
HD P3 DIGITAL

MULTIBAND SIGNAL PROCESSOR

TECHNICAL MANUAL



HD P3 Digital Multiband Signal Processor Technical Manual

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Attention!

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

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



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

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

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



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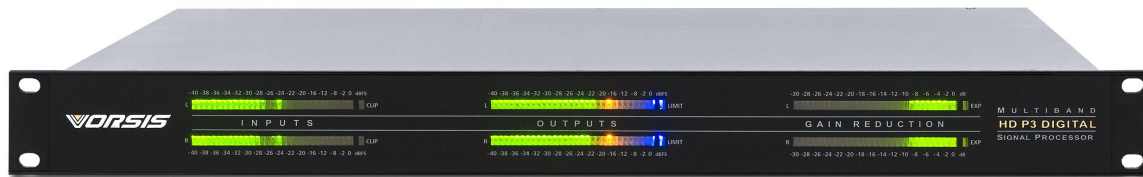
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General Information

Introduction

The Vorsis HD P3 Digital Multiband Signal Processor unit occupies one 19" wide rack space (height 1 3/4"), and is 17.346" deep. The two-channel/stereo audio processor provides an array of analog/digital inputs and outputs, four-band parametric EQ, three-band compressor, and tunable filters, along with overall AGC, limiting, and expansion. All settings can be stored and replayed as password-protected presets. The front panel display includes input, output, and gain reduction meters. Remote monitoring and operational control is possible using Ethernet-enabled GUI software.

The XLR, DB-25, DB-9 and CAT5 connectors located on the rear of chassis give you maximum installation versatility.

Energizing

Assuming the Vorsis HD P3 Digital Multiband Signal Processor is correctly rackmounted, you may now energize it by connecting the factory supplied power cord to the rackmount unit and then plugging it into the AC mains.

Note: To de-energize the Vorsis HD P3, unplug its AC cord from the AC mains.



I/O Connections

All audio input and output, control, Ethernet, and power supply connections are made via various connectors mounted on HD P3's rear panel. Seven XLR connectors are provided for analog and digital audio input and output, and all the audio connections also appear on the DB-25 connector. An RJ-45 connector is provided for Ethernet connections. The DB-9 connector provides 8 GPI inputs for the most used presets. The pinout drawings on pages 1-8 and 1-9 summarize all wiring connections.

Inputs

Both HD P3 processor signal paths can be fed from two different types of inputs: analog line level and AES-3 digital.

Two female XLR connectors, or the appropriate pins on the DB-25 AUDIO I/O connector, are available for line level analog audio input (nominal +4dBu balanced) connections. The analog line inputs exhibit a bridging impedance and can handle signals up to +20dBu.

A third female XLR connector is available to handle digital line level sources; this AES-3 digital input is also available on the DB-25 connector. The HD P3 will accommodate digital inputs having sample rates of 44.1kHz or 48kHz. The internal sample rate of the HD P3 will automatically sync up to the applied input signal's valid sample rate, allowing the HD P3 to be used in an existing digital system. AES sources are by design stereo; if the HD P3 is not set to be stereo the appropriate Left or Right signal within the AES stream will be applied to the signal path.

Line In—XLR-F

Pin 1 XLR1 SH – LINE 1 IN SH
 Pin 2 XLR1 HI – LINE 1 IN HI
 Pin 3 XLR1 LO – LINE 1 IN LO

Pin 1 XLR2 SH – LINE 2 IN SH
 Pin 2 XLR2 HI – LINE 2 IN HI
 Pin 3 XLR2 LO – LINE 2 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

AUDIO INPUTS—DB-25

Pin 25 – LINE1 IN SH
 Pin 24 – LINE1 IN HI
 Pin 12 – LINE1 IN LO
 Pin 11 – LINE2 IN SH
 Pin 10 – LINE2 IN HI
 Pin 23 – LINE2 IN LO
 Pin 5 – AES IN SH
 Pin 4 – AES IN HI
 Pin 17 – AES IN LO

Outputs

The HD P3's two output signals are available as individual line level analog (+4dBu, balanced) on the two male XLR ANALOG OUT connectors, and in digital AES formatted on the two male XLR AES OUT connectors. These analog and digital outputs are available also on the DB-25 connector.

ANALOG OUT—XLR-M

Pin 1 XLR1 SH – LINE 1 OUT SH
 Pin 2 XLR1 HI – LINE 1 OUT HI
 Pin 3 XLR1 LO – LINE 1 OUT LO

Pin 1 XLR2 SH – LINE 2 OUT SH
 Pin 2 XLR2 HI – LINE 2 OUT HI
 Pin 3 XLR2 LO – LINE 2 OUT LO

AES OUT—XLR-M

Pin 1 XLR1 SH – AES 1 OUT SH
 Pin 2 XLR1 HI – AES 1 OUT HI
 Pin 3 XLR1 LO – AES 1 OUT LO

Pin 1 XLR2 SH – AES 2 OUT SH
 Pin 2 XLR2 HI – AES 2 OUT HI
 Pin 3 XLR2 LO – AES 2 OUT LO

AUDIO OUTPUTS—DB-25

Pin 8 – LINE 1 OUT SH
 Pin 7 – LINE 1 OUT HI
 Pin 20 – LINE 1 OUT LO
 Pin 19 – LINE 2 OUT SH
 Pin 18 – LINE 2 OUT HI
 Pin 6 – LINE 2 OUT LO
 Pin 16 – AES 1 OUT SH
 Pin 15 – AES 1 OUT HI
 Pin 3 – AES 1 OUT LO
 Pin 2 – AES 2 OUT SH
 Pin 1 – AES 2 OUT HI
 Pin 14 – AES 2 OUT LO

General Purpose Interface

This general purpose interface is used to remotely trigger (recall) Presets 1-8. For example, a 5VDC signal applied between Pin 1 (GPI Com) and Pin 6 (GPI 1 In), will take the HD P3 preset stored in location PRE #001. The + (plus) side of the 5V signal is applied to the active selector pin (Pin 6 in the example) and the - (minus) side of the 5V signal is applied to GPI Com, Pin 1.

GPI—DB-9

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

Ethernet Interface (100Mbps)

Connect your HD P3 to your local Ethernet network using a “straight” (pin to pin) CAT5 cable. The default IP address and subnet is 192.168.1.191/255.255.255.0. Please refer to the HD P3 software for details on changing the default IP address. Use a crossover cable for peer-to-peer connection (i.e. direct connection between a computer and the HD P3). For typical CAT5 cable pinouts see below.

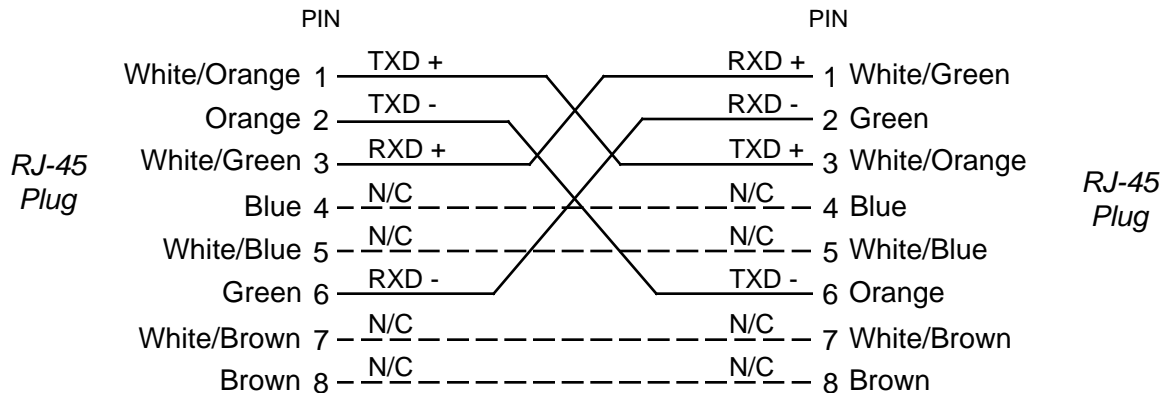
ETHERNET—RJ-45

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

	PIN		PIN	
	White/Orange 1	TXD +	1	White/Orange
	Orange 2	TXD -	2	Orange
<i>RJ-45 Plug</i>	White/Green 3	RXD +	3	White/Green
	Blue 4	N/C	4	Blue
	White/Blue 5	N/C	5	White/Blue
	Green 6	RXD -	6	Green
	White/Brown 7	N/C	7	White/Brown
	Brown 8	N/C	8	Brown
				<i>RJ-45 Plug</i>

TYPICAL CROSSOVER CABLE



Digital Audio Connections

CABLE - All AES/EBU input and output digital audio connections are balanced and should be made using a high quality digital audio cable. Be sure to select a digital audio cable with an integral drain wire of the same wire gauge (AWG) as the twisted pair as this facilitates an easier consistent termination process. Typical AES/EBU digital audio cable has a very low characteristic capacitance per ft (pF/ft), and a nominal impedance of 110 Ω . High quality digital audio cable offers better signal transmission performance versus typical analog audio cable, especially over long cable runs. Check the cable manufacturer's data sheet to be sure the cable you plan to use will work in your application.

CONNECTORS - The AES/EBU connections are made with the supplied DB-25 male mating connector, or with XLR connectors. The DB-25 crimp style connector will accept wire gauge 22 - 28AWG. Please refer to the XLR mating connector manufacturer's recommendations for termination instructions.

Unbalanced Analog Connections

ANALOG INPUTS — Wire to the processor input end with typical shielded, two conductor cable (like Belden 9451), just as if you were connecting a balanced source. At the unbalanced source machine's output, connect the + output to the HI input wire and connect the source machine GND wire to LO. Connect the shield at the HD P3 end only.

Note: Unbalanced analog sources typically have -10dBv (316mV RMS) signal levels and will not match the processor nominal operating level of +4dBu (1.23V RMS). We highly recommended that you first externally balance any unbalanced sources you plan on connecting to the Vorsis HD P3. Many third party "match boxes" are commercially available for this.

ANALOG OUTPUTS — The HD P3 analog outputs use a balanced output circuit which behaves exactly like the secondary of a high-quality transformer, with no center tap—this output is both balanced and floating.

For unbalanced operation, either the HI *or* LO side of the analog output must be strapped to ground of the unbalanced input, with the output taken from the other side. (Normally you would strap LO to ground, and use HI to feed your unbalanced equipment input.) Leave the SH floating at one end.

Unbalanced Digital Connections (SPDIF)

SPDIF INPUTS — The SPDIF (Sony/Phillips Digital Interface) or “consumer” digital audio interface is a two wire unbalanced signal typically on a single RCA style connector. Note that the SPDIF signal level of approximately 500mV and 75 ohm impedance does not correctly match the processor’s AES-3 input. We highly recommend using a “balun” or format converter when interfacing “consumer” grade source devices to the processor.

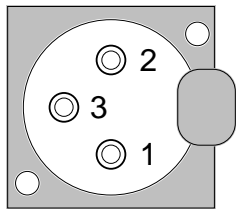
In cases where a consumer grade device must be interfaced and the appropriate matching device is not available, try wiring the SPDIF center conductor (HOT) to the HI input pin and SPDIF shell (ground) to the LO input. Connect SH at the processor end only.

SPDIF OUTPUTS — The Vorsis HD P3 digital outputs are fixed, professional, AES-3 formatted outputs. SPDIF consumer format is not supported. Use an external format converter to connect the digital outputs to consumer gear.



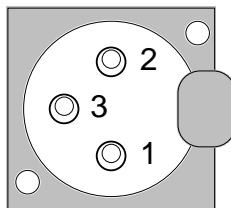
Line In - XLR-F

XLR1-F



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

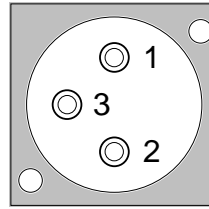
XLR2-F



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

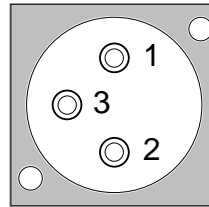
Analog Out - XLR-M

XLR1-M



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

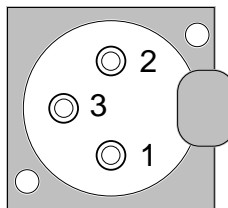
XLR2-M



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

AES In - XLR-F

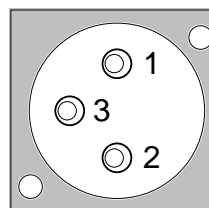
XLR-F



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

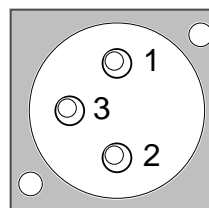
AES Out - XLR-M

XLR1-M



PIN 1 XLR1 SH - AES 1 OUT SH
PIN 2 XLR1 HI - AES 1 OUT HI
PIN 3 XLR1 LO - AES 1 OUT LO

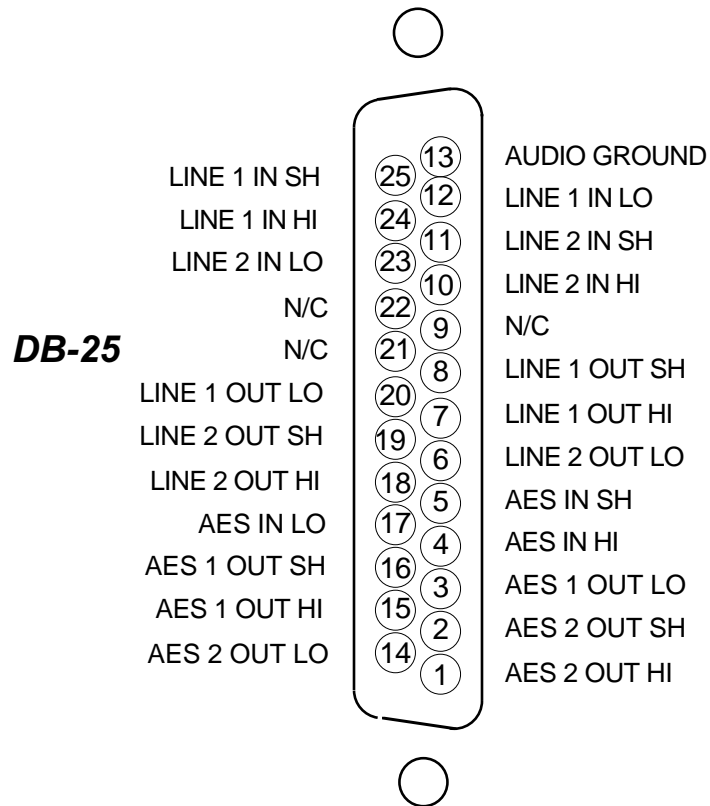
XLR2-M



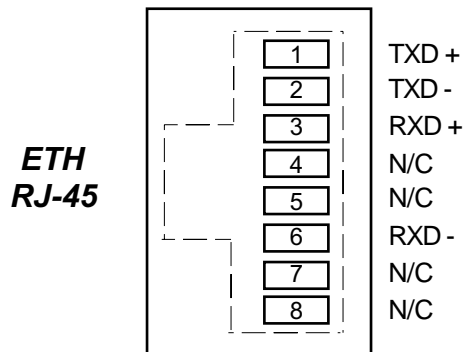
PIN 1 XLR2 SH - AES 2 OUT SH
PIN 2 XLR2 HI - AES 2 OUT HI
PIN 3 XLR2 LO - AES 2 OUT LO



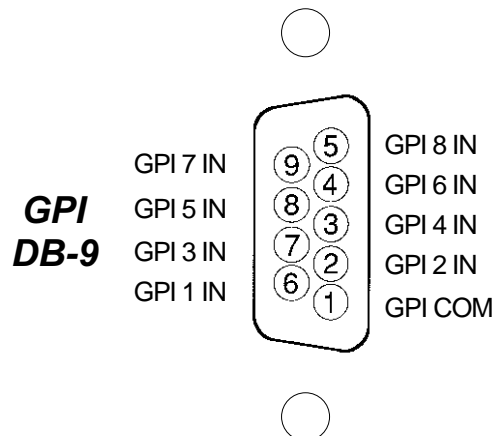
Audio Input / Output



Ethernet - RJ-45



GPI



Vorsis HD P3 GUI

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Vorsis HD P3 GUI

Getting Started

A PC application program to control one or many HD P3s is supplied with the unit. It is a Windows™ Graphical User Interface (GUI) program, intended to be straightforward in use, controlling and displaying the powerful features of the HD P3 to best effect. It is supplied as a self-installing program which may be executed from any directory on the computer, the resulting files being installed under “Program Files\Vorsis\HD P3”. The program may be started from the Windows Start menu. A registration key is required from Vorsis to operate the program longer than a week.

NOTE: 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 HD P3 GUI could be problematic.

Connecting the HD P3 and the GUI

The HD P3 and PC running the HD P3’s GUI program can be straightforwardly connected together over a standard Ethernet Local Area Network (LAN).

The HD P3 should be installed, powered-up, and verified to be operating normally; a CAT5 cable connected to the 100baseT LAN should be inserted into the appropriate socket on the rear. Likewise, the GUI should be installed on the desired PC which is verified to be working correctly with the LAN.

It is also possible to work without a LAN by connecting the HD P3 and the PC Ethernet ports together using a crossover cable.

Constraints for the system are not extraordinary, in that that the PC should be preferably at least 1GHz in speed, and that the screen be at least 1024 x 768 pixels in size. The LAN should be 100baseT (100MHz) capable. It is indeed possible for the GUI to be run on a portable computer connected to the LAN by wireless (802.11b as a minimum) as long as the HD P3 itself “sees” a 100baseT network. The dodgier the radio link, though, the more may be the impact on smoothness of the GUI’s real-time graphics.

IP Address Configuration

The HD P3 requires a *Manually Configured* IP address type. Automatically assigned network addressing using Dynamic Host Control Protocol (DHCP) is not supported. Your HD P3 was assigned the following default parameters at the factory:

IP address: 192.168.1.191

Subnet Mask: 255.255.255.0

Gateway: 255.255.255.255

The default IP address, subnet mask, and gateway may be changed using the HD P3 GUI's *Hardware>Assign IP Addresses...* menu item. **Right-click** anywhere to open this menu. Note that your PC's IP address does not need to be on the default HD P3 192.168.1.xxx subnet *in order to change* the HD P3's default IP address, but you **do** have to be on the same subnet in order to actually go *Online* with the HD P3 GUI.

TIP: Right-click anywhere on the HDP3 GUI window to bring up the menu.

You will need to know the Media Access Control (MAC) address printed on a label affixed to the rear of the top cover on your HD P3. This value is a string of twelve hexadecimal characters, for example:

MAC Address = 00 50 C2 23 C2 CD.

The easiest way to change and check the IP address is to create a new *Device*, then change the default IP address. Make sure both your PC and HD P3 are connected to the same physical network, or use a crossover cable for a direct, peer-to-peer connection between a PC and the HD P3. For instant verification via the GUI's metering, connect an audio source like a CD player and let it play. Once connected and Online, the meters will be moving on the HD P3. Try an analog source, then digital source if analog does not get the HD P3's meters going. You can also "ping" the HD P3 from the Windows Command Prompt.

Creating a New Device

This step creates a virtual device and points the GUI to it.

- Right click anywhere on the GUI and select *Hardware >Devices...* from the pop-up menu - the *Devices* dialog box opens.
- Press the **Add** button and enter an 8 character Name and the desired IP Address.
- Click OK when completed.
- Click on the newly created Device name and press the **Select** button. The *Devices* dialog box closes and the newly created Device name will appear in the *Devices* window at the top of the Vorsis HD P3 GUI.
- Click the **Online** button at the top of the control pane, when enabled a green LED is "lit" on the button, the Status window will display *Trying...*

You've created the Device and pointed the GUI to it, now let's change the HD P3's IP address.



Changing the HD P3 IP Address

This step changes the IP address of the HD P3's NIC. Note that your PC's IP address does not need to be on the default HD P3 192.168.1.xxx subnet.

- Right-click anywhere on the GUI and select *Hardware > Assign IP Addresses...* An *IP Assignment* dialog box opens.
- Enter the MAC address from the HD P3's label (affixed to top cover).
- Enter a new Name, IP Address, Subnet Mask, and Gateway.
- Press **Start** - you will be asked to re-boot the HD P3 (power down, then up).

In a few seconds after you cycle power you will see the *Requests* counter begin to increment, usually, up to 2. In a few more seconds you should see the *Status* window change from *Trying...* to *Online*. The audio meters will also begin moving if you have audio playing and are actually Online.

You may now Close the *IP Assignment* dialog box. That's it!

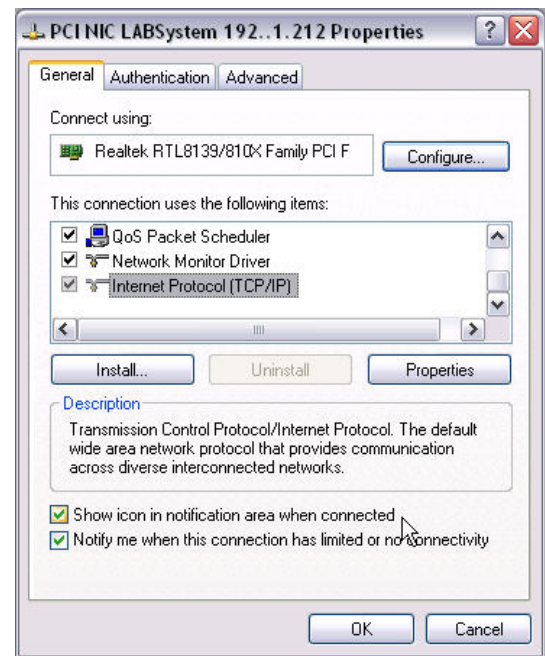
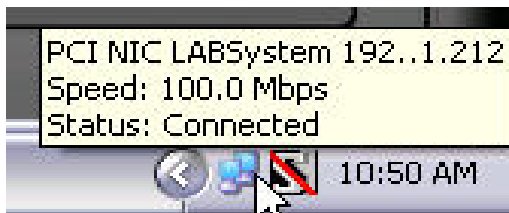
Troubleshooting IP Addressing Problems

- Once you have changed the default HD P3 IP address, be sure your PC's NIC is on the same subnet as your new HD P3 IP address.
- Be sure the *Device* is defined correctly in the *Hardware > Devices...* menu.
- Be sure you have *Selected* the desired Device in the *Hardware > Devices...* menu.
- Check that the Ethernet cables are plugged in and LINK led's are lit.
- Use a *crossover* cable if connecting the PC directly to the HD P3. Try using Windows XP *Repair* connection to reset the PC's NIC.
- Check that the HD P3 is powered ON.
- If Online but no audio, check that the Input source is correct, Analog or AES.

If for some reason the device does not connect and go Online, try following the instructions for *Creating a New Device* and *Changing the HD P3 IP Address* again.

Windows Network Status Icons

The Windows NIC Properties form includes a checkbox to show an extremely useful network connection icon in the Windows XP System Tray. You can access IP info, connection status, repair the connection, and basically “see” when Windows is connected or trying to establish a network connection. To turn on the ICON; Open *Control Panel>Network Connections*. Right-click on the LAN connection for your NIC card and select *Properties* then check the *Show icon..* checkbox.



A Word About Networks

Most DHCP networks assign addresses starting at the bottom of the group 192.168.0.xxx; choosing an address high in that group, say 192.168.0.200 will likely keep it out of the way of the busiest DHCP addressing. However, it is always wise to check with your local friendly network administrator. If the network warrants it, the “Subnet” and “Gateway” values may need to be differed, but the suggested values of 255.255.255.0 (Subnet) and 255.255.255.255 (Gateway) will suffice for all but complex situations.

For security, the HD P3 will not act as a “web device” or support open connection to the Internet. However, connection between a GUI and HD P3s through the Internet between remote locations is entirely possible by arranging for the local LANs upon which they are residing (say “Studios” and “Transmitter_Site”) to be connected via a “Virtual Private Network”, or VPN; this acts as a “tunnel” through the Internet allowing the participants to behave as if they are all on a single big LAN. Setting up a VPN – although not difficult - is beyond the scope of this manual owing to the uniqueness of each situation.

Using the GUI

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

Description 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 containing the Dynamic Displays, then the Control Area above it, followed by the Side Bar buttons, and lastly the Title Bar.



Dynamic Displays Region

Frequency-Domain Graph

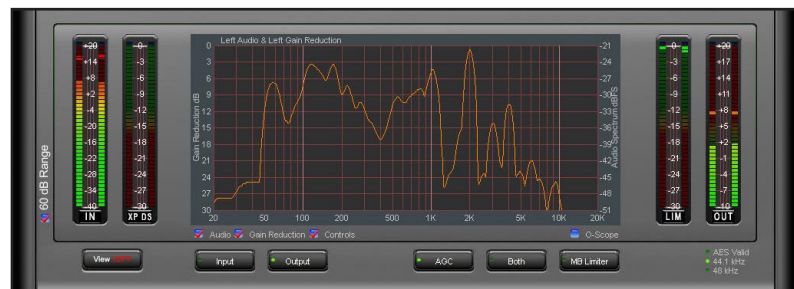
Visually, the most important element on the main screen, dominating the lower half of it, is a large graphical display; the standard audio frequency of 20Hz – 20kHz is ranged in logarithmic form across this graph with gradations at salient frequencies. The vertical axis scale changes according to context, although the overall vertical range remains at 30dB, with gradations every 3dB.

A number of real-time dynamically varying spectra and other frequency-response-versus-time effects may be displayed simultaneously. A row of large buttons under the graph determine which are displayed, and also determine the presence or otherwise of equalizer frequency response, filter responses, de-esser sidechain filter response, etc. (The de-esser's dynamic frequency response shows here, too, as a dynamic “upside-down shark-fin” when in operation.):

INPUT — A Fast-Fourier Transform (FFT) based real-time spectral analysis of the selected (Left/Right, A/B) input signal.



OUTPUT — This is a real-time spectral analysis of the selected output signal. (The scaling of either the input or output display may be altered by clicking on and vertically sliding the right-hand “dB” scale, to best position the curve.)



COMPRESSOR — A curve representing the instantaneous gain-reduction by frequency of the three-band compressor / multiband limiter is drawn.

AGC — Displays the gain-reduction due to AGC action across the spectrum; since the AGC is three-band, a curve joining the LF, MF and HF gain reductions is drawn.



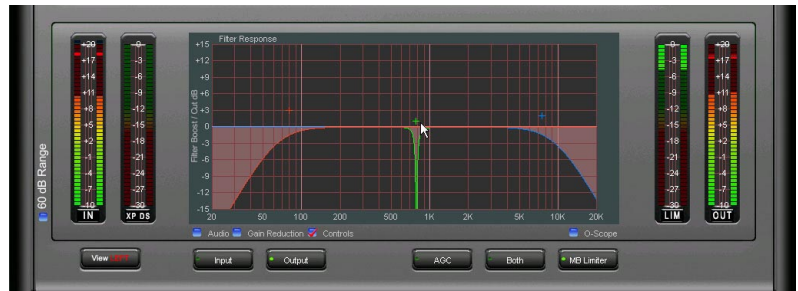
BOTH — This displays a curve of the composite (instantaneously greatest) gain reductions spectrally from both the AGC and the compressor.

VIEWS — “Views” cycles through a short set of display options:

- “Live Audio” - the above selection of real-time information,
- “EQ/Filters” – the frequency responses of applied processing,
- “Both” - dynamic information and response curves together.

Bargraph Metering

Adjacent to the graph, both to the left and right, and sharing its 30dB vertical range (0-30dB of gain reduction, 30dB signal level meter range), are a number of bargraph level indicators. They indicate signal levels and gain reductions of elements in the signal path; those ahead of the multiband compressor (input side) are to the left of the main graph; those after to the right:



Left of Graph:

- left input signal level
- right input signal level
- “XP” expander gain reduction
- “DS” de-esser gain reduction

Right of Graph:

- left limiter gain reduction
- right limiter gain reduction
- left output signal level
- right output signal level

The level meters are dual-indicating peak-over-average, 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.

To the left of the screen is a small button labeled “60 dB Range”, allied to the input signal level bargraph; when clicked, it toggles between normal (30dB, un-ticked) range and 60dB (ticked). This latter is useful for “finding” signals during initial set-up, or when the signal source is of high dynamic range, such as live classical.

Signal Path Selection

Beneath the input metering bargraphs is a large button, which toggles between “View Left” (colored red – port) and “View Right” (green – starboard) if the HD P3 is set up for stereo, or “View A” and “View B” if operating as two discrete channels. This selects which of the two signal channels within the HD P3 is being operated upon / displayed.

Control Area Region

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

All the variable knobs and faders are within this area. To operate them, either:

- place the cursor above 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. In either case, the resolution of the control variation can be increased - made finer - by holding down the keyboard’s control key (“Ctrl”) while adjusting the control as above.

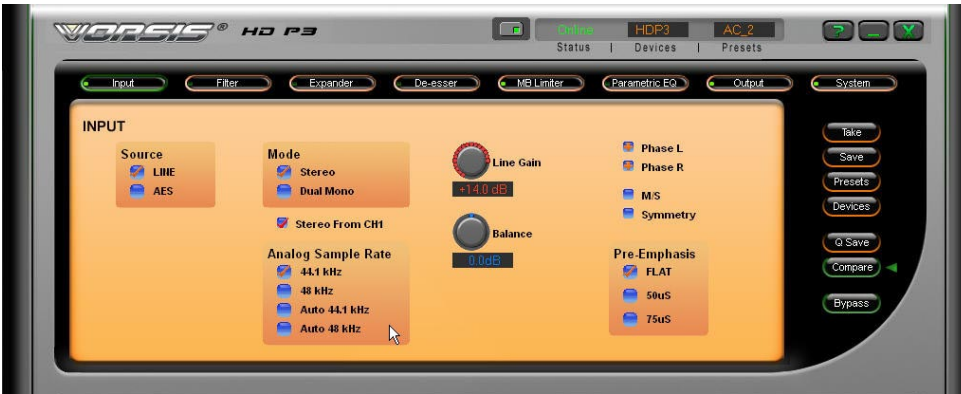
Left-to-right across the upper portion of this area are a series of buttons corresponding to individual or logically grouped processing elements in the signal flow, in input-to-output order. A small indicator at the left end of each button indicates if any signal processing within that block is currently active. Left-clicking a button gains access to the controls for that portion of the signal-processing.

NOTE: The order of EQ and compression can be reversed on the HD P3 by the condition of “Pre-EQ” selection. This is achieved on the GUI by clicking upon the “SWAP” button linking the EQ and compressor blocks visible in either of these functions’ control screens.



Input

Representing input source selection, operational mode, input gain control, balance/offset, symmetry decorrelator, absolute and relative L/R phase, M/S capability, and pre-emphasis selection.



ANALOG SAMPLE RATE — Allowing selection of 44.1kHz or 48kHz internally generated sample rates, and the option of Auto 44.1kHz and Auto 48kHz under which the HD P3 synchronizes to the external sample rate of a valid applied AES/EBU input signal. Virtual LED's in the lower right corner of the HD P3 GUI indicate the status of the external sample rate present at the AES input connector. The VALID LED will go out If the Auto select frequency and the actual sample rate present at the AES input jack differ.

MODE — Through two buttons, this determines if the HD P3 is going to operate in “Stereo” or as two discrete channels (“Dual Mono”).

STEREO FROM CH1 - when checked, the common stereo parameters will be controlled from the LEFT channel controls. Right channel controls are ignored. If un-checked, the RIGHT channel controls are active and the LEFT channel controls are ignored.

Signal Path Routing

The various HD P3 modes and features allow for simple or complex signal paths depending on the state of certain checkboxes in the HD P3 GUI. The following table outlines the various combinations of “switch” settings and the results that can be seen at the outputs.

HD P3 ROUTING

Source Mode	Mode Options	Input M/S	Output M/S	Analog Out L R	AES SPLIT	AES Out 1 L R	AES Out 2 L R
Stereo	None	—	—	L R	—	L R	L R
Stereo	AES SPLIT	—	—	L R	ON	L L	R R
Stereo	M/S(i)	ON	—	L+R L-R	—	L+R L-R	L+R L-R
Stereo	M/S(o)	—	ON	L+R L-R	—	L+R L-R	L+R L-R
Stereo	M/S(i) + SPLIT	ON	—	L+R L-R	ON	L+R L+R	L-R L-R
Mono	None	—	—	1 2	—	1 2	1 2
Mono	AES SPLIT	—	—	1 2	ON	1 1	2 2
Mono	M/S(i)	ON	—	1+2 1-2	—	1+2 1-2	1+2 1-2
Mono	M/S(o)	—	ON	1+2 1-2	—	1+2 1-2	1+2 1-2
Mono	M/S(i) + SPLIT	ON	—	1+2 1-2	ON	1+2 1+2	1-2 1-2

NOTES:

1. M/S(i) = M/S switch on Input pane.
2. M/S(o) = M/S switch on Output pane.
3. If both M/S(i) and M/S(o) are ON and the input mode is Stereo, the M/S function is canceled out.
4. 1 = Analog input channel 1 or Left side of AES input.
5. 2 = Analog input channel 2 or Right side of AES input.
6. L = Left analog or AES Input; R = Right analog or AES input.

Symmetry

Specifically intended for use with voice, for greatest benefit it needs to occur before any dynamics processing has taken place. By decorrelating the fundamental and harmonic elements of a voice (far less painful than it seems) the relatively high asymmetry (“positive” going signal peaks vs. “negative” going peaks) of some 8 to 10dB of a normal voice can be substantially reduced, with little audible detriment. This, by in effect reducing the “peak to average” ratio of the voice signal, can allow a significant increase in apparent voice loudness in a dynamically controlled path.

Phase

Phase reversal of the signal path, in case there is an inadvertent reversal elsewhere in the signal chain of which the HD P3 is a part.

Filter

High-pass, low-pass filters, and notch.

High-Pass Filter

A fast rolloff-rate high-pass filter, adjustable between 20Hz and 1kHz.

Notch

The notch is tunable over the entire audio range of 20Hz – 20kHz.

Low-Pass Filter

A fast rolloff-rate low-pass filter, adjustable between 20kHz and 1kHz.



Expander

An expander such as the one in the HD P3 (often called a “downward expander”) is a useful tool for reducing unwanted background noises. These could be variously air-conditioning rumble or noise, background conversation, phone-line noises, recording hiss, etc. It is also useful for reducing the inevitable general increase in background noise of some recorded material when subject to heavy compression. A common usage in live sound is to effectively turn a microphone off when not being talked/sung into, so as to reduce corruption of a mix or reduce the chances of feedback with an unwanted open microphone.

The expander is slightly counter-intuitive when first encountered, in that unlike nearly any other processing element it is active - i.e. working, attenuating away the input signal - when the input signal is at its quietest, at or below the threshold. If the expander is on, there will be gain reduction when no signal is present. The gain-reduction reduces as the threshold is approached, and there is none above the threshold.

The controls are:

THRESHOLD — Below which the automatic attenuation starts to take effect.

RATIO — Being the proportion of how many dB the input signal is attenuated for every dB it drops below the threshold. 1:3 indicates 18dB loss for 6dB drop in input signal level.



DEPTH — The maximum amount the expander is permitted to reduce the input signal level.

OPEN — The time-constant of the rate at which the expander un-attenuates, or opens; sometimes called “attack”.

HANG — An adjustable period of time the expander remains open without attenuating, before starting to close. Handy to keep the expander open during, say, speech inter-syllables or other short pauses, without having to resort to excessively long...

CLOSE — ... close times, being the rate at which the expander attenuates away the input signal once below the threshold.

An input/output plot, a graphical representation of the relationships between threshold, ratio, and depth, is on the Expander Control screen of the GUI; it is a handy visual aid.

Almost always, the trick is to set the threshold of the expander - below which it starts to attenuate away the input signal - high enough to capture the noise, but not too high as to snatch at the lower levels of the desired parts of the program material. That can sound really irritating.

Sometimes the gain reduction is required to be subtle so as not to draw attention to the fact that the expander is in operation; under these conditions shallow expansion ratios, such as 1:1.5 or 1:2 are preferred, as are restricted depth - 6dB, or 10dB, is plenty and makes a substantial subjective improvement to the noise.

These, too, are the kind of settings used for another application of an expander: effectively shortening an excessively long room reverberation time, or an instrument's ring-out that is overly persistent. In these cases the threshold is set somewhat higher, well up into the desired audio levels - in this way the attenuation becomes part of the overall sound, but the gentle ratio prevents a sense of anything “odd” happening. Again, relatively shallow depths of 12dB or so are plenty to achieve the desired effect.

More aggressive expansion, or “gating”, is accomplished with steep ratios (1:3, 1:5) and with shorter open and close times than for “unobtrusive”. It is still best not to go overboard with depth - even just 14dB, 20dB tops, is enough to make a signal “disappear” in the context of a mix; the whole gating sound, especially surprisingly its opening, is less obvious with shallower depth. Sometimes the “Surprise!” element is required, though, for effect.



De-Esser

The de-esser is a dynamic equalization section, designed to restrict the output level of a chosen frequency range within the spectrum. The classic usage is to hold down the sometimes prominent and objectionable sibilant “hissy” noises in speech, particularly if poorly recorded or subject to poor or overly close microphone technique. It is also often used to help control the “honky” effect on phone lines; mercifully, with digital phone systems this is less of an issue than it once was, but it still crops up. The controls are:

THRESH — Threshold is the signal level (within the de-esser’s controlled bandwidth) at which the de-esser holds the output level, again, within the de-esser’s bandwidth. In other words, a signal within the de-esser’s band will not be allowed to exceed this threshold level.

ATTACK — The time-constant controlling how quickly the de-esser responds to a signal exceeding the threshold.

RELEASE — The time-constant controlling how long the de-esser takes to recover from a signal that has exceeded the threshold.

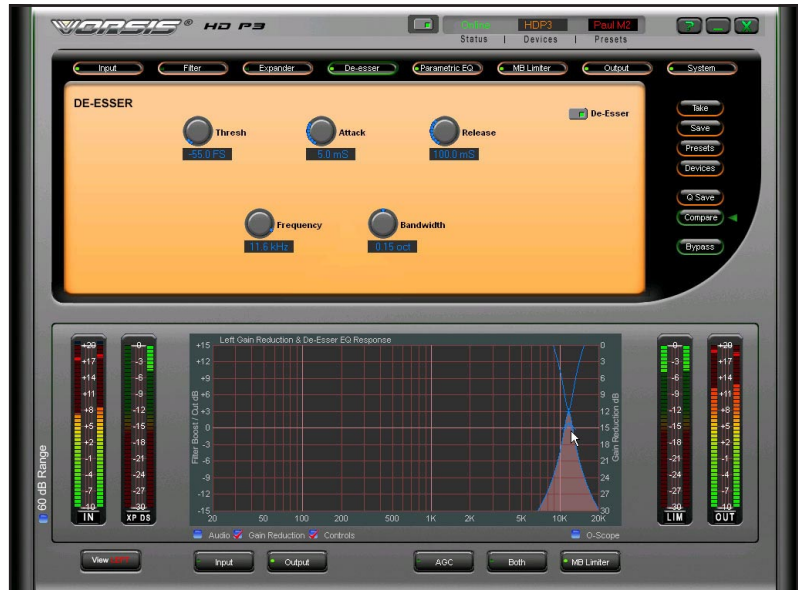
FREQUENCY — This is the center-frequency of a filter which determines the band of frequencies over which the de-esser operates. It is operated in conjunction with . . .

BANDWIDTH — ... which determines how wide on either side of the center-frequency the de-esser is sensitive.

The de-esser screen on the GUI brings up a graphical representation on the big frequency-domain graph of the de-esser’s filter, as controlled by “Frequency” and “Bandwidth”; the 3dB-bandwidth of the filter is arranged to be at the center-line of the graph for easy visualization. Particularly when the input signal spectral analysis is being displayed, it is often easy to visually “tune in” the de-esser to the visual clues of the objectionable sound on the analyzer.

Once the threshold within the de-esser’s band has been broached, an “inverted shark’s fin” reaches down from the top of the frequency-domain graph, indicating the width in frequency and depth of attenuation due to the de-esser.

Regardless of the visual clues available thanks to the GUI, the most direct way of tuning in the de-esser is the old-fashioned way - by ear. Make the “Bandwidth” small, narrow; crank down the threshold (i.e. make it more sensitive), then carefully tune the “Frequency” until the undesired artifact is getting audibly affected. Careful (!) adjustment of “Bandwidth”



and “Threshold” is then undertaken to best control the artifact. It is nearly always a compromise - as with any “garbage disposal” type processing - between reducing the undesired artifact, and mitigating its effect on the rest of the sound.

A less conventional use for the de-esser is as a dynamic EQ. Using it to shape the frequency response at higher signal levels (leaving it alone at lower levels) can be far less sonically obvious than digging a big hole with conventional EQ. An example would be using the de-esser to control the mid and mid-high frequencies of a music bed over which is to be laid a voice-over; the bed can be kept loud and energetic - and still natural-sounding - while not interfering with the important voice frequencies.

It should be noted that the de-esser is quite sensitive to changes in input level; ensure that the input signal is at a reasonable level, and stays that way, for best and consistent effect.

Emphasis and De-Emphasis

Emphasis is available at the HD P3’s input, and both pre-emphasis and de-emphasis at the output.



Emphasis (also known as pre-emphasis, particularly in circumstances where it is operating in conjunction with a complementary de-emphasis) is the imposition on the frequency response of a system a fixed and known frequency-dependent gain; the most usual implementation – and that used here - is an emulation of the fixed 75uS (Americas) or 50uS (elsewhere) high-frequency boost used in FM broadcast transmission. These, roughly speaking, have a turnover frequency (at the 3dB gain point) of 2kHz and 3kHz respectively, with a 6dB-per-octave gain increase above.

The reasoning is that since most program material has decreasing energy with progressively higher frequency above the mid-range, higher frequency signals may be boosted accordingly and still not breach headroom or deviation limits. Why? So that a complementary de-emphasis filter at the far end of the transmission chain (receiver) can reduce them equally, so as to render a flat frequency response again. Why? So that high-frequency noise inherent to and introduced into the transmission chain may be reduced by the rolloff of the de-emphasis filter, too, dramatically improving the noise performance of the path.

Since the HD P3 has great utility in both FM and AM broadcast paths, a highly comprehensive set of pre- and de-emphasis controls is made available. In approximate order of complexity, typical applications are listed:

Transmitter pre-emphasis

The usually built-in pre-emphasis in FM stereo encoders and exciters may be replaced by pre-emphasis in the HD P3. The advantages are that it is then possible to monitor pre-transmitter exactly what the overall processed signal is doing; the STL (Studio-to-Transmitter Link, often a radio itself) may be included within the pre-emphasized section of the path, and avoiding (by its disablement) a separate pre-/de-emphasis sub-path within that link; the excellent zero-overshoot output peak limiter in the HD P3 captures any overshoots that may spring from the pre-emphasis.

Using the pre-emphasis from the “Output” section allows the modified signal to be bounded by the output peak limiter, as described. Alternatively, pre-emphasizing at the “Input” allows the full gamut of processing to be operational on the pre-emphasized signal; this can be advantageous, in that the multiband compressor/limiter in particular can offer a greater degree of control over the now-contoured signal, and leave less to the output limiter to catch; this has sonic benefits. This also makes the HF band of the multiband processor “work harder”, which may at times be of benefit if “LOUD” is the desired end result, but at the very least will require deliberate attention and scrutiny - “too much of a good thing” department. The triband processor certainly will have to be set up differently for pre-emphasis; it will otherwise do a good job of neutralizing the boosted high frequencies, with the net result it will almost be as if pre-emphasis were never even applied!

When adjusting processing pre-emphasized at the input, it is “vital” that monitoring is done with complementary de-emphasis in place, either within the HD P3, or through end-to-end off-air monitoring. “Raw” pre-emphasis sounds very, very bright and indeed unnatural, and it is very easy to inadvertently adjust things such that the overall chain sounds dull, since an innate tendency is to make a path sound “flat”, or at least more familiar and natural.



Transmission path emulation.

Particularly when an HD P3 is being retrofitted into an existing airchain path, or one which maybe does not allow the flexibility to early-pre-emphasize, as above, the HD P3 may be used to internally capture, control and so neutralize any downstream “surprises” from pre-emphases in STLs, exciters, or encoders.

This is achieved by pre-emphasizing at the “Input”, and complementarily de-emphasizing at the “Output”. In this way, the processing “sees” a pre-emphasized signal (such as would exist in an STL and/or sender) and can control it appropriately, catching things and handling them perhaps more gracefully than the more rudimentary limiters downstream might. Additionally, a fully processed yet “flat” signal is available for monitoring; DJs like to hear the processed signal, rather than the usually much less exciting sounding console output.

So, in short: The pre-emphasis is to emulate that required of a typical transmission scheme. By bringing it into the HD P3, superior control over the overall signal path, and superior airchain processing, may be achieved. The de-emphasis in the box allows for normalization of a pre-emphasized signal within the box either for monitoring, or where a “flat” signal is required for onward passage.

M/S

This is a technique for stereo used in many places for different reasons, and is often known as “Sum / Difference” processing.

An “M/S” switch, as implemented in the HD P3, will turn a conventional left/right stereo pair of signals into a sum-and-difference pair; the “left” path then becomes a mono sum of the applied left and right signals, the “right” path now carrying the difference (left minus right) of the applied pair. Conversely and conveniently, the same “M/S” processing turns an applied sum-and-difference pair into their constituent left and right signals. Neat, eh?

The HD P3 has “M/S” switches at both “Input” and “Output”. This allows a wide range of additional processing possibilities:



M/S microphones

The name “M/S” in fact comes from “Mid / Side” - a stereo microphone technique where instead of a left and right sensing pair of microphones, there is a single forward-sensing (or omni) microphone, and a second figure-of-eight sensing microphone aimed sideways. Particularly in classical and other acoustic-space-rich style recordings, this technique offers latitude for easy remote-pattern and stereo image manipulation, and far better sound-stage positional accuracy than possible with a conventional pair of cardioid or super-cardioid microphones.

This sum/difference pair may be converted to conventional stereo by either the “M/S” switch at the input or output of the HD P3; the latter offers the possibility of differential processing (meaning different between sum and difference paths) way beyond the usual “width” control afforded by merely adjusting their relative gains.

Mono Airchain Processing (e.g. for AM)

The “M/S” switch will present a mono sum of the applied left and right signals from a stereo production or air-suite into the “Left”, or “A” path of the HD P3. The “Right” or “B” path would be ignored in this case.

FM Airchain Processing

Particularly in conjunction with the earlier described internal pre-emphasis, the “sum” and “difference” signals required for stereo multiplex encoding may be created within the box; this, again like local pre-emphasis, allows greater control over the ultimately broadcast signal.

Sum / Difference Processing for fun

With both the “Input” and “Output” “M/S” switches set, a stereo source will emerge from the box in stereo, yet the whole gamut of intervening processing will operate on their sum and differences. If the box is kept in “Stereo” mode, with identical processing occurring on the two paths, the net effect will be substantially the same as if the “M/S”ing hadn’t been done (dynamics tend to act slightly differently when driven 6dB harder, as will occur on a substantially “mono” source, but otherwise it’s pretty much a wash). If, however, once baseline configuration of the two paths has been done in “Stereo” mode, the HD P3 is then made to operate in “Two Monos” mode, things get interesting.



Width

As alluded to under “M/S Microphones”, changing the difference path gain in relation to the sum affects the apparent stereo “width”; reducing the difference to zero leaves solely the sum, i.e. mono; if the path gains are equal normal stereo occurs, but if the difference gain exceeds the sum, then a “super-wide” image is possible. It is quite extraordinary to “hear” sonic images come from outside the physical separation of a pair of speakers. This technique is called “shuffling”, and like all good parlour-tricks is best used in moderation - it is too easy to create “hole in the middle”, for the stereo effect to disappear and with two mangled images huddling around the speakers. Bear in mind that most good recording guys know this trick already and use it to some degree, usually on reverbs; an inadvertent excess added by your dainty hand can sound horrible.

Mono Low Frequencies

For disparate reasons, it has long been common for the heavy energy containing low-frequency elements of a mix to be panned central on recordings. Thankfully setting aside the constraints of disc-cutting(!), the main reason is that it is possible to get up to 6dB more level when using both channels mono-like instead of just one, as would be the case of something panned hard left or right. (More level = loud = good, Igor!) However, some program material isn’t like that, either because it’s old (“stereofied” fifties and sixties singles, Beatles in particular) and having instruments panned hard; the producers didn’t get the message; were trying to be artsy; or it just naturally isn’t like that (jazz, classical). In the kind of world the HD P3 inhabits, centering the LF energy has many benefits: in FM the difference subcarrier is excited less, which translates into less “scratchiness” on marginal signals; the LF can be made louder (that two channels vs, one thing again); perceptual encoders can create more efficient data compression with less difference energy to “aim”.

By high-pass filtering away the “difference” LF energy, one is just left with that in the “sum” path, i.e. mono. Compression / limiting downstream will allow any additional gain to be realized.

Naturally, this trick is not limited to just LF; particularly in conjunction with “Width” (above), filtered, EQ’d and dynamically modified “difference” paths can lead to some interesting, some wild effects.

But a word of warning. If the station’s music library, shall we say, owes much to the glory days of Napster, or otherwise contains aggressively perceptually encoded material, one should be highly circumspect of doing much in the way of “sum”/“difference” processing. Some perceptual encoders play games with, indeed take great liberties with, “difference” information during encoding, using the absolute bare minimum to convince the ear of an image’s stereo positioning. By and large it works very well in context, but not when the difference information is isolated or exaggerated - like by the very techniques just described above. Then, there can be excruciatingly nasty-sounding images at best, space-alien “beepity-boop” noises at worst. (Go on, try it.)



Output Limiter

Understanding the output limiter is also key to grasping the Multiband Limiter's operation, detailed later.

The HD P3's output limiter is designed to strictly control the unit's maximum output level such that overdrive or clipping the input of a downstream device is prevented. Typical concerns are the introduction of distortion products in perceptual encoding systems (resulting in unnecessarily high processing/bandwidth, inefficient encoding, and "funny noises"), FM sender over-deviation (naughty), and AM sender over-modulation (calamitous). A set of recommended parameters to achieve this performance are given; in addition, though, the limiter may be used with more relaxed settings for dynamic control not requiring these stringent constraints.

There are a number of controls on the HD P3's output limiter which are unusual or seemingly unconventional. But first, the easy ones:

THRESH—The signal level which output level is desired not to exceed.

ATTACK—The period of time an applied signal must instantaneously exceed the Threshold in order to reduce the output signal down to the threshold. 0.5mS is recommended for zero-overshoot limiting.

RELEASE—The time the limiter's gain takes to recover from a typical degree of limiting once the signal no longer exceeds the threshold. 50mS is recommended for transparent zero-overshoot limiting.

GAIN—This is used to adjust the output level of the limiter; a typical use is if it is required to have the limiter "brick wall" at an effective threshold of +20dB (which equates to 0dB full-scale digital). The limiter's threshold would be set at say +8dB (allowing 12dB "headroom" for the limiter to operate without constriction), and then +12dB of gain to be added using the "Gain" control.

CLIP ENABLE—When enabled a zero-attack-time limiter at the set threshold level is activated, beyond which no overshoot is then possible. It is used to contain transients shorter than the limiter's attack-time and other such anomalies that escape the limiter. Used as such, it is sonically invisible.

Warning: "Clip" is intended to be used in conjunction with the limiter; serious distortion can result otherwise. Alone it can be anything other than sonically invisible.



CLIP BACKOFF—The indicated threshold on the output limiter represents the peak output level; it is actually the level of the clipper. Ordinarily, if the limiter is running with it's default values for “look-ahead zero-overshoot” (0.5mS Attack, 50mS Release) the limiter will catch pretty much everything leaving the clipper to take care only of the occasional sub-attack-time transient and such. If the attack time is lengthened, however, longer transients will escape through it; at the longer attack times (say 50mS or longer) the limiter ceases to be a peak limiter and tends to become more of an averaging detector. (This is readily observed with nearly any peak limiter - the peak output increases by 6 - 10dB, or even more depending on source material, when the attack time is increasingly relaxed.)

Under this condition - long attack time, for a less “tight” sound - peak signals many dB higher than the threshold will be allowed through the limiter. If “Clip” is enabled, distortion as these are peremptorily removed can occur. The “Backoff” control allows the limiter’s threshold to be reduced in relation to the clip threshold (set by “Threshold”); this affords the running of looser limiter attack times whilst retaining the absolute maximum output limit set by the Clip.

In general, it is set by ear; like many other settings, the degree of clipping which is tolerable is truly subjective; no amount of blinky-lights or cute graphics can help with this decision. Naturally, looser limiter attack times often sound better, and the clipper helps maintain peak level control; the result though is less “loud” in comparison to the limiter set up for zero-overshoot. A judgment call.

The “Backoff” control, which adjusts the limiter’s threshold to a value lower than that of the clip level (set by “Threshold”), allows one to run looser attack times (often less “constrained” sounding) without the resulting excess level clipping, or clipping too regularly.

POSITIVE MODULATION—This is the percentage by which the positive-going output level can exceed the negative-going, represented by the output limiter’s “Threshold”. The FCC presently allow 125% for AM senders in the US (some other countries being more liberal). This affords a noticeable increase in on-air loudness.

The positive-going Limiter and Clip threshold is increased by the indicated amount over the negative-going threshold, which remains fixed at the value set by “Threshold”.

For all normal purposes, this control should be set to 100%.

(Should be ignored for all applications other than AM broadcast, except for the bizarre experiments of obscure PhDs somewhere in academia.)



Output Limiter: In depth

To achieve guaranteed lack of overshoot, advanced look-ahead and up-sampling algorithms are employed, to respectively catch most overshoots before they occur, and mitigate untoward “aliasing” effects (“grittiness”) that plague lesser digital limiters. When set up correctly, the limiter will not allow excess output at all, and do so gracefully and sonically transparently.

“Look-ahead” refers to the technique of fractionally delaying the input signal such that the limiter’s detecting sidechain has a chance to react, generating an appropriate control signal at the exact time it is required to control the passing audio signal. A short delay is used so as not to incur problematic additional latency, and also to minimize “escapees”. These refer to impulses and transients which are shorter than the attack time of the limiter, and which, obviously, cannot then be accurately captured and controlled regardless of the look-ahead technique.

The longer the attack time, the more energy escapes the limiter; at (not very great) extremes, a clipper - by definition zero-attack time - allows no escapees, while a 100mS attack time is essentially an averaging detector, allowing sometimes wildly high instantaneous peaks through.

To this end the HD P3’s look-ahead limiter with attack times shorter than 1mS dialled in (0.5mS is the specified and recommended value) captures almost all transients found in program material; the downstream zero-attack-time limiter - “clipper” - takes care of the rest. A release time of 50mS (recommended) on the limiter will catch and control signal peaks quite transparently; by itself it can add a considerable amount of “LOUD”, but at the cost of the well-known sonic detriments that come from excessive broadband limiting – and which are the reasons for multiband processing in the first place! It is recommended that the output limiter just be used to control signal peaks just a few dB deep - to this degree it is quite transparent. Deep processing should be left to either of the triband processors.



AES Output 2 Delay

The HD P3 has two physical AES output jacks. The second output may be delayed relative to the first, by setting a delay value in milliseconds using the Coarse and Fine delay controls. The delay function is intended for use in applications where AES output 1 connects to the digital HD air chain and AES Output 2 connects to the analog FM air chain.

COARSE DELAY - this slider is used to set the approximate amount of delay required at the AES 2 output. The slider's range is from OFF to 10 seconds of delay adjustable in 50mS increments.

FINE DELAY - this slider is used to “tune” the amount of delay applied to the AES 2 output. The slider's range is either OFF or adjustable from 6.0 to 49.9 milliseconds of delay in 100 microsecond increments.

DELAY UNPROCESSED - checking this box routes the AES input audio directly to the AES-2 delay circuitry, effectively bypassing the primary processing chain. The signal available at the AES 2 jack is then simply the AES input plus any added delay.

Entering Delay Values

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 *Ctrl* key on your keyboard while adjusting a delay slider to effect precise, incremental steps. As indicated above, note that the minimum amount of useful delay is set at 6mS, just above the HD P3's latency.

HD P3 Features for AM Broadcast

The HD P3 is an ideal AM broadcast airchain processor given the inherent multiband and broadband capabilities and a number of key AM-specific features:

Brick-wall Bandwidth Filters.

These are accessible on “Output”. Three bandwidths are available:

4.5kHz—common where 9kHz channelization is in use (outside the Americas).

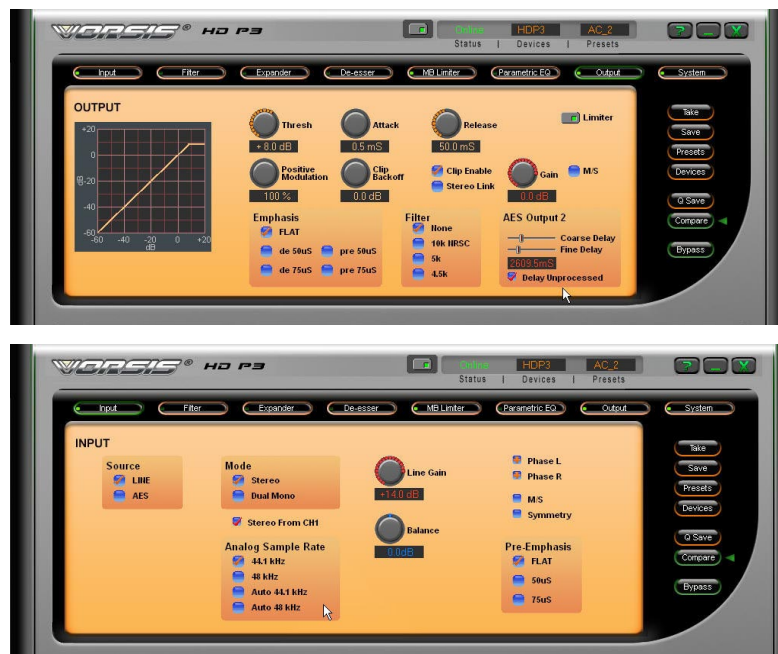
5kHz—required should an AM station be simultaneously broadcasting IBOC, and also recently proposed as at least a “night-time” bandwidth standard within the US, aimed at reducing adjacent-channel chatter.

10kHz—the US NRSC AM bandwidth standard.

Emphasis

It is common to add a rising gain-vs.-frequency emphasis to combat the inevitable high-frequency rolloff of the IF filters in AM receivers. The NRSC standard, for use in conjunction with the above 10kHz brick-wall filter, is for use of a modified 75uS pre-emphasis, conveniently available on your local friendly HD P3’s “Input” or “Output”. The “modification” consists of a gentle rolloff of the pre-emphasis slope approaching 10kHz, and with the HD P3 may be nicely emulated by applying the low-pass filter (“Filters”) set to a frequency of approximately 12kHz.

Truth be told, the niceties of matching exactly such a paper specification get substantially blown away when complex processing is used, but arranging for a non-excessive rising trend in response either by pre-emphasis, multiband processing, or EQ to counter receiver IF “droop” is a good idea; the brick-wall filter will keep sideband energy happily within the mandated spectral mask.



Asymmetric Output Peak Limiter

The look-ahead output limiter in conjunction with its zero-attack-time “clipper” ensure zero overshoot, and allow one to get very close to sender linear modulation extremes with confidence. The limiter’s “Threshold” represents the negative-going peak level, whilst the positive-going threshold is separately controllable to afford up to 150% positive modulation. (The current US limit is 125%).

Bear in mind that if set at say 130%, that is 130% of whatever depth the negative-going level is set to in the real-world on the actual sender; if that happens to be 90%, then the “actual” positive modulation will be 90% of 130%, or 117%. For a typical real-world depth of 95% negative modulation, the “Positive Modulation” control would need to be set at 131% to achieve actual 125% sender positive modulation. Confused yet?

Pos. Mod. Dial % = (100 / actual neg. mod.%) * desired Pos. Mod. %

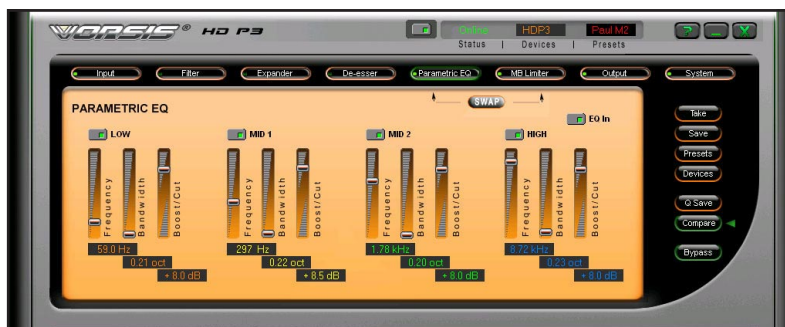
The closer to 100% negative modulation one has the nerve to achieve, the closer the displayed and actual positive modulation numbers will agree.

Mono

In the event that there is no mono program source available from the console / program suite, the left and right signals may be applied to the appropriate inputs on the HD P3; the “Input” “M/S” switch will sum the left and right sources to mono in the “Left” or “A” path of the unit, which can be set up for mono processing, the “Right” or “B” path being ignored.

Parametric Equalizer

This powerful four band parametric equalizer is selectable to be prior to or after the compressor (“pre” or “post”).



Multiband Processing

The three-band dynamics section is the keystone of the HD P3. Two different styles of multiband processing are available in the HD P3 II, entitled loosely “Production” and “Airchain”. The former uses a very high-quality soft-knee compressor for each of the three bands, and is intended to achieve excellent transparent control over “raw” program sources, whether musical or spoken voice. The emphasis is on using the strengths of multiband processing to render a surprising amount of dynamic control “invisibly” and with a minimum of audible artifacts.

“Airchain”, on the other hand, is geared to aggressive attention-grabbing processing, as for better or worse is a feature of the broadcasting landscape. Its reason for existence is to be loud. There are three frequency bands comprising “Climps” - extremely clean zero-overshoot limiters each followed by zero-attack-time “clippers” - which can dramatically reduce the peak-to-average ratios of the audio within their passband, and hence substantially increase their apparent loudness. A choice of “clip” styles allows the use to be better tailored to differing types of program material.

The choice between these two styles of multiband processor is made by check-boxes on the ‘Multiband’ screen of the GUI, and as a “ProcMode” selection under “Multi” on the unit itself. A description of each follows.

Multi-Band AGC/Compressor - “Production”

This affords both highly transparent compression free of the artifacts endemic to normal compressors, and yet at the other extreme wild dynamic equalization effects.

It was felt an unnecessary disservice to “dumb down” the control of this compressor; all the “handles” allowing full control over it are made available. Yes, at first glance the wealth of controls can seem overwhelming; cognizant of this, pre-cooked presets and the addition of a couple of controls have made the multi-band compressor simpler to use, with little need to delve into deeper mechanics unless really desired.

The basic operational model for the compressor is that of the classic UREI 1176 or DBX 160 units of yore; the compressor “core” is enclosed within an input level, or “Drive” control, and an output level, or “Makeup” control. Simply, the depth of compression is adjusted by driving the compressor harder or more gently using the “Drive” control, while the output level is adjusted accordingly with the “Makeup” control. These controls are at the beginning and end of the top row of controls on the GUI’s compressor screen, so are easy to find.



The wide variety of presets supplied with the HD P3 have compressor settings ranging everywhere from slight to brutal; it is recommended that some time is spent calling in different presets and playing with the “Drive” and “Makeup” controls on each to get a feel for their effect on program material. From this will eventually spring the desire to explore the compressor’s entrails.



Why three bands? Even the best of broadband compressors (including the ones that play games with their sidechains to mitigate the effect) suffer from auto-modulation; this most usually manifests itself as heavy low-frequency information (kick drum, bass) audibly controlling and pumping the level of higher frequencies. Splitting away the bass region into a separate compressor from the rest of the spectrum gets rid of the worst of this effect (two-band unit) but similar, if lesser, issues between mid and high frequencies then become plain. The advantages of a second split between mids and highs (three-band) primarily stem from the highly tempting ability to treat the mid frequencies (by and large the intelligence-bearing frequency range) differently from high frequencies. Of course this line of thinking can and does extend to five, six or more band processors; three was felt to convey most benefit, the most “band for the buck”, for the Vorsis HD P3’s intended applications.

Multi-band processors have the reputation - born of being able to apply far deeper and more aggressive compression than broadband compressors for the same amount of sonic “damage” - of being LOUD. Yes, indeed, as a bus compressor the HD P3 can be persuaded to make a VU or PPM meter stand still, but the unit’s greatest strength is its ability to provide firm and positive dynamic control over originating source material quite transparently.

The heart of the three-band compressor is a highly regarded full-blown soft-knee compressor very similar to the design used as a broadband compressor in Wheatstone Corporation’s high-end TV and radio broadcast consoles. As such, the design’s ability to get deep compression and level protection without “funny noises” was paramount. A good start for a multi-band processor.

In the HD P3’s compressor, the input signal is fed through a crossover, producing nominally High Frequency, Mid-Frequency and Low Frequency bands; the high-accuracy phase-accurate crossover frequency points are determined by the “HI X” and “LO X” controls; these are vertically disposed on the right-hand side of the GUI’s compressor screen. These crossovers are fairly gentle, so as to minimize odd-sounding spectral transitions between the frequency bands.

These three bands are then fed into three identical compressors, signified blue for HF, yellow for MF, and red for LF on the GUI. These colors are also used on the dynamic graph, indicating gain-reduction by

frequency. Each of these compressors has the normal controls to be expected of a good compressor. In order:

THRESH — Threshold, the level (within the frequency band) above which gain reduction is applied.

RATIO — The proportion by which a signal exceeding the threshold is reduced in level. “3:1” means that a change in signal level above the threshold by 15dB will be reduced to only a 5dB change in level.

ATTACK — How quickly the compressor responds to a signal exceeding the threshold.

RELEASE — The speed with which the compressor recovers as the exciting input signal reduces or disappears.

These all act just as would be expected on an ordinary compressor. The nature of a multi-band compressor means that slightly more complex approaches to inputs and outputs have to be employed, paradoxically to make the unit simpler to operate. In particular, the relationships of “Drive” and the three “Thresholds”, and again “Makeup” and the three “Trims” will be explored.

On the GUI’s compressor screen it is no accident that the three bands’ “Threshold” controls are in the same vertical line as “Drive”, and that the three bands’ “Trim” controls are in line under “Makeup”.

In the HD P3, the three traditional “Threshold” controls may be regarded as simply offset trims per band of the overall “Drive” control. In other words, it is the **differences** between the three threshold values that are far more salient than their absolute values, which get subsumed by the overall “Drive” control. For example, it is common (since there is generally less energy at higher frequencies) to want to reduce the HF band’s threshold (i.e. make it compress sooner) in relation to the lower bands; in that way all the bands are more likely to start compressing at the same time with applied program material. Again, it is the three thresholds’ relative differences that are important, not their absolute indicated values.

Similarly, the three bands’ “Trim” controls may be regarded as offsets to the overall “Makeup” control; it is usual to need to adjust a band’s output in relation to the other two for best tonal balance, or lunatic effect.

The power of these approaches is that overall amounts of compression and output level may be adjusted without worrying about disrupting the hard-won internal balances of the three compressor bands settings.



Transparent Compression

This is for applications where compression is necessary, but all effort needs to be made to make it transparent, that there be little clue that any compression is taking place. Multi-band compressors can be pretty good at this, but nevertheless require careful application.

First consideration is that the sound, the spectral balance, should not seem to change regardless of the depth of compression, or in comparison to out of compression. To this end, the thresholds of the compressors should be adjusted such that with the expected type of program material (Speech? Classical? Jazz?) the compressors should all start to act simultaneously. Differing source material types will have differing overall spectral balances, and need adjusted thresholds and crossover points accordingly to achieve this hoped-for simultaneity.

Compressor ratios should be gentle, 1.5:1 or 2:1 outside, and the same across the three bands; attack and release times should be fairly relaxed - mid-scale on the controls is not a bad start.

The band output “Trim” controls will need to be carefully adjusted such that a natural spectral balance when in compression is not dissimilar to that in the unprocessed source material. It won’t take much, a few dB here or there across the three.

For pre-recorded music, a subtle “sweetening” effect may be introduced by reducing the mid “Trim” just a few dB, and adjusting the crossover frequencies to achieve a gentle dynamic “smiley” curve.

With nearly everything except modern music, the center of energy is in the mid-range: the crossovers are likely best disposed to bracket this energy region, rather than divide it. This is true with speech especially, where the 200Hz to 4kHz region carries the brunt of intelligence. Disrupting this critical spectral area with a crossover (implying differing processing either side of it) is asking for an unnatural effect.

Compressing for effect

No.1 - ignore all the guidelines under “transparency”.

A good example of compressing for effect can be the transformation in spectral balance and indeed overall sound between when the source signal is below threshold, and when the source is loud enough to be well into compression. Such is the “Wolfman Jack” effect. (For those unaware, but to hopelessly understate his history, this character was a disk-jockey between the 50’s and 80’s with phenomenal command of microphone technique. When speaking quietly, his voice was deep, rich, gravelly and envelopingly warm, but as he started to wind up eventually to screaming, it transformed into a thin hard shriek.) Such is an effect, for want of the native human talent, for which a three-band compressor was made.



An approach is to arrange for the three bands to have descending amounts of compression; the LF band would be highly compressed at a high ratio, with the LF crossover set to encompass the lower speech fundamentals. The mid threshold would be set higher and be at a middling ratio, and the HF threshold set to trigger a touch higher yet and with a gentle slope, with the HF crossover low enough to capture a voice's higher elements and fricatives.

The effect would be that the highly compressed LF range would predominate at low levels, but would be held down relatively below the MF and eventually the HF bands as the source level increases. At high levels, the HF band would predominate.

In general, when compressing for effect (any effect), there are a couple of items worth bearing in mind:

- Listen intently as the crossover frequencies are adjusted; moving them even just an octave can have huge effect. Crossing over “LO XO” below 100Hz can lend a lovely deep “Deutsche HiFi” feel to the bass, but leaves the low-mids at the mercy of the mid compressor (or vice-versa). Likewise the “HF XO”; it can be a trial finding the right crossover point. Placing it high (8kHz, 10kHz) can lend a real sweetness to the extreme highs, but at the expense of potential loudness to be gained by sharing the mid-frequency compressor's load.

- Be aware of over-compressing the high-frequency band. This can lead to the “Gabriel Effect”, named after Peter Gabriel's song “In Your Eyes”, in which a triangle is tastefully laid way back in the mix. If animals flee and eyes water, it is a possible indication of having overcooked the highs somewhat.

- Steep compression slopes, say 3:1, 5:1 or even higher, and short attack and release times, can overall lend more “urgency” to the sound, and tend to make it sound louder than its actual level would imply. Not too short on the attack time, though - a feature of the compressor design used is that it can act as either a peak-sensing detector, for use in peak limiting, or as an average-sensing detector for compression. Attack times shorter than say 10mS are by default starting to make the compressor act more as a peak sensor. The end-effect is that the output actually starts to get “quieter” as the attack time is reduced below that point, since the peaks rather than the average energy are being captured; the peak-to-average ratio can be anywhere from 6 to 12dB, depending on the source. Although peak-sensing can sometimes be interesting, it is not necessarily best for Loud. If Loud is the goal, use of the Multiband Limiter option (“Airchain”) is the route.

AGC

The AGC is an additional sidechain operating in conjunction with each of the three compressors; generally the AGC has much slower integration times for attack and release, so that it responds more to the medium or long-term energy of the program material in a given band, rather than more typically energetic actions of the compressor. Its time-



constants are controlled by “AGC Attack” and “AGC Release”, in the middle of the top row of controls on the GUI’s compressor screen. A single set of time-constants for all three bands was felt more than adequate, and has proven so. This single set of parameters does not imply that the three bands’ AGCs are coupled in any way; all three operate independently.

The AGC may be used alone (by enabling it and turning off the compressor) for gentle control of inconsistent input signals. It can also be used in conjunction with the compressor; in this case, the AGC action rides “underneath” the compressor; depending on the time-constant settings and character of the program material, the AGC can follow the average long term level changes and create a gain-reduction “bed” some 12dB to 6dB under the compression gain-reduction peaks. The AGC rides the general level, the compressor processing for effect takes place on a consistently controlled signal.

On sudden application of an input signal, the faster compressor’s attack captures the onslaught, with the AGC eventually catching up. On release, the effect is identical to the much-vaunted “two-slope release” of classic compressor units such as the Audio and Design F760xrs and Joemeek SC2. On departure or reduction of the input signal the usually faster compressor release predominates until its gain-reduction contribution falls below that of the AGC, whose much slower release rate takes over. A big advantage of the compounded processes is that on normal program material, the compressor does not have to “move as far” to capture signal peaks, so reducing the “snatching” which can occur at deep compression onset.

For most purposes the combination AGC/compression is the most transparent; if the intention is wild effect, deriving all the gain-reduction from the compressor alone is probably better.

Multiband Limiter - “Airchain”

No bones about it, and no pretence: this processing section’s sole reason for being is to make things **LOUD**. As with the three-band compressor, it was felt that ultimately the user is better served by having all the “handles” on the processing available. There are a different set of controls for this Multiband Limiter, although the two truly important ones operationally remain similar to the “Production” style processor: “**Drive**” which adjusts overall depth of limiting and gain-reduction, and “**Output Level**” which sets the overall output from the multiband stage.



These two, assuming that initial operation is going to be from an existing preset, are the only ones of concern, at least until confidence has risen and the control screen no longer seems as intimidating as Concorde’s flight deck. The specific controls for the three bands are ranged horizontally across the GUI screen:

DRIVE — Yes, Virginia, there are four controls called “Drive” on the GUI. The top one is the important one - it is the master drive for the whole three-band section. The three below it are for the high-frequency, mid-frequency, and low-frequency processing bands. These allow the relative amounts of signal drive into each band to be varied; the range is +8dB to -20dB, more than enough to (over) emphasize a band, or effectively remove it. In practice these should rarely be more than a very few dB adrift of the nominal **0dB**.

ATTACK — The period of time an applied signal within the band must instantaneously exceed the threshold in order to reduce the output signal down to the threshold. **0.5mS** is recommended for zero-overshoot limiting. Free-form range is 0.2mS to 1Sec.

RELEASE — The time the band’s limiter’s gain takes to recover from a typical degree of limiting once the signal no longer exceeds the threshold. **50mS** is recommended for transparent zero-overshoot limiting. Overall range is 33mS to 1Sec.

BACKOFF (CLIP BACKOFF) — The discussion on “Backoff” as applied to the Output Limiter (on page 3-17) is hereby set as homework. Review is strongly recommended since it is directly appropriate to the limiters used in this section. For the recommended zero-overshoot / zero-attack settings, “Backoff” should be set at **0dB**.

In short, “Backoff” is the amount that the limiter’s threshold is reduced below it’s downstream zero-attack-time “clipper” threshold. It becomes relevant if the attack time is made greater than approximately 2mS. Slower attack times allow greater peaks to ‘escape’ through a limiter; reducing its threshold with “Backoff” can prevent too much signal being clipped, which could potentially sound unpleasant. The range is 0dB to -10dB.



CLIP “SOFT” / “HARD” — The zero-attack-time “clipper”, is downstream of the band’s limiter. “Hard” clipping is, in effect, saturation. It does not affect the signal at all unless its threshold is breached. “Soft” clipping starts bending “gracefully” well before the saturation level, reducing the peak-to-average and hence increasing the apparent loudness of many program types. And also making a mushy mess of some other program types, which is why there’s a choice between the two styles.

TRIM — There is a ‘Trim’ control for each of the HF, MF and LF bands. They control the relative amount of each band contributing to the overall section’s output; the range is $\pm 18\text{dB}$, plenty enough to emphasize a band, or effectively remove it. In practice these should rarely be more than a very few dB adrift of the nominal **0dB**.

The “Trims” are probably the most powerful controls in determining the ‘sound’ of the Multiband Limiter; they can be thought of as Big Knobs on a Gigantic three-band EQ. Paraphrasing, in general these need to be handled with care, and small adjustments make a big sonic difference.

There are a number of controls common to the three bands, or affecting the overall Multiband Limiter section: “AGC Attack”, “AGC Release”, and “AGC In”:

These control the operation and time constants for a three-band AGC system, which when selected underlies the operation of the three bands’ limiters. The time constants are common to all three bands, but this is not to imply they are coupled; the three band’s AGCs operate independently. The time constants are usually set to be very significantly slower than those of the limiters; the intention is to provide an average-energy gain-riding, along with the instantaneous level control of the limiters.

In combination, the triband AGC and the triband “Climps” achieve enormously deep amounts of gain-reduction far fewer ‘breathing’ artifacts and such, as would occur with a corresponding depth of solely limiting. It may be thought of, and works in fact, as a few dB of instantaneous limiting riding atop the AGC’s background gain-reduction.

This way of operating is highly suited to airchain processing; in addition to the requirement for the processor to do its “loud magic”, compensation for sometimes wickedly askew signal levels from the studio complex has to occur; these can often be 20dB adrift moment-to-moment, disc-to-disc. The AGC does the auto-levelling, the limiters do the “loud”. Obviously, in order to accommodate very low occasionally expected source signals, for the normal or higher levels the airchain processor will already be very deep into gain reduction, far more than the limiters need to work effectively. In

days of old, and indeed still today, it was common to use a ‘pre-processor’, to ensure that a consistent level was sent to the limiters so that they could operate in their “sweet spot”, above and below which their sound suffered for a number of reasons. The triband limiters in the HD P3 have no such limitations and will happily work 30+dB deep, but that can sound breathtaking when they “come up for air” in a program gap; AGC and “Freeze” (below) solve that issue. The integrated AGCs in the HD P3 dispense with the need for pre-processing entirely.

What should be borne in mind here is that “very” deep gain reductions will be the norm, are expected and encouraged, using the Multiband Limiter; they are the combination of that incurred normally from a pre-processor / AGC and that of the main airchain processor. Be not of faint heart when the gain-reduction meter zooms well beyond 20dB.

“Freeze” — An adjunct to or alternative to the expander, for use specifically in the Multiband Limiter. If the freezer senses the input signal’s instantaneous level to the section dropping below the threshold set by the “Freeze” control for more than 1/4 second, it applies a *very long* time-constant to the AGC (approx 30 seconds). This prevents the gain suddenly rising precipitous amounts, causing “breathing” and such noises, and also means the AGC and limiters’ sidechains have much less “catching up” to do when the signal reappears.

A typical threshold is -40dB; this is below the level of any meaningful signal, and above room noises and well above system noises.

Loud! - The life and times of the Airchain Multiband Limiter

No pretences here about transparency or naturalness; the name of the game is Loud. This is where the Multiband Limiter (Airchain) option excels, Aggressive settings, “Drive” high, and short attack and release times.

MIDS — The ear is most sensitive at mid frequencies - aim to compress hard in this region, more so in fact than elsewhere. The less compressed “space” in the adjoining bands can add significantly to an illusion of loudness. Consider EQ’ing in this frequency region, too. EQ’ing pre-compressor has the benefit that an amount of EQ which would be impossible otherwise (through eating up headroom and just plain sounding horrid) just happily gets squashed by the compressor, adding a density and power rather than a screeching sense of over-EQ’ing.



BASS — Bass can be recovered, indeed exalted. After the three-band limiting, there is often “room” to bring up the level of the bass, which *definitely* lends a sense of increased weight and loudness to the whole; it is probably this region which benefits the most from such processing. Subtle EQ’ing post-limiting - rather than simply turning up the trim - to bring up those specifically required elements of the sound, can add worthwhile enhancement.

HIGHS — As with all rules which are merely to be broken, it is a good idea not to squash the highs too much. It generally gives one a headache. Fortunately the nature of energy distribution by frequency in music is such that there is less signal level in this top band than the other two, it doesn’t drive its limiter as hard and remains fairly “natural”. Since as a consequence less gain-reduction is applied to it, the high-frequency “Trim” control needs usually to be reduced a few dB in relation to the LF and MF in order for the output spectral balance to be somewhat sane.

Creating a “sound”

Talking of spectral balance, far more than anything else, the “sound” of the processing (that with which you are trying to impress the Program Director) is controlled by the three “Trim” controls. As earlier alluded, these act as three knobs on a giant three-band EQ. Although there is very wide latitude here, an effort should be made to keep the average of the trims’ settings around 0dB, with overall lift-and-cut being done by the “Output Level” control. By the time the signal has passed through the multiband limiter, each band’s level is well-and-truly fixed in relation to the other two, and this relationship is at the mercy of the “Trims”. Nearly anything can change spectrally ahead of the multiband (including EQ - subtle EQ in particular) and its effect will as good as disappear, with the three-band’s relationship overwhelmingly impressed upon it. The consistency of this sound is one of a multiband processor’s greatest strengths. Considerable thought and listening to (unprocessed) material to find a sound one would like to emulate with the box is well worthwhile; it is always easier working towards a known sonic goal. Post EQ’ing to provide detail shading to the spectrum, having used the “Trims” as broad-brushes, is a good approach.

Deep Gain-reduction

One should not be shy with depth of limiting in this section; a little is absolutely not enough. Gain-reduction depths (AGC plus Limiting) should be of amounts to make genteel ladies swoon and fly totally against one’s better nature, previous training, and instinct. There is a master “Drive” control there: Use it. There are good reasons for this:

For all normal program material intended to be heard, the multiband section should be in limiting. This way the known and desired sound of the processor is definitely having effect on the signal. This NOT like normal “Production”-style or “Bus” compression where one is merely

interested in reducing the dynamic range and preventing overload - that's what the "Production" compressor is for. The multiband limiter is used to ensure that whatever is intended to be heard is heard, and heard in the way one wants it to be heard. In conjunction with their integral AGCs they allow the aggressive processing necessary to increase apparent loudness at the same time as regulate and compensate for a wide variety in source levels. Under normal circumstances, this can result in jaw-dropping total gain-reductions. Just remember they are in effect the combined gain-reductions of a "leveller" or pre-processing AGC plus that of the limiting stages, which with other solutions would be separate, but total to no less.

Secondly, be aware that the sound whilst in limiting is likely to be radically different to the unprocessed sound, once it drops out of limiting. So, unless this is a desired effect (and it can be a valid and interesting one), ensure the signal stays in limiting if you intend it to be heard!

Setting Up the Output.

Once a satisfactory spectral balance has been achieved using the individual bands' trims and possibly post-EQ, it is necessary to match the multiband limiter's output with the rest of the world. It is presumed that a desired output level from the box has been decided, and that the Output Limiter has been set up to capture at that point. Let's say it is +8dB. The Output Limiter should be set to that threshold with zero-overshoot settings (default, but 0.5mS attack, 50mS release). With normal program material running through the multiband, and gain-reductions plumbing untold depths, the "Output Level" control on the Multiband Limiter should be carefully adjusted such that the Output Limiter is active by just a few dB, say 3dB, or 5dB at most. The output of the Multiband, although each band is of a known level, is a mix of the three, and so is instantaneously uncertain within limits. Running it a few dB into the Output Limiter reduces this uncertainty without radically altering the carefully crafted sound.

No-free-lunch department: The downside to all these games is that highly Loud processing although initially mightily impressive can become tiresome, and eventually aggravating with time. There is an inverse relationship between aggressiveness and long-term listenability.

Be aware too that heavy compression is not kind to music which has been low-rate perceptually encoded (e.g. <128k MP3); the encoding artifacts start to stand out, adding to an already probably grating experience.

System

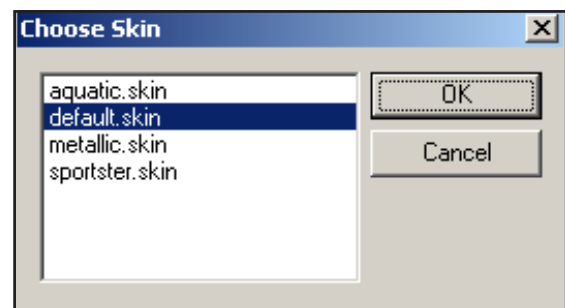
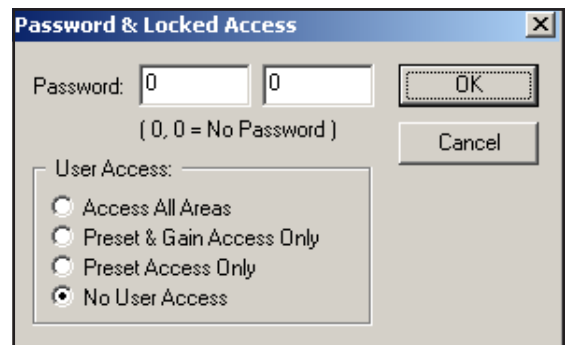
This tab contains miscellaneous system wide functions such as input settings protection from presets, security (passwords etc.), skin selection ("look" of the GUI).

INPUTS — "Inputs follow presets", if unchecked, allows the HD P3 to ignore input source (Line/AES) and input gain settings within a preset. The signal processing setup within a preset may be of interest, but the preset may have been made using a different source/gain than that which is hard-wired into the HD P3.

OUTPUTS — "AES Outputs are Split" - Checking this selection causes the Channel 1 processing chain to be routed to AES 1 output jack and the Channel 2 processing chain is routed to the AES 2 output jack. Used primarily when the input mode choice is Dual Mono though some presets utilizing M/S processing utilize the "split".

SECURITY — Allows access unto password-world ("Passwords" button). It is presumed that access to the GUI grants full access to all features and functions of the connected HD P3, internal password-limited access notwithstanding; indeed the password and access levels may be altered freely from the GUI. Clicking "Password" brings up a dialog box displaying the existing pass-code for the connected HD P3; it may be altered by directly entering a new number between 0 and 255 in each field. The level of access outside password protection (user access) may be selected by clicking on the appropriate "radio" button.

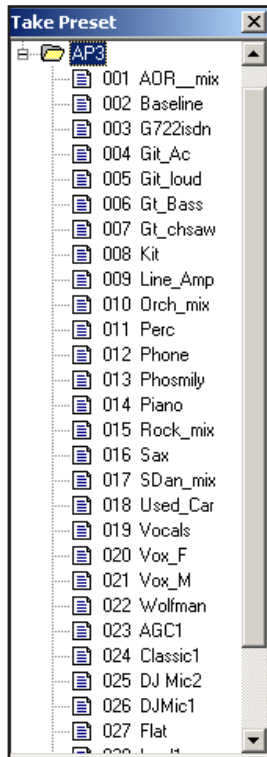
INTERFACE OPTIONS — Clicking "Skin" shows a selection of differing colors and styles available for the GUI. Selecting one and "OK" will cause the screen's look to change in a few seconds.



Side Bar Region

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

Take

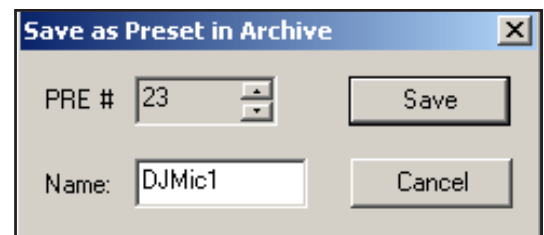


When left-clicked, a “Windows” box appears, showing folders that contain presets, typically “Favorites” and “HD P3”, this last being presets already in place in the presently connected device. Opening a folder (if not already so) lists the presets in order. Double-clicking on a preset brings it immediately into use in the HD P3. 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.*



Save

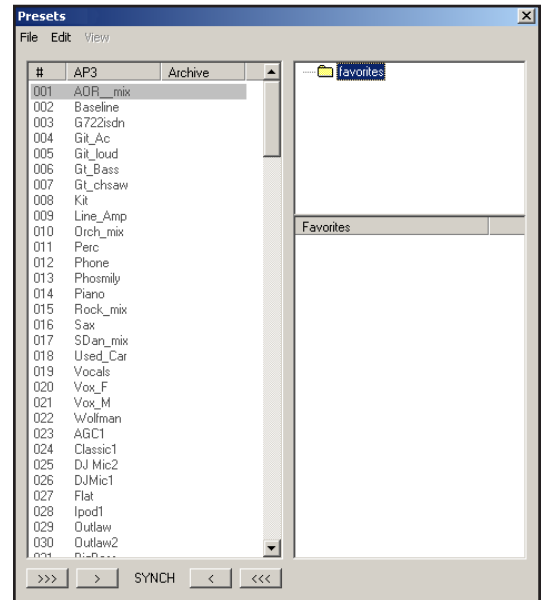
A small “Windows” dialog box appears, which prompts for a name under which to save the present settings of the HD P3. The preset will be saved into the next available empty slot. Alternatively, by nudging the preset number (“PRE #”) up or down, it is possible to save the present settings in place of an existing preset’s settings. Needless to say, this erases, as in loses, as in destroys, the overwritten preset’s contents.



Presets

A three-panel “Windows” box appears. The large panel on the left has three columns, being the preset number, HD P3 preset contents, and the GUI’s mirror archive contents for that HD P3; often the last two are the same or similar. If they are dissimilar, it has to be decided which is in error. The two lists may be synchronized using the “Synch” buttons at the bottom; with these it is possible to move one selected preset from one column to the other (single arrow) or all of them (triple arrows). Treat With Respect! “Windows” file drag-and-drop techniques work to move presets around, too.

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 HD P3 and its mirror archive in the GUI.



Devices

A list of HD P3 devices with which the GUI has had cause to become familiar is shown in a “Windows” box. Highlighting the desired HD P3’s name and hitting “Select” causes the GUI to attempt to connect to it.

In the event no HD P3’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.



Q Save (Quick Save)

This saves the immediate present HD P3 settings in a temporary buffer, allowing continued adjustments to be made without losing a known working point. “Q Save” works in conjunction with . . .

Compare

. . . which toggles between the settings in the “Q Save” buffer and the immediate settings; an arrow indicates which of the two is active on the HD P3 at any instant. This arrangement greatly facilitates incremental adjustments while building a “sound” or a new preset.

Bypass

Allows a “sanity check” by bypassing the signal processing in the box altogether; the input and output gain controls remain active.

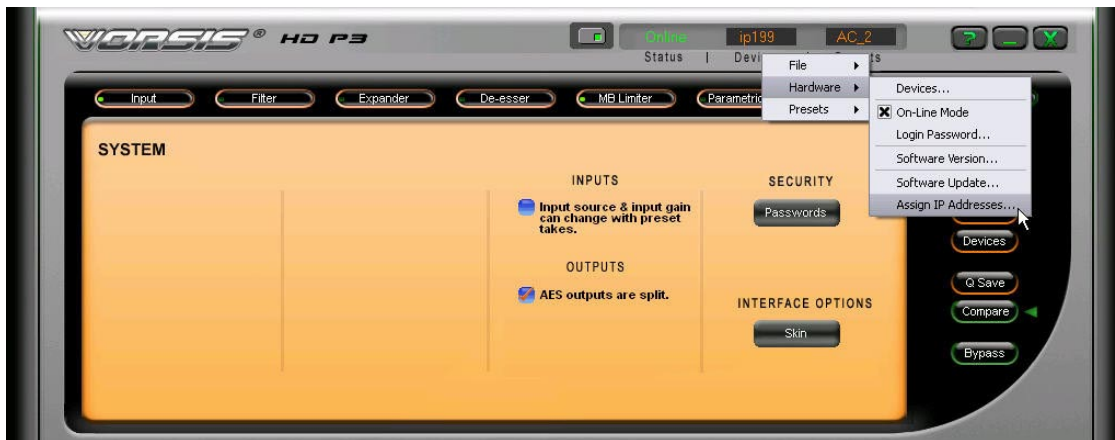
Title Bar Region



Along the top edge of the HD P3 GUI screen (in line with the “Vorsis HD P3” 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 HD P3 GUI is capable of controlling multiple HD P3s (“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.

Accessing Menu Options

Right clicking anywhere on the Vorsis HD P3 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 HD P3 control panel. As with many Windows programs, there are multiple ways to access menu trees - go ahead and explore!



Status

This indicates: “Online”, meaning the GUI is in communication with and directly reflects an HD P3; “Offline”, meaning the GUI and HD P3 are disconnected; “Trying”, meaning the GUI is attempting to find the HD P3 on the LAN and connect to it. An adjacent button toggles between “Online” and “Offline”.

Devices

The name of the HD P3 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 HD P3 devices with which the GUI has had cause to become familiar is shown in a “Windows” box. Highlighting the desired HD P3’s name and hitting “Select” causes the GUI to attempt to connect to it.

In the event no HD P3’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 HD P3’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 HD P3 and the GUI mirror archive.

“SYNCH” allows the presets within the HD P3 and the GUI to be made the same. The red color indicates a discrepancy between the actual current HD P3 settings and the GUI’s mirror archive for the preset; this usually reflects that settings have been twiddled with since the preset was “Taken”. At this point a choice should be made: if the original preset is still valid a re-Take should be performed; but if the current modified version is the valid one, the current settings should be saved back into the originating preset.

To paraphrase: There are many ways during normal usage in which one or some of - (a) the actual current settings of the HD P3, (b) the preset within the HD P3 from which the settings originated, and (c) the GUI’s archived version - can become different. A red preset label indicates this. Determine which of the three is actually correct, then make the red flag go away and synchronize the system by:

Differences			
Setting	Ch	Current	Active Preset
Name			Baseline
Num Parameters		138	138
CRC		953D0E3A	2D16ADFA
Input Source	0	LINE	AES
Input Source	1	LINE	AES
Input Mic Gain	0	+20.0 dB	+23.0 dB
Input Mic Gain	1	+20.0 dB	+23.0 dB
Input Line Gain	0	0.0 dB	0.0 dB
Input Line Gain	1	0.0 dB	0.0 dB
Input AES Gain	0	0.0 dB	0.0 dB
Input AES Gain	1	0.0 dB	0.0 dB
Input Gain Offset	-	0.0dB	0.0dB
Input Phase A	-	+	+
Input Phase B	-	+	+
Symmetry	0	Out	Out
Symmetry	1	Out	Out
Filter HPF In	0	HPF OUT	HPF OUT
Filter HPF In	1	HPF OUT	HPF OUT
Filter Notch In	0	NTCH OUT	NTCH OUT
Filter Notch In	1	NTCH OUT	NTCH OUT
Filter LPF In	0	LPF OUT	LPF OUT
Filter LPF In	1	LPF OUT	LPF OUT
Filter HPF Freq	0	80.0 Hz	20.0 Hz
Filter HPF Freq	1	80.0 Hz	20.0 Hz

- a. Current settings correct:
 - Save the current settings back into the originating preset.
 - Copy the newly refreshed HD P3 preset into the GUI archive.
- b. Originating HD P3 preset correct:
 - Re-”Take” the preset into the HD P3.
 - Ensure the GUI archive accurately reflects that preset, copy over if necessary.
- c. GUI mirror archive version correct:
 - Copy the archived preset over into the HD P3 preset.
 - ”Take” the preset into the HD P3.

Notes on “Online” and “Offline” Working

Most often operationally it will be required that the GUI act directly and instantaneously upon the HD P3 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 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 HD P3 itself, which can continue to be processing away obliviously. (Preset list manipulation can be undertaken “Online”, too, but only by those with nerves of steel. One click from potential disaster . . .)

Importantly, any adjustments made on the GUI “Offline” will not take effect until the HD P3 is rendered “Online” again.

It is easy for the HD P3 itself and the GUI to lose agreement; either the HD P3 can have been adjusted using the hardware front-panel, or adjustments made on the GUI, either “Offline” or when the HD P3 and GUI have had a tiff and weren’t talking to each other for some other reason. Extensive tools to reconcile such wayward units are available under “Presets” in the Side Bar screen region.

Invoking a preset - by double-clicking a preset from the box brought up by clicking the “Take” button - does one of two things, depending whether the GUI is in “Online” or “Offline” mode:

- “Online” - the double-clicked preset is immediately sent to the HD P3 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 HD P3’s audio.

Hardware Menu Items

The *Hardware* menu tree may be accessed by right clicking anywhere on the main HD P3 Control Panel. Please note that many of these functions require you to be connected (Online) to a HD P3. 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.

Login Password - opens the Passwords dialog box for editing login passwords.

Software Version - displays the current software versions running on your HD P3.

Software Update - opens the “Choose file to download” dialog box. Only files of type *.vbn are visible.

Assign IP Address - opens a dialog box that allows the user to change the IP address parameters of the HD P3. Please see the *IP Address Configuration* section near the beginning of this manual for complete details on changing the network settings.

Software Updates

Future versions of the HD P3 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 HD P3. Upon completion, you will be asked to restart the HD P3 (cycle power).

Only verified updates provided by Vorsis will work!

I/O Schematic Drawings & Load Sheets

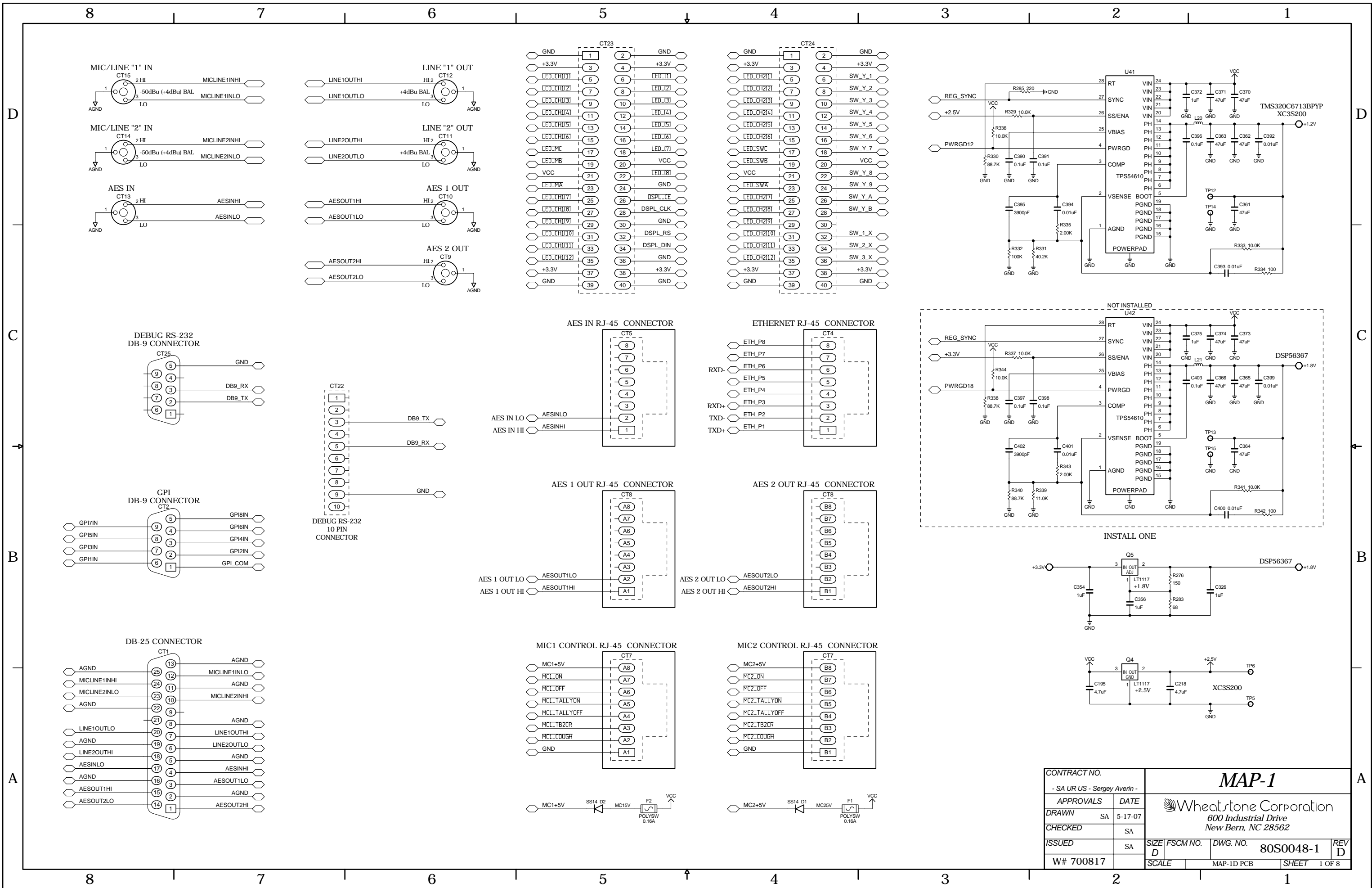
Chapter Contents

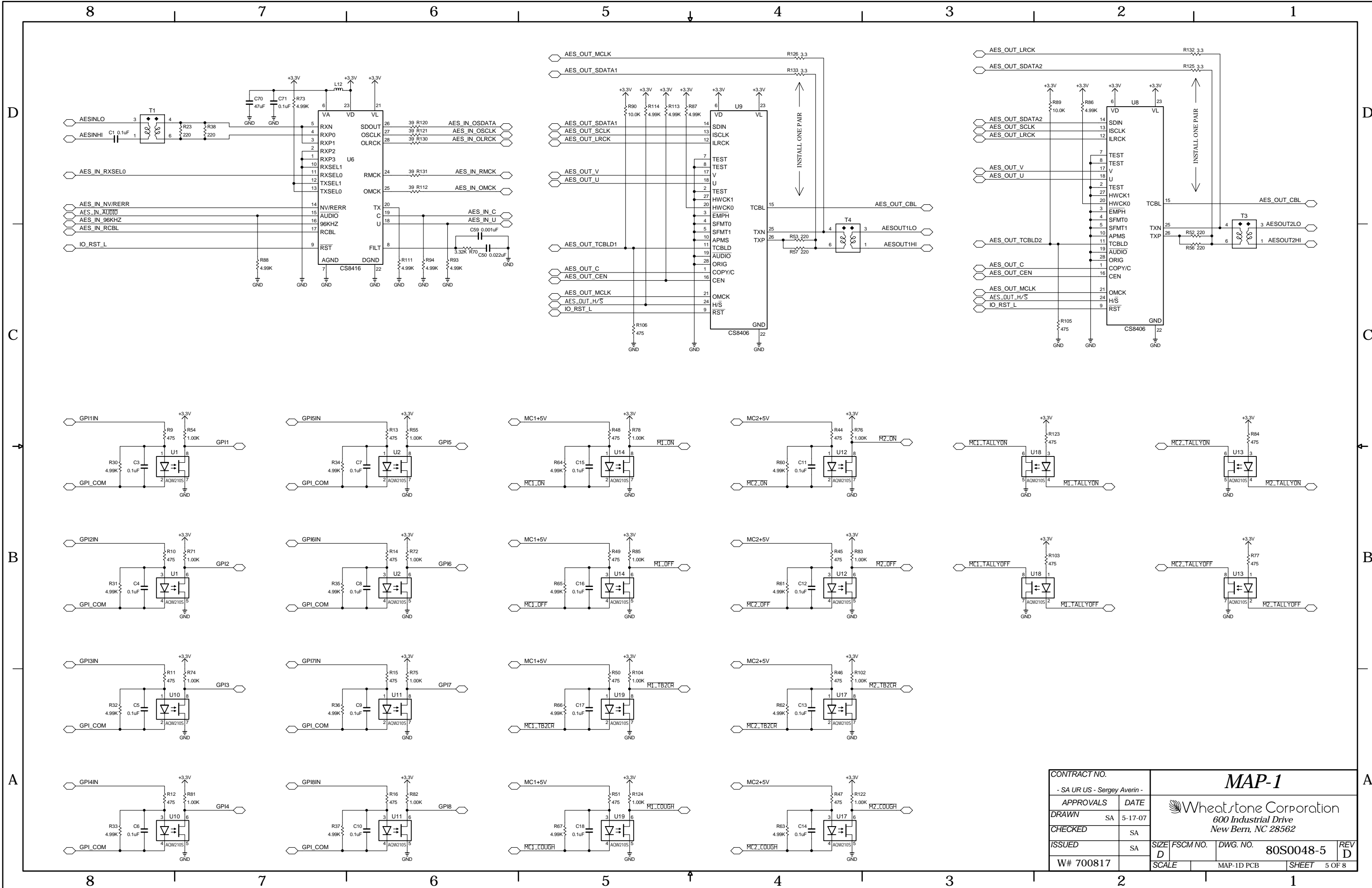
HD P3 Processor Main Card

Schematic	3-2
Load Sheet	3-5

HD P3 Processor VU Card

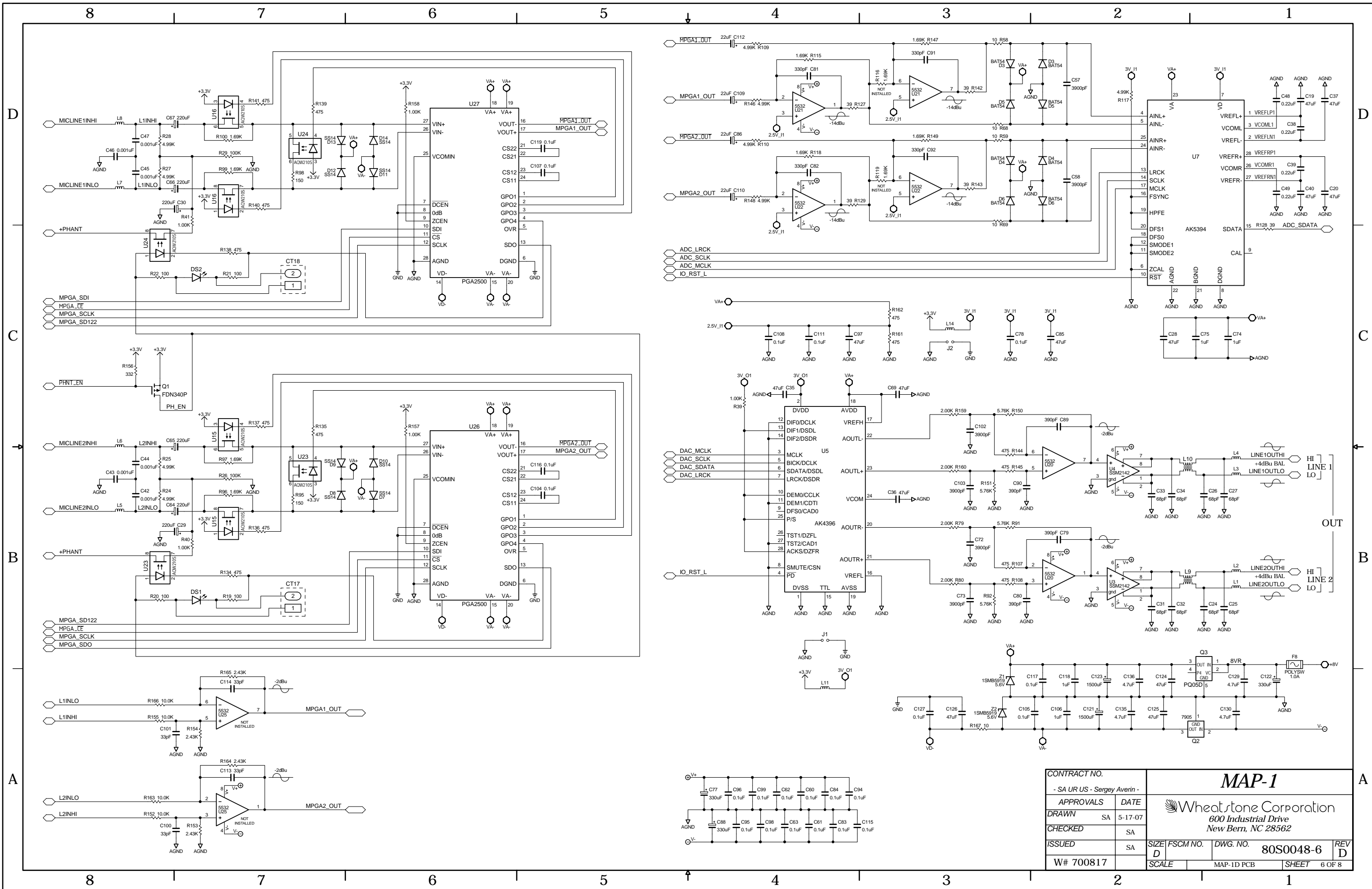
Schematic	3-6
Load Sheet	3-7



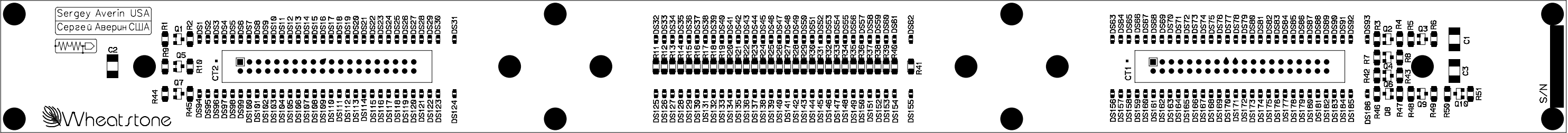


CONTRACT NO.		MAP-1	
- SA UR US - Sergey Averin -		Wheatstone Corporation	
APPROVALS	DATE	600 Industrial Drive	
DRAWN SA	5-17-07	New Bern, NC 28562	
CHECKED	SA		
ISSUED	SA	SIZE D	FSCM NO.
W# 700817		DWG. NO.	80S0048-5
		SCALE	MAP-1D PCB
		SHEET	5 OF 8

HD P3 Processor Main Card Schematic



HD P3 Processor Main Card Schematic



HD P3 Processor VU Card Load Sheet

Appendices

Appendix 1

Parameters, Units and Ranges A-3

Appendix 2

Replacement Parts List A-8

Appendix 1

Contents

Parameters, Units and Ranges	A-3
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Parameters, Units and Ranges.

Approximately following the HD P3's signal path, these are the values and ranges appropriate to each type of processing.

System Level

Headroom:	20dB
Nominal Operating Level:	-20dBFS digital +4dBu analog

Input Gain

a. Analogue Line Input

Type:	Electronic Differential
Input Impedance:	> 10kohm (bridging)
Optimum Source Impedance:	< 1kohm

b. Digital Line Input

Type:	AES3/EBU
Maximum Gain:	+12dB
Minimum Gain:	-12dB
Default Gain:	0dB
Gain Increments:	0.5dB

Filters

a. High-Pass Filter

Filter class:	24dB/oct. Butterworth
Frequency:	20Hz – 1kHz (80Hz)
Defaults:	Off

b. Low-Pass Filter

Filter class:	24dB/oct. Butterworth
Frequency:	1kHz – 20kHz (4kHz)
Defaults:	Off

c. Notch

Frequency:	20Hz – 20kHz (60Hz)
Depth:	> 20dB
Defaults:	Off

Equalization

Four identical, independent bands of parametric equalization:

Frequency:	20Hz – 20kHz
Bandwidth:	0.2 – 3 octaves
Lift / Cut:	+ / - 14dB
Defaults:	All off
	1. 50 Hz, 1 oct., +6dB
	2. 800Hz, 0.5 oct., -4dB
	3. 2.8kHz, 0.66 oct. , +3dB
	4. 16kHz, 0.4 oct., +8dB

De-Esser

Recursive-style true (not broadband) De-Esser.

Threshold:	-10 to -60dBfs (-20dBfs)
Attack:	0.1mS – 100mS (5mS)
Release:	50mS – 500mS (100mS)
De-Ess Frequency:	20Hz – 20kHz (4kHz)
De-Ess Bandwidth:	0.15 – 4 octaves
Defaults:	Off

Expander (“Downward expander”)

Threshold:	-60 – 0dBfs (-40dBfs)
Open:	0.1mS – 100mS (0.1mS)
Hysteresis:	3dB
Hang:	0 – 1S (100mS)
Close:	50mS – 3Sec. (300mS)
Ratio:	1:1 – 1:20 (1:2)
Depth:	0 – 40dB (14dB)
Defaults:	Off

AGC

A common set of parameters control the behaviour of the three independent band-split AGC systems.

Attack:	50mS – 500mS (200mS)
Release:	100mS – 3Sec. (1.5Sec.)
Defaults:	Off

Compressor - “Production”

A three band frequency split compressor/limiter, each band having individual parameters:

Threshold:	-30 to -70dBfs (-40dBfs)
Attack:	0.1mS – 300mS (1mS)
Release:	33mS – 1Sec. (150mS)
Ratio (soft-knee):	1:1 – 20:1
LF Crossover Frequency:	20Hz – 1kHz (250Hz)
HF Crossover Frequency:	1kHz – 20kHz (4kHz)
Drive (Overall “Threshold”):	0 – 100% (50%)
Trim (HF, MF, LF):	+/- 18dB (0dB)
Makeup Gain:	-20 to +48dB (0dB)

Multiband Limiter - “Airchain”

An alternative to the above three-band compressor, this is a three-band peak limiter, each band having individually set parameters:

Drive	0 to 100% (50%)
Attack	0.2mS – 1S (0.5mS)
Release	33mS – 1S (50mS)
Backoff	0dB to -10dB (0dB)
Clip	Hard / Soft (Soft)
Trim	+/-18dB (0dB)
Freeze	0dB to -60dB (-40dB)
LF crossover frequency:	20Hz – 1kHz (250Hz)
HF crossover frequency:	1kHz – 20kHz (4kHz)

Output Limiter

A zero-overshoot look-ahead peak limiter.

Threshold:	-50 to -10dBfs (-10dBfs)
Asymmetric Threshold:	100 – 150% (100%)
Attack:	0 – 100mS (5mS)
Release:	10mS - 330mS (50mS)

M/S “Matrixing”

L/R to M/S (and vice versa) conversion available at both input and output,

Defaults: L/R

Pre- and De-emphasis

Pre-emphasis (50 or 75uS) available at input.

Pre-emphasis or De-emphasis (50 or 75uS) available at output.

Defaults: Off

AM Bandwidth Filter

Available at output: 4.5kHz (Europe)
5kHz (US night and/or with HD sidebands)
10kHz (NRSC for US)

Defaults: Off

Delay

Latency matching delay, FM vs. HD output, available on second AES output and analog outputs: 0 - 10 Seconds, 100 microsecond resolution.

Defaults: Off, 0 Sec

Control

Supplied Windows® GUI software affords control of all parameters, via Ethernet.

Appendix 2

Contents

Replacement Parts List	A-8
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For the most part there are no user-replaceable parts in the Vorsis HD P3. A complete list of available components is shown on the next page. Contact Vorsis technical support for further information.

Vorsis (600 Industrial Drive, New Bern, North Carolina, USA 28562) may be reached by phone at 252-638-7000, fax 252-637-1285, electronic mail "techsupport@vorsis.com".

REPLACEMENT PARTS —HD P3 PROCESSOR

COMPONENT	DESCRIPTION	WS P/N
HD-P3 FACE	RACK FACE ASSEMBLY	"008837"
HD-P3 LOADED CARD	PROCESSOR LOADED CARD ASSEMBLY	"008832"
HDVU-3 LOADED CARD	VU LOADED CARD ASSEMBLY	"008831"
CABLE	50 COND FLAT RIBBON CABLE	"150007"
POWER CORD	7 1/2' BLACK POWER CORD	"150017"
RIBBON PLUG	40 PIN RIBBON PLUG	"250053"
I/O CONNECTOR	DB-9 RIGHT ANGLE PC MOUNT CONNECTOR	"220016"
I/O CONNECTOR	RIGHT ANGLE 25 PIN PC MOUNT CONNECTOR .318 FEMALE	"220120"
I/O CONNECTOR	FEMALE XLR, PCB	"260003"
I/O CONNECTOR	MALE XLR,PCB	"260004"
I/O CONNECTOR	RIGHT ANGLE SHIELDED RJ-45 CONNECTOR	"260049"
POWER CORD CONNECTOR	POWER CORD CONNECTOR WITH PRINTED CIRCUIT TERMINALS	"230071"
MANUAL	OWNER'S MANUAL	"008885"