrupt and generates low order harmonics. Afterwards, and in a process that borrows from the technology of the Timbral mode in our Vorsis AP-2000 flagship processor, we mathematically calculate and reduce significant harmonics above the fifth.

**Soft** - This transfer function is less abrupt than the hard setting, generating fewer higher order harmonics, and in this mode we calculate and reduce energy above the third harmonic.

The two VBMS styles may appear to be nearly identical due to the small difference in transfer functions. Rest assured, however, that there is a *significant* difference in the way the two styles sound on low frequency program material.

### VBMS Output




Adjusts the bass contribution from the output of the VBMS algorithm and is adjustable over a range of +/-10dB.

Generally speaking, the VBMS Drive and Output controls may be adjusted to personal taste and format requirements. Care should be exercised in the setting the VBMS Output control, because setting it higher than approximately -5dB *might* cause bass energy to interfere with the main limiter when very heavy bass energy is present. Such interference can cause distortion on certain material in peculiar and non-intuitive ways.

### VBMS Frequency

The VBMS operating bandwidth is adjustable from 60Hz to 300Hz. Lower frequency settings produce a “heavier” and “tighter” sound - higher settings have less bass coloration while still controlling bass energy. This control should be adjusted to personal taste and program format with one caveat:

Bass energy *not* handled by the VBMS *must* be managed by the final limiter. By this we mean that if the VBMS Freq is set to 80Hz then the advantages of VBMS will *only* extend below 80Hz! Bass energy *above* 80Hz will be (necessarily) managed by the look ahead limiter, and under certain circumstances it may cause audible pumping.

The higher the VBMS Frequency setting the more low frequency energy is managed by the VBMS algorithm and therefore the more active the VBMS activity indication will be. There is no right or wrong amount of VBMS processing — whatever settings sound best are correct.

Overdriving the VBMS will not cause obnoxious forms of distortion because the VBMS algorithm is intelligent and “knows” about energy that is within the non-VBMS path, and continuously adapts to minimize unintentional intermodulation distortion.

Note also that in the VP-8 there can be as many as *three* bands of multiband limiting operating within the VBMS bandwidth when the VBMS Freq. control is set to 300Hz. Even at a “typical” VBMS setting of 180-200Hz there are still two full bands of the eight band limiter working on bass energy. No other on-air processor in this product class has such high precision control of low frequency energy.

When program content has very heavy low bass, such as in R&B, Rap, and Dance, the VBMS works best with the Frequency set between 118 and 210Hz - a good setting for most music formats is 180Hz. Formats such as Classical and Jazz sound best when the VBMS frequency is set to 250Hz, and Talk formats sounds best with the VBMS Frequency set near the 300Hz maximum upper limit.

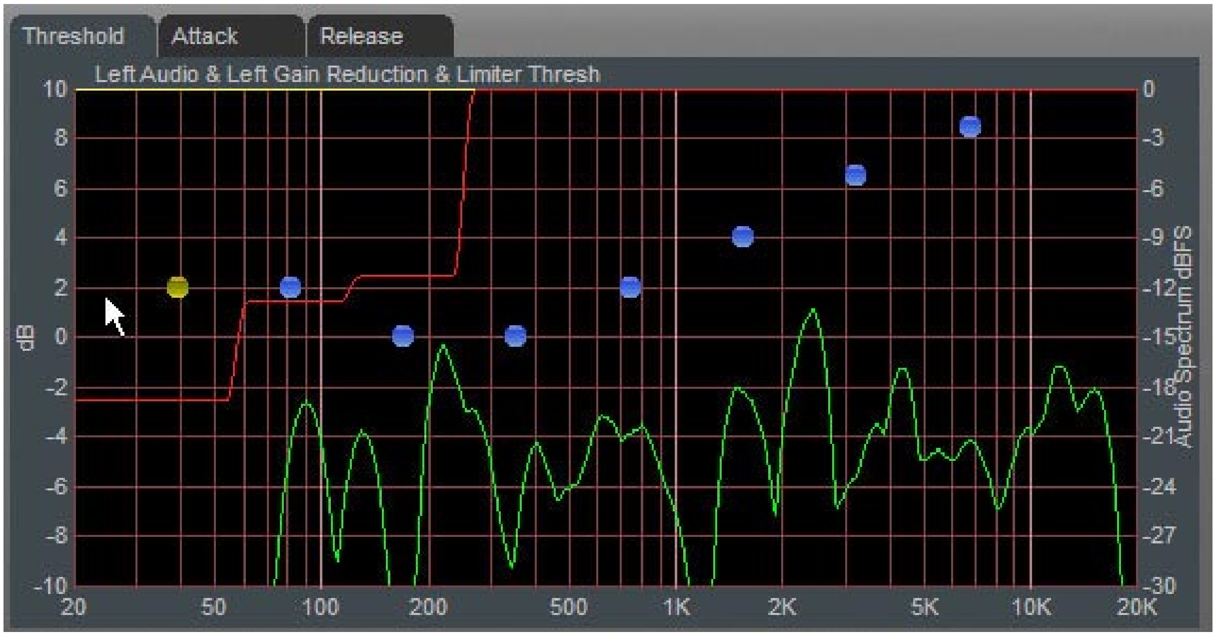
Because the VBMS is not a simplistic “bass clipper” it does not generate signals that will tease a perceptual codec. Therefore the VBMS can be used to create bass impact in coded applications that would simply be impossible to do otherwise without creating codec-induced artifacts.

Note that although we provided a means to do so, we do not recommend operating the Multiband Limiter with the VBMS algorithms turned off!

## Eight Band 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.




**Eight Band Limiter Threshold Screen**

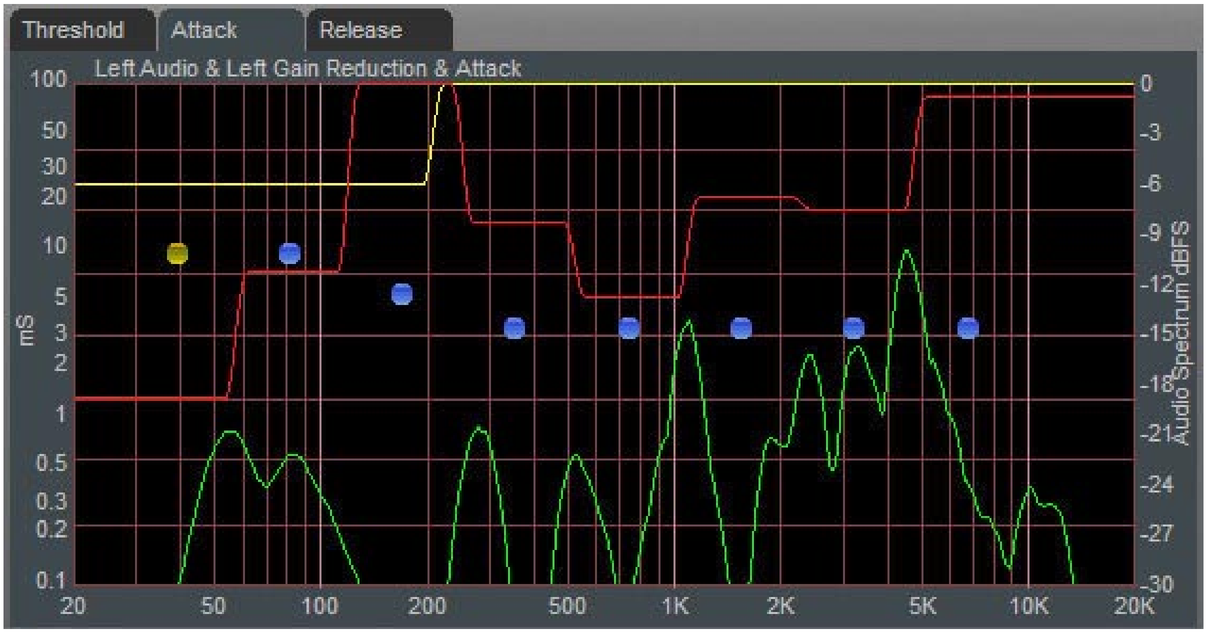
The limiter thresholds are adjustable over a +/-10dB range permitting frequency conscious limiting to be created if needed.

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.




**Eight Band 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.

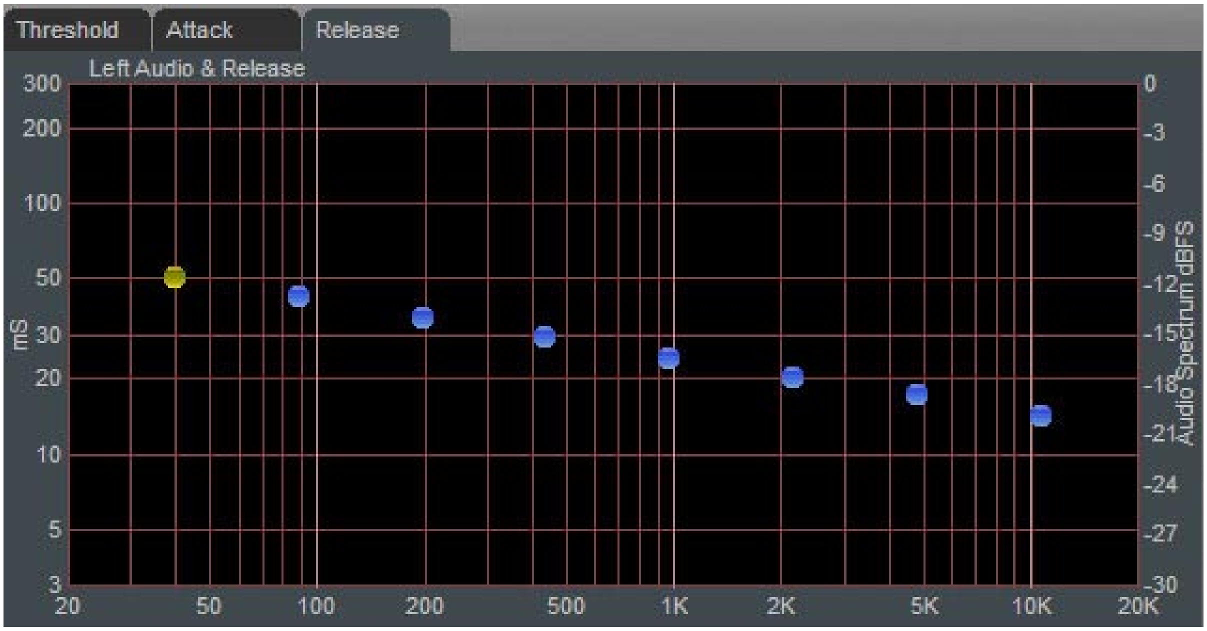
Attack times are generally adjusted to taste, keeping in mind that the slower the attack time the more work the final limiter must do. Conversely, the faster the attack time the fewer short term dynamics the program material will have.

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 3, page 3-15).

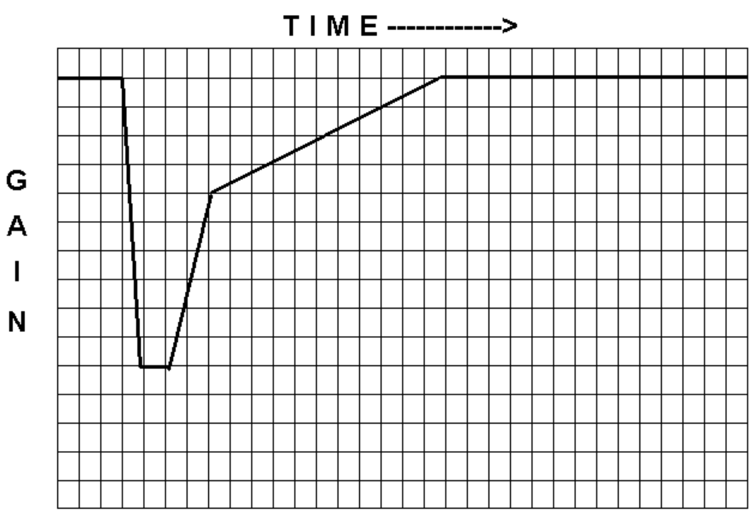
## Release Time Screen

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




**Eight Band Limiter Release Time Screen**

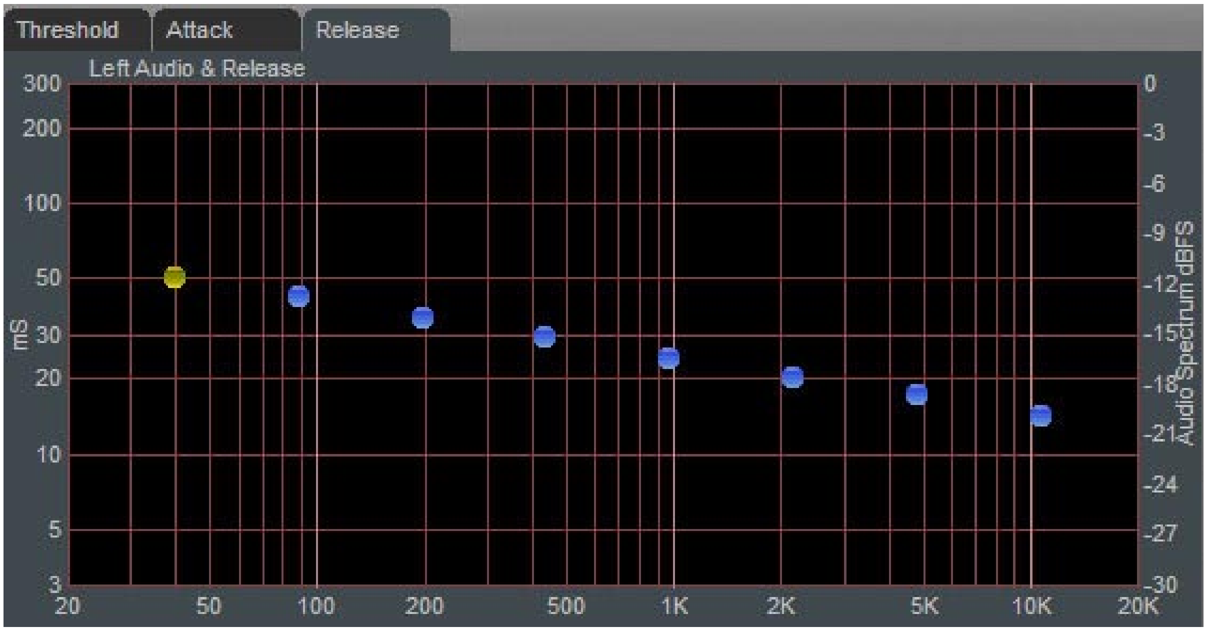
The release time of a limiter determines how fast it recovers from an audio peak after the peak has caused a momentary decrease in gain. Shorter (faster) release times create a denser sound, and excessively short release times might increase intermodulation distortion (the mixing of low and high frequency energy — or a muddying of the sound). The multiband limiters in the VP-8 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 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.

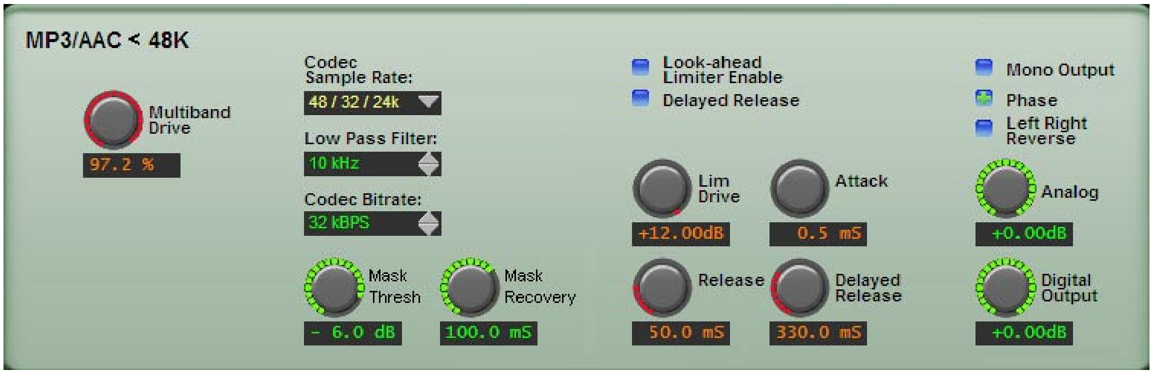
In effect, such drastic settings create harmonic and narrow-band IM distortion. But because the resulting distortion is constrained to the quite narrow frequency bands of the VP-8’s multiband limiters it is possible to create aesthetically pleasing “warmth” on the low end without blatant and irritating 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-8 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-8'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 Control Screen and though it is multiband in structure it is completely independent of the multiband AGC and eight band limiter.

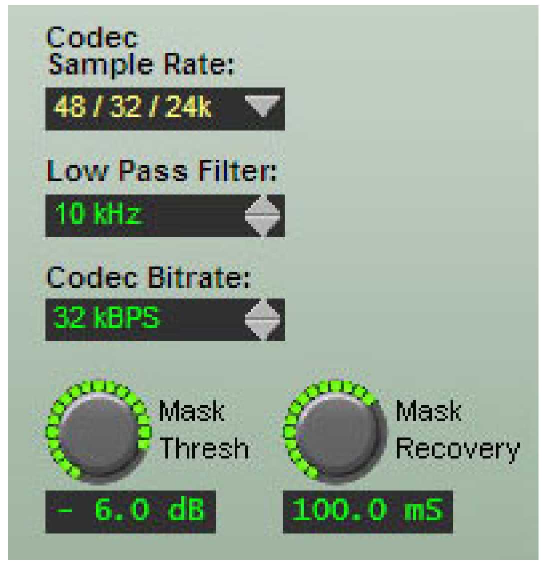
The purpose of the Spectral Energy Gate™ is to find, through a series of look ahead calculations, opportunities for lowering the priority of certain audio information that, because of its spectral or temporal characteristics would contribute minimal improvement during reconstruction of the signal at 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-8 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, 48/32/24kHz and 44.1/22.05kHz which correspond to different multiband operating parameters within the Adaptive Bandwidth Controller™.

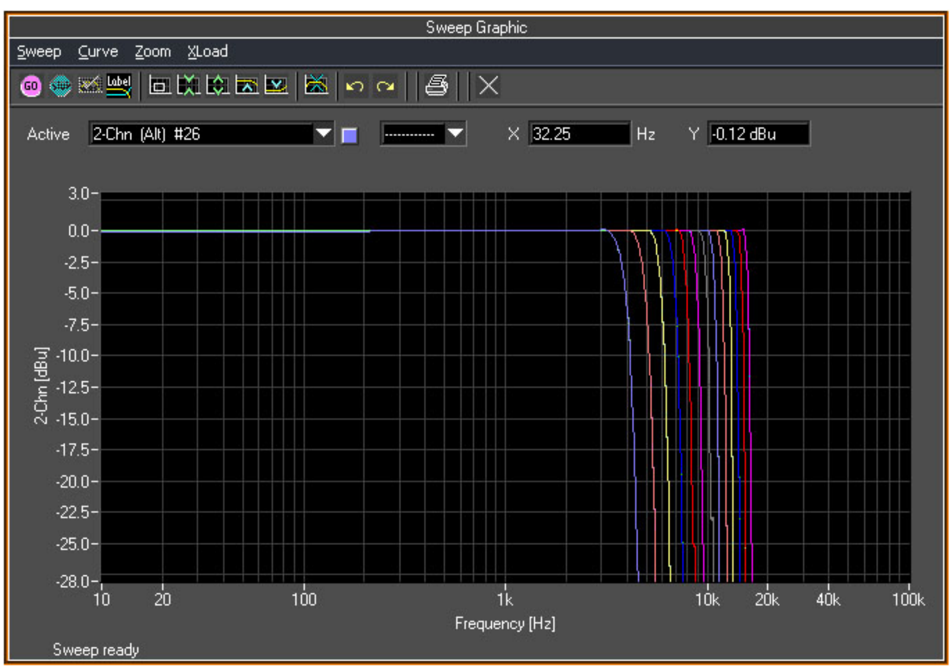
The differences between the two settings serve important roles in achieving the best sound from codecs at different sample rates. We recommend utilizing the 48/32/24 kHz or 44.1/22.05kHz position for codecs operating at those sample rates. Sample rates lower than 22.05 kHz may benefit most from the use of the 44.1/22.05kHz setting.

### Low Pass Filter

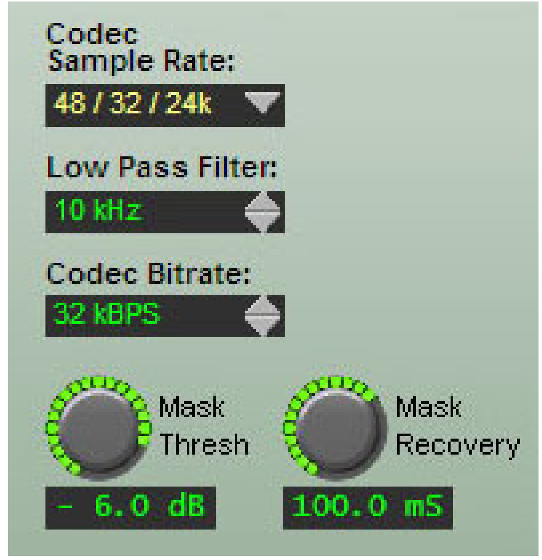
The purpose of the low pass filter is to remove program energy higher than the frequency selected by the control. By removing audio energy above a certain frequency, higher subjective audio quality can be achieved by the codec when operating at a particular bitrate. Low pass filters become especially 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 32 kHz 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 Threshold 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-8.

### Achieving the Proper Settings

There is naturally some interaction between the Mask Threshold 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 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 Bitrate control is set to 32kBPS and the 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 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 Bitrate, Mask Threshold and 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-8'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-8 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-8 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-8 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.

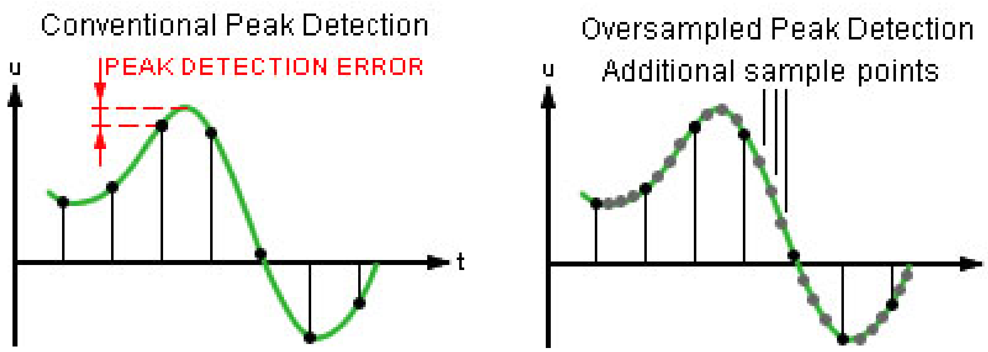
**Our specialized Vorsis algorithms can greatly <u>improve</u> the sound quality of low bitrate audio.**

**The VP-8'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.


### Limiter Enable

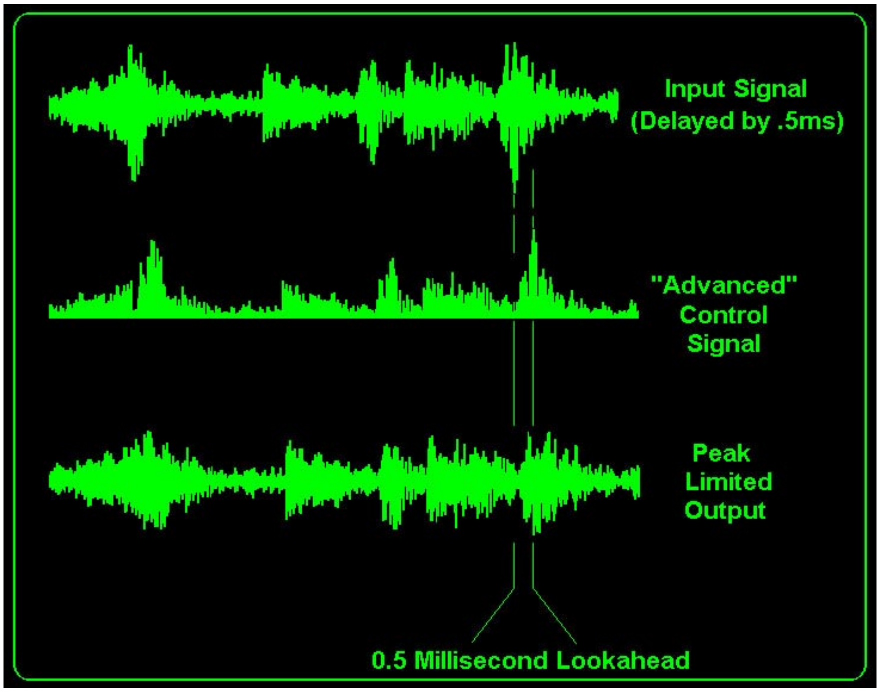
When this checkbox is checked the look head limiter is enabled. When the box is not checked no peak limiting is performed and the peak output levels will be uncontrolled and therefore undefined!

### Lim (Limiter) Drive

This controls the amount of drive to the look ahead limiter, nothing that, unlike other VP-8 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.

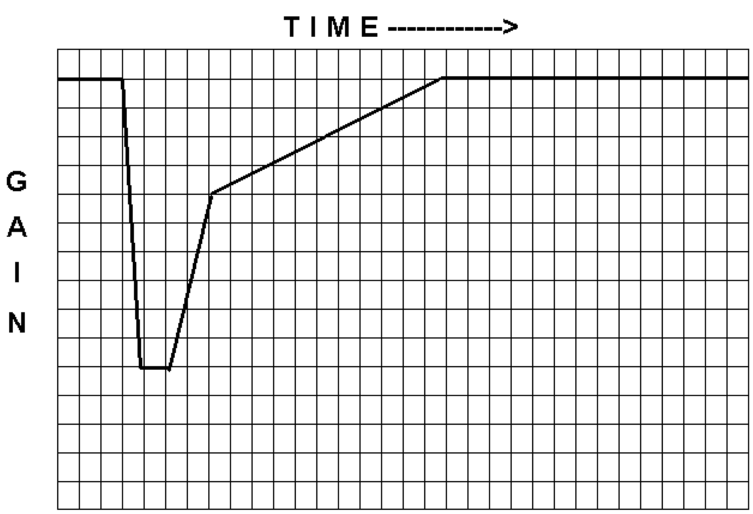
Because the look ahead limiter is the only mechanism by which peak levels can be accurately controlled, if the attack time of this limiter is slowed down (set to larger numbers) in order to create a sound that is more open and less restrained then due consideration must be given to the resulting peak 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-8.
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-8 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-8 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 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 checkbox is checked.

When Delayed Release is engaged via the Delayed Release checkbox 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-8 is mixed to Mono - the left and right channels carry identical information, <u>even if the input is in stereo</u>. This control is useful for forcing mono when desired.

### Phase

When a Plus (+) shows in this checkbox the electrical phase of the left and right signals are in phase with the signal applied to the input of the VP-8. 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-8. To change this setting merely click on it and its status will toggle.

## Left Right Reverse

When this checkbox 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-8, 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 35.95 dBFS to 0 dBFS.

**Peak Output Level Note:**

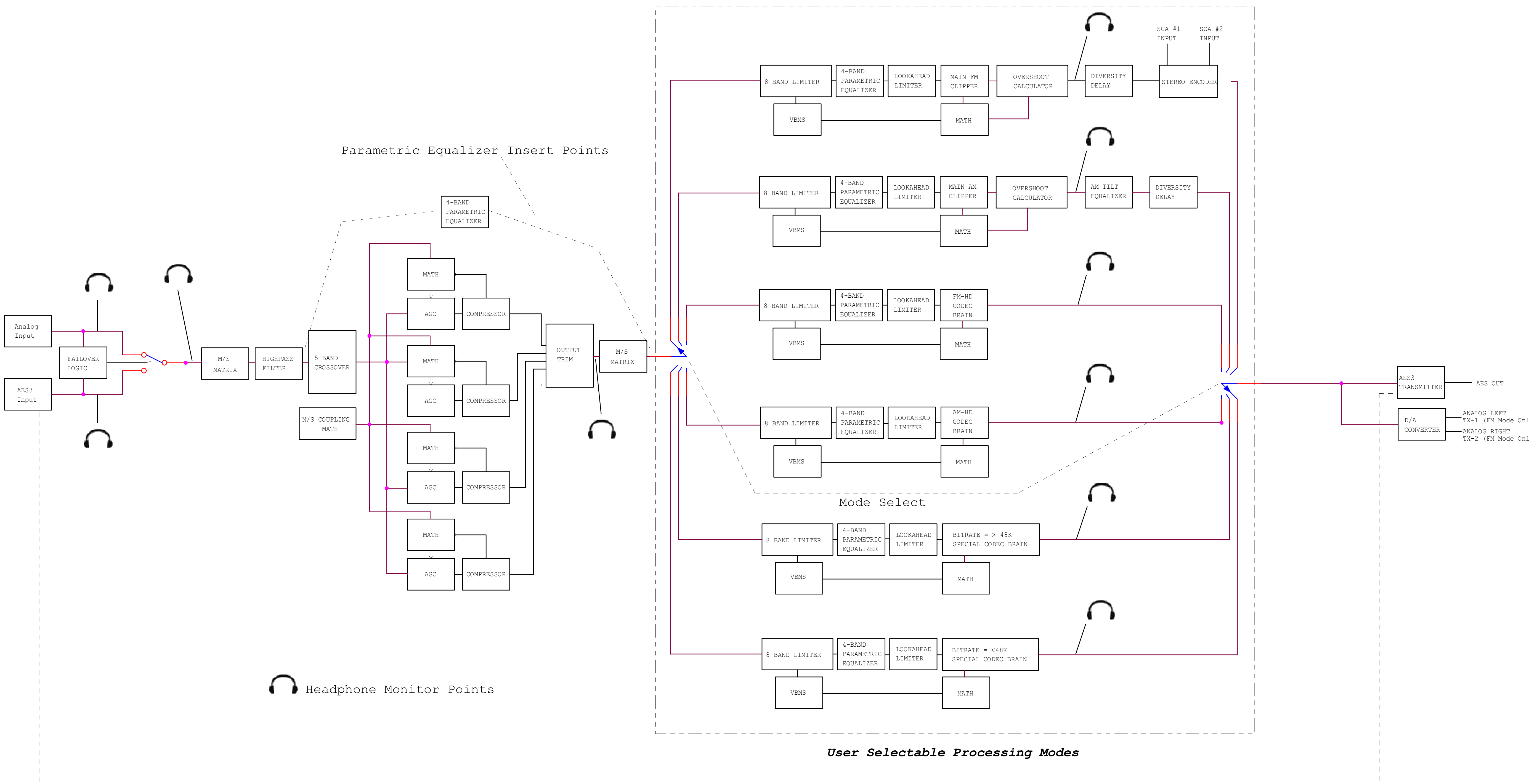
As mentioned previously, the "peak level" accuracy, that is, the calibration, of the output level controls is highly dependent on the setting of the Attack time of the Look Ahead Limiter.

# I/O Schematic Drawings & Load Sheet

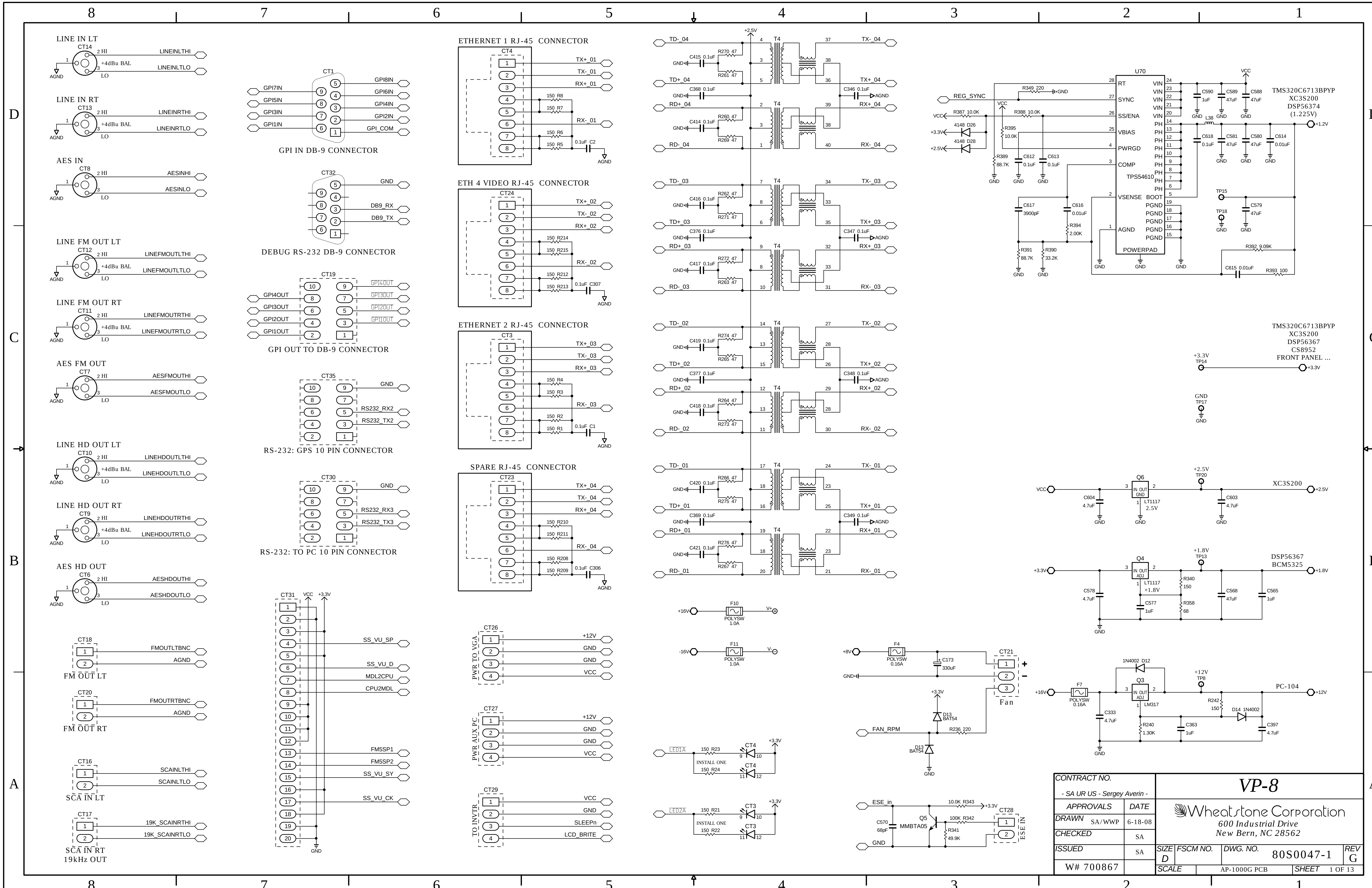
## Chapter Contents

## VP-8 General SIgnal Flow

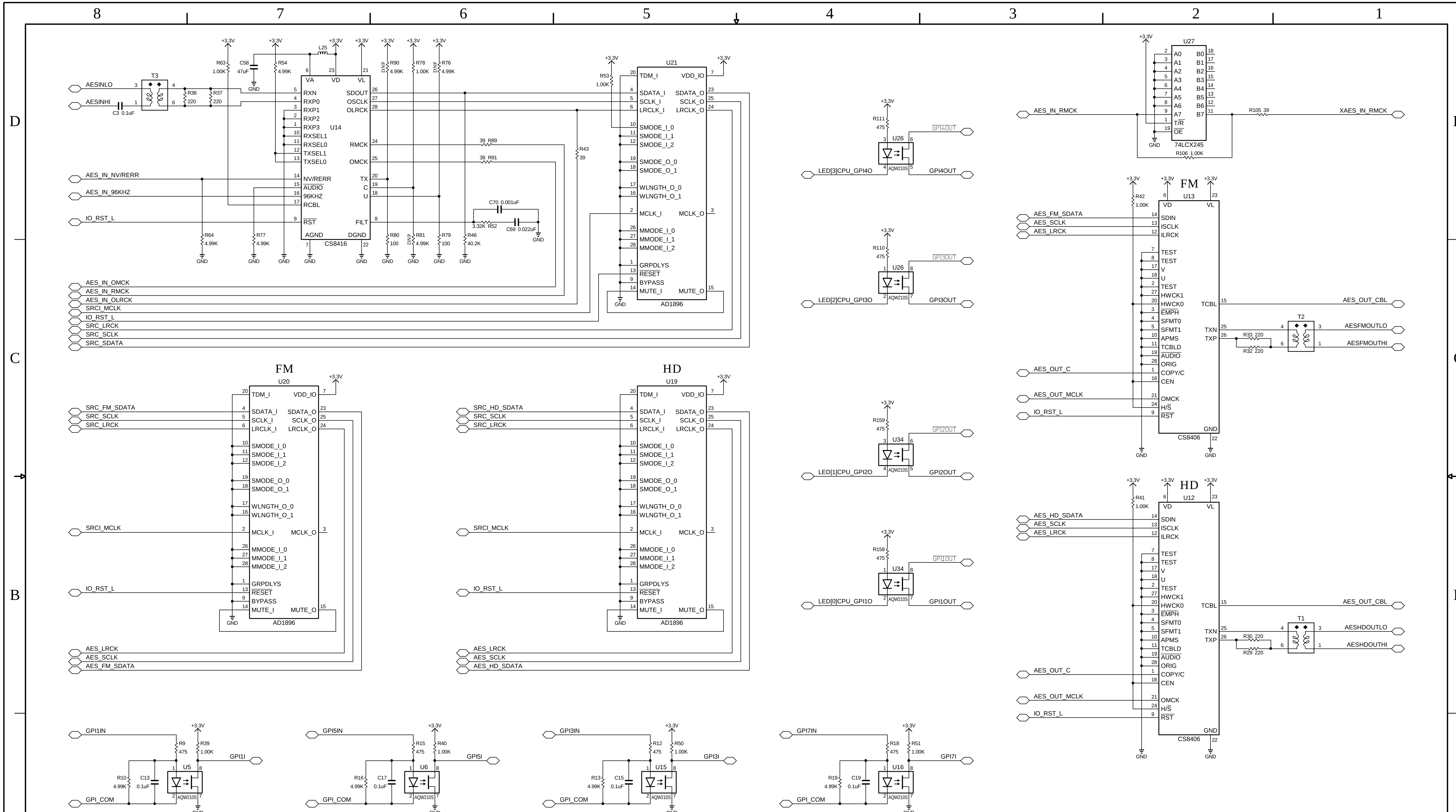
**(Version 2.0.0 Software and Above)**




VP-8 Signal Flow Diagram

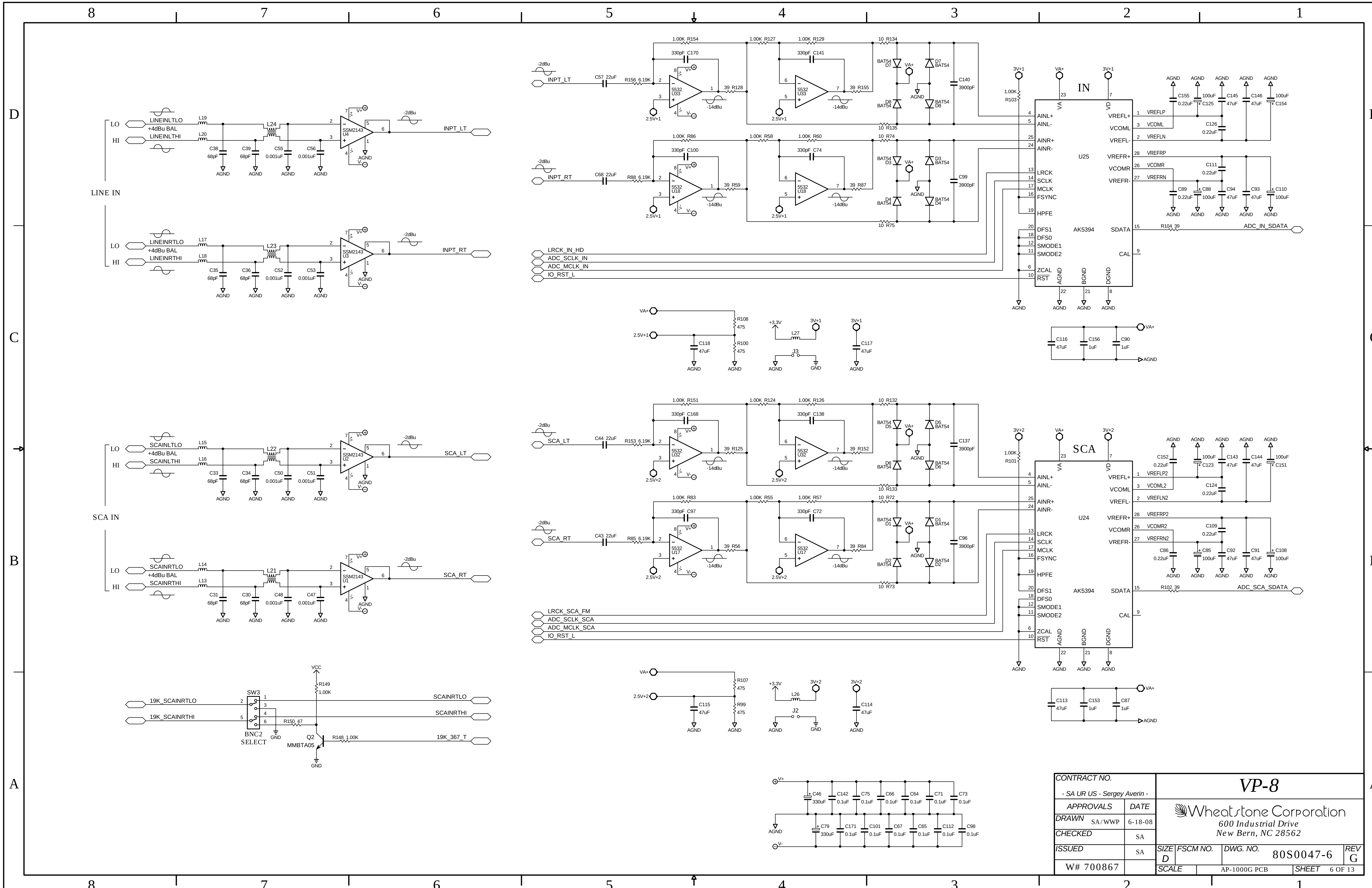



VP-8 Audio Processor Schematic

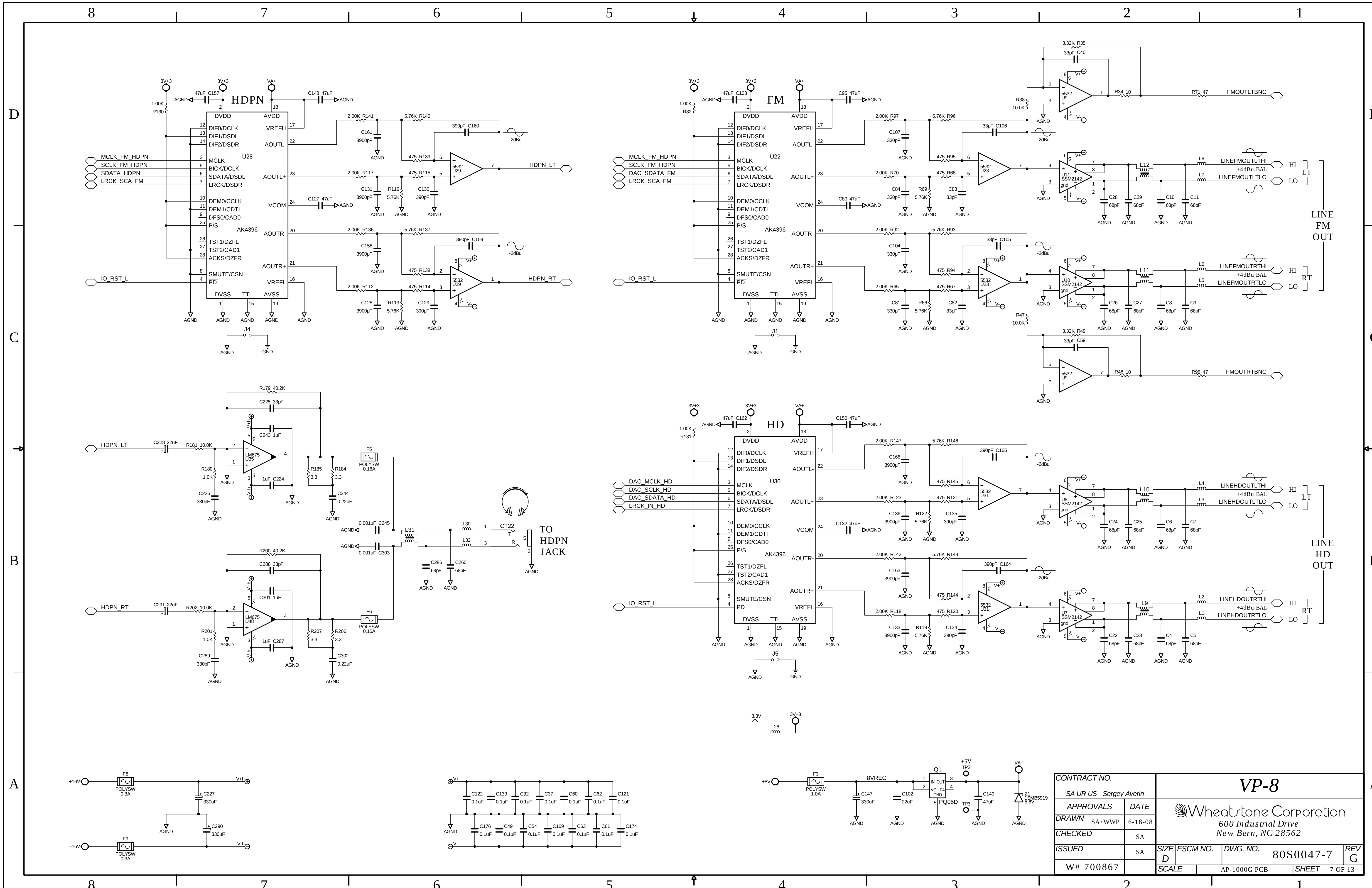



| | | | |
|---|---|---|---|
| CONTRACT NO. | | VP-8 | |
| - SA UR US - Sergey Averin - | | | |
| APPROVALS | DATE | Wheatstone Corporation | |
| DRAWN SA/WWP | 6-18-08 | 600 Industrial Drive | |
| CHECKED | SA | New Bern, NC 28562 | |
| ISSUED | SA | SIZE D, FSCM NO., DWG. NO. 80S0047-5 | REV G |
| W# 700867 | | SCALE AP-1000G PCB | SHEET 5 OF 13 |

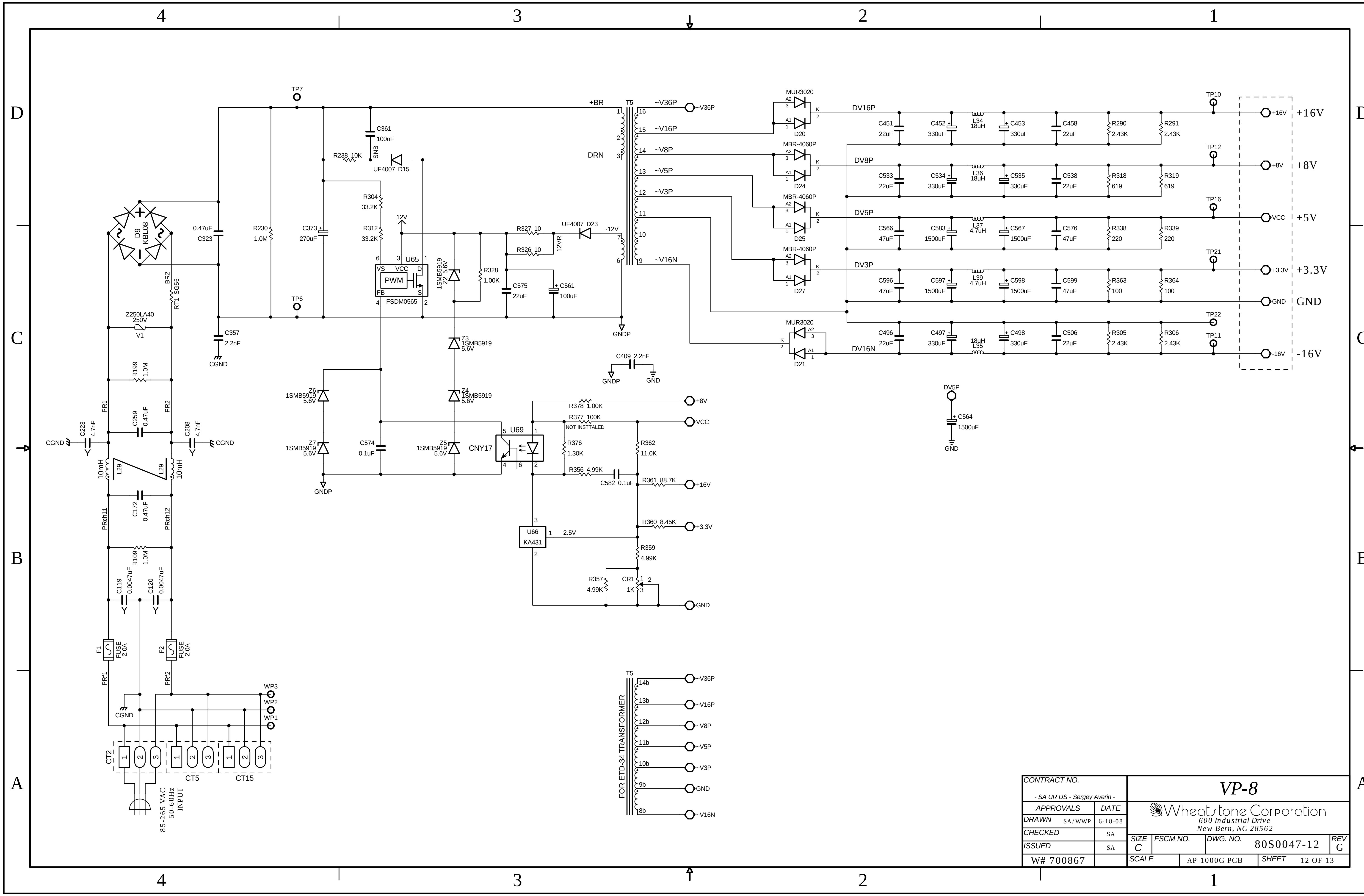
VP-8 Audio Processor Schematic



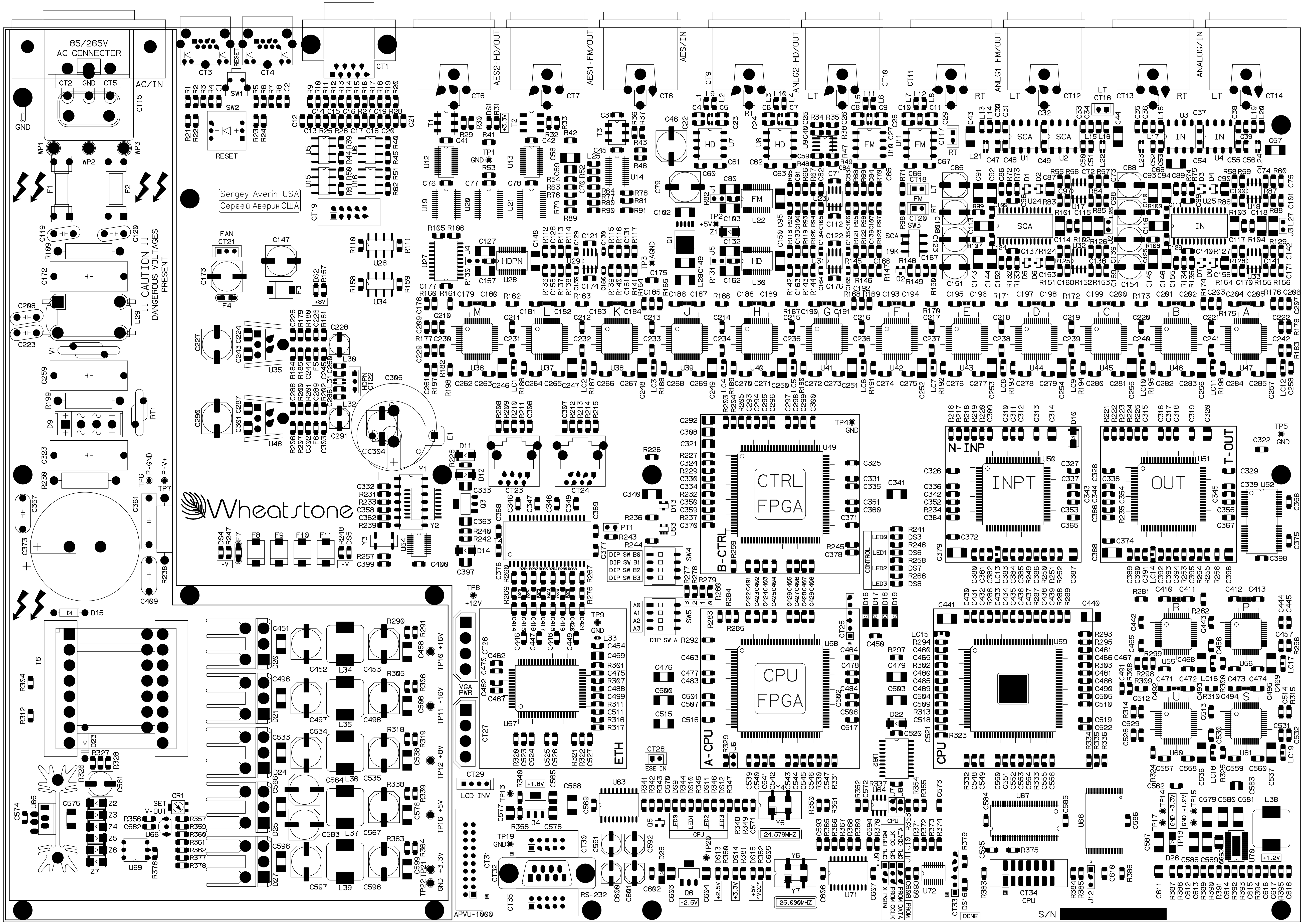

VP-8 Audio Processor Schematic




VP-8 Audio Processor Schematic




# VP-8 Audio Processor Schematic



VP-8 Audio Processor Load Sheet

# Appendix

## Contents

## Parameters, Units and Ranges

Approximately following the VP-8'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-8'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: | -20.0dB to +48.0dB, 0.5dB steps |
| AGC/Compressor Thresholds: | - 35dBFS to -65dBFS |
| AGC Threshold Backoff: | 0dB to -12dB |
| AGC Attack: | 50mS – 1500ms (1.5 sec) |
| AGC Release: | 100mS – 7.0 seconds |
| AGC Band Coupling: | 0dB to -30.0dB, 0.5dB steps |
| Compressor Attack: | 3.0mS – 1000mS (1Sec) |
| Compressor Release: | 20mS – 1000mS (1Sec) |
| Ratio: | 1:1 – 20:1 |
| AGC Gate Threshold: | -20.0dB to -79dBFS, plus OFF |
| Gate Thresh Trim: | +/- 6dB per band |
| Gated Modes: | Ooze/Hold |
| 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 VBMS 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: | 0 – 100% |
| VBMS Drive: | +/- 10.0dB |
| VBMS Style: | Soft/Hard |
| VBMS Output: | +/- 10.0dB |
| VBMS 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<br>Pre-look ahead limiter<br>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. |



## 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 | OFF/+8.0dB to +2.0dB, 1dB steps<br>+1.5dB to -2.0dB, 0.5dB 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,<br>+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<br>May be selected to provide pilot tone only |
| Composite Processor Modes: | Oversampled look ahead limiter<br>Soft Clipper<br>Hard Clipper |
| Stereo Encoder Output Operating Modes: | Analog L/R<br>Analog L/R De-emphasized<br>Unbalanced and Balanced Multiplex |
| Output signals available as: | |
| Digital: | AES 3-pin XLR (digital), and Stereo, pre or post Diversity Delay |
| Analog: | Analog Left/Right<br>Analog Left/Right, De-emphasized<br>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 VBMS 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: | 0 – 100% |
| VBMS Drive: | +/- 10.0dB |
| VBMS Style: | Soft/Hard |
| VBMS Output: | +/- 10.0dB |
| VBMS Frequency: | 60 - 300Hz |
| Multiband Knee: | Soft/Hard |
| Multiband Threshold: | +/-10dB relative to Drive |
| Multiband Attack: | 0.1ms – 100ms |
| Multiband Release: | 3ms – 300ms |

AM

| | |
|---|---|
| Pre-emphasis Modes: | Off/NRSC |
| Pre-emphasis insert points: | Pre-MB Limiter<br>Pre-look ahead limiter<br>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-8'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 |
| 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 VBMS with multiple operating styles, then an oversampled, zero overshoot look ahead limiter.

FM-HD

| | |
|---|---|
| Overall Controls | |
| Multiband Drive: | 0 – 100% |
| VBMS Drive: | +/- 10.0dB |
| VBMS Style: | Soft/Hard |
| VBMS Output: | +/- 10.0dB |
| VBMS 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 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 VBMS with multiple operating styles, then an oversampled, zero overshoot look ahead limiter.

| | |
|---|---|
| Overall Controls | |
| Multiband Drive: | 0 – 100% |
| VBMS Drive: | +/- 10.0dB |
| VBMS Style: | Soft/Hard |
| VBMS Output: | +/- 10.0dB |
| VBMS 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 VBMS with multiple operating styles, then an oversampled, zero overshoot look ahead limiter.

| | |
|---|---|
| Overall Controls | |
| Multiband/Drive: | 0 – 100% |
| VBMS Drive: | +/- 10.0dB |
| VBMS Style: | Soft/Hard |
| VBMS Output: | +/- 10.0dB |
| VBMS Frequency: | 60 - 300Hz |
| Multiband Knee: | Soft/Hard |
| Multiband Threshold: | +/-10dB relative to Drive |
| Multiband Attack: | 0.1ms – 100ms |
| Multiband Release: | 3ms – 300ms |



## 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 VBMS with multiple operating styles, then an oversampled, zero overshoot lookahead limiter.

| | |
|---|---|
| Overall Controls | |
| Multiband Drive: | 0 – 100% |
| VBMS Drive: | +/- 10.0dB |
| VBMS Style: | Soft/Hard |
| VBMS Output: | +/- 10.0dB |
| VBMS Frequency: | 60 - 300Hz |
| Multiband Knee: | Soft/Hard |
| Multiband Threshold: | +/-10dB relative to Drive |
| Multiband Attack: | 0.1ms – 100ms |
| Multiband Release: | 3ms – 300ms |



## 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-8 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-8 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-8 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-8 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-8.

## Software Remote Control

Supplied Windows® Vorsis GUI software affords control of all system and processing parameters via a 10/100BaseT Ethernet interface.