AP3 Digital Signal Processor

TECHNICAL MANUAL

VORSIS™

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AP3 Digital Signal Processor - 1st Edition

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AP3 Technical Manual

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General Information

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General Information

Introduction

The Vorsis™ AP3 Digital Signal Processor is a highly complex piece of audio processing gear, the first product in a new line of a single-rack space audio processors. This unit occupies one 19” wide rack space (height 1 3/4”), and is 16” 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™ AP3 Digital 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™ AP3, 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 AP3's the rear panel. Six XLR connectors are provided for analog and digital audio input and output, and all the audio connections also appear on the DB-25 connector. Two RJ-45 connectors provide yet a third way to access the digital audio input and output. Three additional RJ-45 connectors are provided for control and Ethernet connections. The DB-9 connector provides 8 GPI inputs for the most used presets. The pinout drawings on pages 1-9 through 1-12 summarize all wiring connections.

Inputs

Both AP3 processor signal paths can be fed from three different types of inputs: microphone, analog line, and digital line.

When processing mic level (-50dBu nominal) inputs, the input is fed, either from one of the two female XLR connectors (one each for the two signal paths) or the appropriate pins on the DB-25 AUDIO I/O connector, to the internal microphone preamplifiers. The mic preamps have digitally controlled gain, up to a maximum of 70 dB, and display remarkably high performance and accuracy. Phantom power (+32VDC) is available.

The same connections can be used for line level analog audio inputs (nominal +4dBu balanced). When configured for analog line, the inputs exhibit a bridging impedance and can handle signals up to +20dBu.

A third female XLR connector, along with a paralleled RJ-45 connector, are available to handle digital line level sources; this AES digital input is also available on the DB-25 connector. The AP3 will accommodate digital inputs having sample rates of 44.1kHz or 48kHz. The sample rate of the AP3 will automatically sync up to the applied input signal's valid sample rate, allowing the AP3 to be used in an existing digital system. AES sources are by design stereo; if the AP3 is not set to be stereo the appropriate Left or Right signal within the AES stream will be applied to the signal path.

MIC / Line In—XLR-F

Pin 1 XLR1 SH – MIC1 / LINE 1 IN SH
Pin 2 XLR1 HI – MIC1 / LINE 1 IN HI
Pin 3 XLR1 LO – MIC1 / LINE 1 IN LO

Pin 1 XLR2 SH – MIC2 / LINE 2 IN SH
Pin 2 XLR2 HI – MIC2 / LINE 2 IN HI
Pin 3 XLR2 LO – MIC2 / LINE 2 IN LO

AES In—XLR-F & RJ-45

RJ-45

Pin 1 – N/C
Pin 2 – N/C
Pin 3 – AES IN HI
Pin 4 – N/C
Pin 5 – N/C
Pin 6 – AES IN LO
Pin 7 – N/C
Pin 8 – N/C

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 – MIC 1 / LINE1 IN SH
Pin 24 – MIC 1 / LINE1 IN HI
Pin 12 – MIC 1 / LINE1 IN LO
Pin 11 – MIC 2 / LINE2 IN SH
Pin 10 – MIC 2 / LINE2 IN HI
Pin 23 – MIC 2 / LINE2 IN LO
Pin 16 – AES IN SH
Pin 15 – AES IN HI
Pin 3 – AES IN LO

Outputs

The AP3'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 a male XLR connector, paralleled by an RJ-45 connector for CAT5 connection. 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 & RJ-45

RJ-45

Pin 1 – N/C
Pin 2 – N/C
Pin 3 – AES OUT HI
Pin 4 – N/C
Pin 5 – N/C
Pin 6 – AES OUT LO
Pin 7 – N/C
Pin 8 – N/C

XLR-M

Pin 1 XLR SH – AES OUT SH
Pin 2 XLR HI – AES OUT HI
Pin 3 XLR LO – AES 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 2 – AES OUT SH
Pin 1 – AES OUT HI
Pin 14 – AES OUT LO

Control Ports

There are two RJ-45 connectors (one for each signal path) available for the connections of a talent microphone control panel. Functions include remote on and off, cough, and tally.

Control—RJ-45

Pin 1 – DIGITAL GROUND
Pin 2 – COUGH
Pin 3 – N/C

Pin 4 – OFF TALLY
Pin 5 – ON TALLY
Pin 6 – REMOTE OFF

Pin 7 – REMOTE ON
Pin 8 – +5V DIGITAL

Optional ANP-RJ Announcer Panel

The optional ANP-RJ panel connects to the AP3 using a standard straight CAT5 Ethernet cable. This panel has "ON", "OFF", and "COUGH" switches for remote control of the microphone inputs. (Note that the "TB" button is not usable for the AP3). The "ON" and "OFF" buttons are LED illuminated. These panels can be mounted directly into a furniture surface or a turret.

To Turn the Microphone ON & OFF from a Remote Location

NOTE: The remote mic controls are meant to be used when the AP3 is processing two separate mono signals, typically microphones (see "Selecting a Signal Path" on pages 2-7 to 2-8). If the AP3 is in "stereo" mode there will be some non-intuitive interaction between the two sets of remote mic controls.

REMOTE ON — Activates the mic's channel ON switch. Provide a momentary closure between Remote On (Pin 7) and Digital Ground (Pin 1). This will latch the mic ON. (User-supplied momentary contact switch required.)

REMOTE OFF — Activates the mic's channel OFF switch. Provide a momentary closure between Remote Off (Pin 6) and Digital Ground (Pin 1). This will latch the mic OFF. (User-supplied momentary contact switch required.)

COUGH — Temporarily Mutes the microphone. Provide a closure between Cough (Pin 2) and Digital Ground (Pin 1). This will turn the mic OFF. Note this is a non-latching mode; the mic will turn ON again as soon as the closure stops. (User-supplied momentary contact switch required.)

On and Off Tallies

ON TALLY — Lets the mic's channel ON switch control an on-air light or other "microphone on" indicator at a remote location. This control function provides a continuous closure (open collector) between On Tally (Pin 5) and Digital Ground (Pin 1) whenever the module is ON.

OFF TALLY — Lets the mic's channel OFF switch control an on-air light or other "microphone off" indicator at a remote location. This control function provides a continuous closure (open collector) between Off Tally (Pin 4) and Digital Ground (Pin 1) whenever the module is ON.

The On and Off Tallies can be used to control externally powered tally lights that require a continuous closure to function, or external tally lights (i.e., LED) can be powered from the tally output by connecting the external LED to +5V Digital (Pin 8)* and the On Tally port (Pin 5) or Off Tally port (Pin 4). In any case, current should not exceed 30 milliamps.

*We recommend a series resistor between the LED and +5V digital when you are powering the external tally from the AP3; a value of 220Ω (1/4W 5%) is suggested.

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 AP3 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

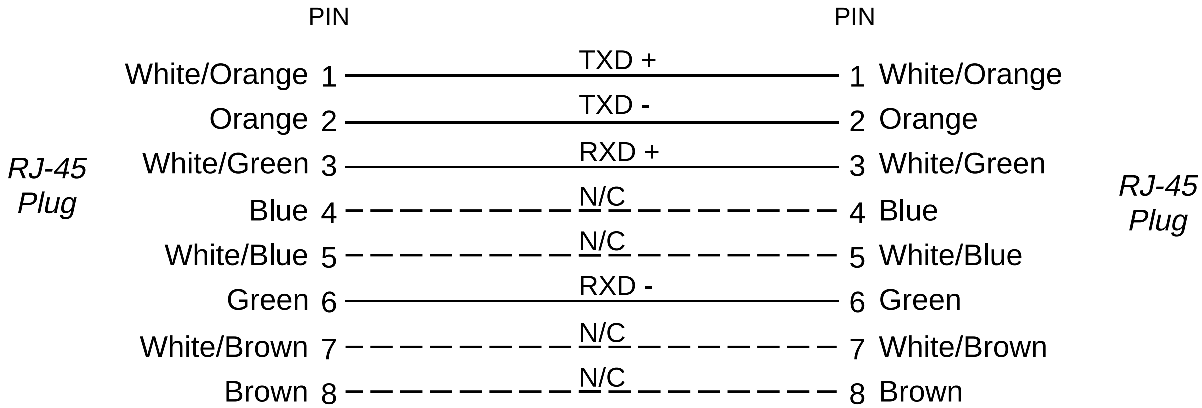
Ethernet Interface

Networked systems are connected to the AP3 via straight (pin to pin) CAT5 cable. For typical CAT5 cable pinouts see below. These connections are for communicating with the configuration computer, via a network. If you are connecting directly between the computer and the AP3 with no network in between, use a crossover cable.

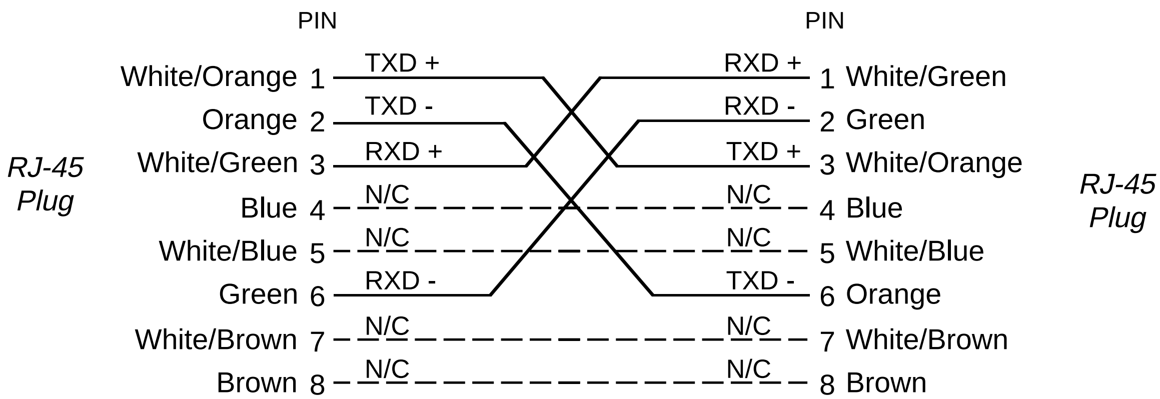
Ethernet—RJ-45

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

Typical Ethernet Cable



Typical Crossover Cable



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 110Ω. 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 or RJ-45 connectors. The DB-25 crimp style connector will accept wire gauge 22 - 28AWG. Please refer to the RJ-45 and 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 AP3 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™ AP3. Many third party “match boxes” are commercially available for this.

ANALOG OUTPUTS — The AP3 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)

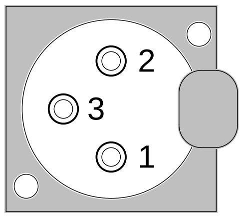
SPDIF INPUTS — The SPDIF (Sony/Phillips Digital Interface) or “consumer” digital audio interface is a two wire unbalanced signal typically on a single RCA style connector. Note that the SPDIF signal level of approximately 500mV and 75 ohm impedance does not correctly match the processor AES inputs. We 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™ AP3 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.

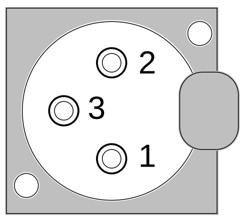
Mic/Line In - XLR-F

XLR1-F



PIN 1 XLR SH - MIC 1 / LINE 1 IN S
PIN 2 XLR HI - MIC 1 / LINE 1 IN HI
PIN 3 XLR LO - MIC 1 / LINE 1 IN L

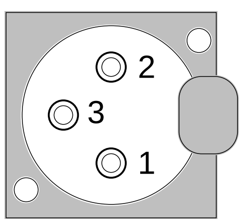
XLR2-F



PIN 1 XLR SH - MIC 2 / LINE 2 IN S
PIN 2 XLR HI - MIC 2 / LINE 2 IN HI
PIN 3 XLR LO - MIC 2 / LINE 2 IN L

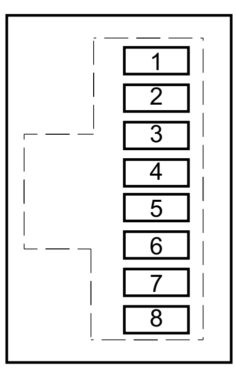
AES In - XLR-F & RJ-45

XLR-F



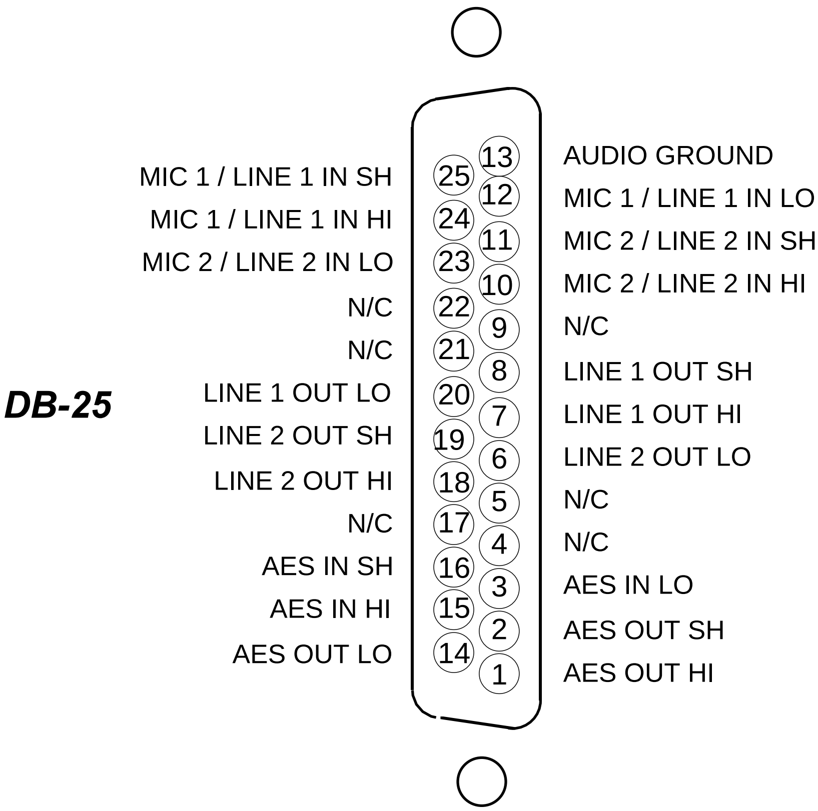
PIN 1 XLR SH - AES IN SH
PIN 2 XLR HI - AES IN HI
PIN 3 XLR LO - AES IN LO

RJ-45



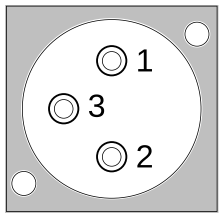
N/C
N/C
AES IN HI
N/C
N/C
AES IN LO
N/C
N/C

Audio Input / Output



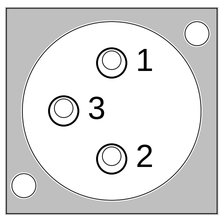
Analog Out - XLR-M

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

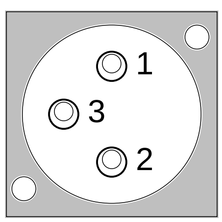
XLR2-M



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

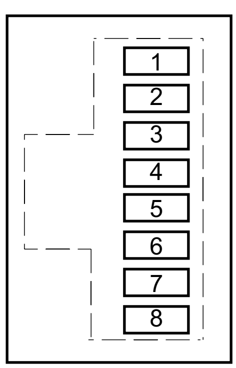
AES Out - XLR-M & RJ-45

XLR-M



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

RJ-45



N/C
N/C
AES OUT HI
N/C
N/C
AES OUT LO
N/C
N/C

Control - RJ-45

MIC 1 RJ-45 #1

1	DIGITAL GROUND
2	COUGH
3	N/C
4	OFF TALLY
5	ON TALLY
6	REMOTE OFF
7	REMOTE ON
8	+5V DIGITAL

MIC 2 RJ-45 #2

1	DIGITAL GROUND
2	COUGH
3	N/C
4	OFF TALLY
5	ON TALLY
6	REMOTE OFF
7	REMOTE ON
8	+5V DIGITAL

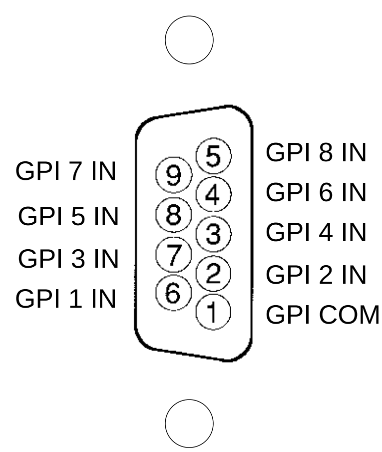
Ethernet - RJ-45

ETH RJ-45

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

GPI

GPI DB-9



AP3 Controls and Functions

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AP3 Controls and Functions

As internally complex as is the AP3, and as small as is the front-panel, it is obviously not possible to provide a control-per-function, as desirable as that may be. However, considerable effort has gone into making the front-panel controls logical and usable.

A wide variety of both frequency-response shaping and dynamics tools (on the right-hand side of the front panel) are available for selection and adjustment in each AP3 signal path via the switches on the right-hand side of the front panel. Despite the considerable overlap in functionality, these will be grouped into Signal Conditioning, Equalization, and Dynamics. This also approximately represents (exceptions will be noted) the order in which these tools are applied to the signal paths.

Most of the functions described in this chapter are covered in greater detail in Chapter 3.

Signal Conditioning

Generally these functions remove undesired signals and artifacts from the source material.

Phase

Phase reversal of the signal path, in case there is an inadvertent reversal elsewhere in the signal chain of which the AP3 is a part.

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.

High-Pass Filter

A fast rolloff-rate high-pass filter, adjustable between 20Hz and 1kHz, serves to aid removal of low-frequency signals and noises, such as air-conditioning rumble, table thumps, or in the case of a TELCO source, line-borne noises below the conducted frequency range (typically 300Hz).

Notch

The notch has been optimized in terms of width and depth to do the most damage to an undesired signal (such as supply-line hum, a TV/ computer line-timebase while, or an annoying instrument “ring”) while doing the least damage to the desired audio; it is at once broad enough to be reasonably easy to use in “finding” the offending signal, yet narrow enough not to impart excessive coloration to adjacent frequencies. It is tunable over the entire audio range of 20Hz – 20kHz.

Low-Pass Filter

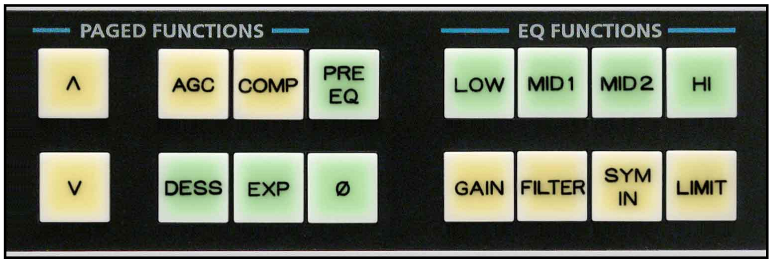
A fast rolloff-rate low-pass filter, adjustable between 20kHz and 1kHz, aids in the removal of distracting high-frequency noises, or to deliberately restrict the program material’s bandwidth.

De-Esser

The de-esser is a dynamic filter which reduces the signal energy within a chosen band of frequencies when it has been sensed to exceed a desired level. This is a valuable tool in reducing prominent sibilance (pronounced “SSS” noises) on some voices, 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.

Expander

Sometimes called “downwards expansion” the expander automatically reduces the signal level further once it has fallen below a chosen threshold. This is commonly used to “gently” or partially turn off a source if no valid signal is present. Examples are the suppression of room noises when no-one is talking into a microphone, reduction of prominent breath noises in speech, or the muting of phone-line noise if used on a TELCO circuit. Such uses become more necessary if the signal is later subjected to high degrees of compression, either within the AP3 and/or by subsequent airchain processing - compression tends to make room/breath/TELCO noises all the more objectionable by raising their level. Among more subtle usages can be to reduce an environment’s apparent reverberation time, and its effects, by accelerating the decay of the reverb “tail”; similarly with an instrument which has excessively long “ringing”. The AP3’s expanders have sufficient adjustable range to perform these subtleties, yet at the other extreme act as a precipitous gate.



Equalization

Parametric Equalizer

A powerful four band parametric equalizer, each band adjustable in center-frequency, bandwidth and degree of lift or cut is available in each signal path. It is selectable to be prior to or after the compressor (“pre” or “post”), each position having benefits depending on the application and program material.

Dynamics

The Dynamics processing within the AP3 is designed to maintain an optimum average level range (AGC, Automatic Gain Control), allow the increase of signal “density” or apparent “loudness” (Compressor) and prevent the output signal from exceeding a critical pre-determined level (Limiting).

Compressor

The compressor within the AP3 is a multi-band processor. It divides the signal spectrum up into three paths (nominally “LF”, low frequencies, “MF”, mid frequencies, and “HF”, high frequencies) each of which has its own individual compressor, the outputs of which are then recombined. The carefully matched and contrived dividing filters are adjustable in frequency to suit differing program types, voices or sources. Each of the three band’s compressors has individually adjustable parameters, making possible a wide range of signal control and dynamic equalization. Although the multi-band compressor can easily be cranked up to make the “wall-of-death” super-compressed effects for which multi-banders are prized as airchain processors, its intent in the AP3 is to provide a high-definition and well-packaged signal (free of all the cross-modulation and “pumping” characteristic of single-band compressors) which is better able to transcend later aggressive processing without being hacked into unrecognizability by it.

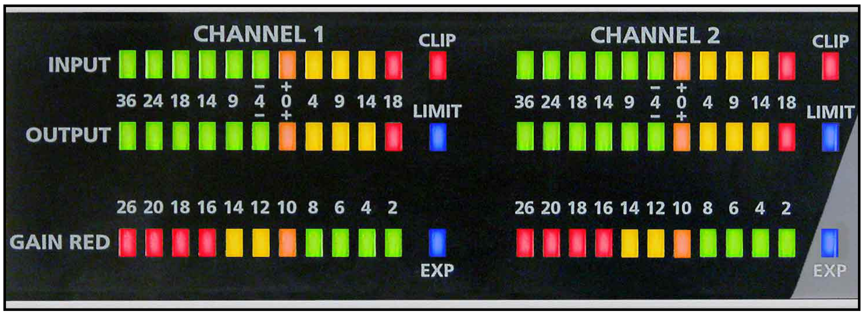
AGC

Multi-banded and operating as an integral part of each band of the multiband compressor, the AGC (Automatic Gain Control) observes the long-term signal energy and adjusts the signal levels accordingly to maintain them within an optimal range. Typically AGC would compensate for someone moving somewhat off-microphone, a reader steadily becoming tired, a DJ who can't read meters, or other such examples of relatively long-term applied signal level variation. A common set of attack and release time-constant values is applied across the three discrete AGC elements of the multi-band processor.

Limiter

Last in the signal processing chain is a precision peak limiter, which is intended to contain the AP3's output within desired level constraints, most particularly to avoid clipping within or overdriving signal destinations.

Metering



On the left-hand side of the front panel, each signal path has bargraph metering for (top) input signal level (adjustable by the input gain controls), output signal level (middle), and, as a default, composite gain-reduction (bottom). This latter indicates the gain-reduction being applied to the signal by any of the applicable processes: expansion, AGC, compression, or limiting.

Slightly distanced at the right-hand end of each of these bargraphs is a separate LED indicator, to show respectively for the input, output, and gain-reduction bargraphs input clipping (overdrive), output limiter activity, and expander activity. These last two will show if any indicated gain-reduction is due to these functions, regardless of control panel settings.

As mentioned, the gain-reduction bargraph reads by default a composite of all effecting signal-path functions: expansion, AGC, multi-band compressor, and output peak limiter. If however one of those functions is selected onto the Main Control Group, then the gain-reduction meter reacts solely to that function; in other words, if the output peak limiter ("Limit") is selected, the bargraph just indicates the gain-reduction due to the limiter. Similarly with the De-Esser, AGC, and Expander.

The multi-band compressor is a slightly different case in point; when selected ("Comp") the bargraph indicates a composite of the three bands' compressions. Overpressing one of the three encoders causes the bargraph to indicate just the gain-reduction for that band, for as long as the encoder is pressed. This permits rapid access to and visualization of the behavior of the multi-band compressor.

Main Control Group

The main method of effecting changes to any of the functions is a Main Control Group of three knobs, driving “endless” digital encoders, their associated 8-character alpha-numeric displays which show the values of the parameters under control at that point, and an “IN” (On/Off) button.

The signal processing functions upon which this Main Control Group is operating are determined by a set of buttons on the right-hand side of the front panel. Simplistically, pressing one of these buttons instantly makes the Main Control Group indicate and display parameters relevant to the button pressed; e.g. pressing LIMIT will bring the limiter’s parameters into the MCG for inspection or adjustment.

Many signal processing functions have more than three elements to adjust; this is indicated by the illumination of a pair of paging buttons (marked with up and down arrows) adjacent to the MCG which allow jumping between groups of adjustable elements. For example, the expander has Open, Hold, and Close times adjustable on one “page”; pressing one of the up/down buttons will instead display another “page” of expander elements, such as Threshold, Ratio, and Depth. The up/down buttons will cycle through available pages, although in different directions.

A variation on the above theme (of paging in different elements of a selected signal processing function) occurs when the multi-band compressor is selected. In this case, the like element of each of the three bands is displayed simultaneously; e.g. the three bands Thresholds will be up on the MCG at once. Paging with the up/down buttons will cycle through each of the variable parameters of the three compressors. This greatly facilitates the adjustment of the triband processor; one is likely to be operating on simultaneous adjustment of, say, output level trims across the bands, which having to jump from band-to-band would make highly tedious.

Selecting a Signal Path

The front panel Channel 1 and 2 buttons determine which of the two AP3 signal paths is being operated upon - the appropriate path’s button will be lit; pressing the other button will cause it to be lit instead, and command and display now act on that path. The AP3 can be made “stereo” (i.e. both paths being controlled together) by simultaneously holding down both Channel 1 and 2 buttons for two seconds; both buttons will light signifying both are under command, and henceforth both paths track together.

Both Channel 1 and Channel 2 will adopt the settings of the path which was last selected for control at the time of making the AP3 stereo.

An important exception to this adoption of a single set of parameters for stereo is Input Gain settings; if there is a difference between the Channel 1 and Channel 2 paths' input gains (such as could easily occur with two microphones, even as a stereo pair) this offset will be maintained between the two input gains as the now "stereo" input gain is adjusted. This offset will be ignored at gain extremes (i.e. both paths will ultimately reach max or min gain) but the offset will be retained and reapplied when the gain returns to saner values.

Additionally, the dynamics signal processing will become married, such that independent Channel 1 and Channel 2 gain variation and resulting stereo image shifting cannot occur.

Reverting to two independent paths again is achieved by holding down either the Channel 1 or 2 button for two seconds. Both path's settings adopt, until subsequently altered, the settings attained during the "stereo" usage.

Presets

A Preset is a snapshot of all the AP3's settings and configuration, with the intention of being able to recall the preset and restore the unit to exactly the same condition at a future juncture. The entire system (i.e. both signal paths, not one signal path at a time) is saved as a preset.

The AP3 is shipped with a number of likely useful presets included which are ready to use as-is, or as departure points for customization. All presets other than these will initially be "soft" and capable of being modified and over-written. 180 such presets are available. AP3 presets are "sticky" as opposed to "tunable". What this means is that if an AP3 preset is invoked and adjustments are made, whether by hardware front-panel or by the GUI, those changes do not become part of the preset unless deliberately saved back into it.

A DB-9 connector on the rear panel allows presets numbers 1 to 8 to be invoked, by application of a 5VDC signal between its associated DB-9 connector pin and the GPI COM pin. This is a simple and direct way of allowing AP3 configurations to be changed in accord with changing users, presenters or station day-parts.

It is envisaged that most configuration of presets will be performed using the AP3's corresponding GUI PC application, but it is nevertheless still entirely possible to create and manipulate presets from the hardware front-panel.

Immediately to the left of the Main Control Group of encoders and switches are three buttons for preset manipulation:

Presets Button

In the same manner as the function attention buttons on the right hand side of the panel, pressing this button calls preset management into the Main Control Group, and illuminates. The left-most encoder when rotated scrolls through the saved presets, the reference number of which is displayed in the above alpha-numeric display. Any text / label associated with the preset is displayed in the central display, and any attributes such as “locked/unlocked” (write-protection) will show in the right-most display.

Take Button

Upon being pressed and held for two seconds (it illuminates then darkens in this period), the preset showing in the displays will be immediately impressed into the AP3 which will become active with the contained settings and configuration.

Save Button

Pressing and holding this button for in excess of two seconds results in the existing and operating AP3 settings and configuration being recorded in the next available empty preset slot.

Modify (Take and Save simultaneously pressed)

Pressing both “Take” and “Save” buttons together for two seconds saves the current settings back into the preset showing in the displays.

These allow all operationally useful preset actions to be achieved easily from the front panel, including preset order rearrangement. There are however a number of preset management functions which are only done using the associated GUI:

• Preset order rearrangement while the unit is “live” and in use, and when changing the actual settings of the AP3 would be otherwise unwelcome.

• Labeling. We refuse to inflict the simply awful dial-up-a-letter-at-a-time Chinese-water-torture thing on anybody.

• Attributes, such as write-protection of created “gospel” presets.

• Transferring of presets from one AP3 to another.

System Features and Settings

"System" is a grab-bag of once-and-for-all set-up functions, or occasionally required features. This sub-menu is accessed by over-pressing (holding down for longer than two seconds) the "Presets" selection button. The "Presets" button slowly flashes while "System" is accessed.

The system element or feature is named in the left-most alpha-numeric display, while its present settings are displayed in the middle and right-most display. The elements may be cycled through using the left-most encoder.

Password

The password sub-system allows the user to enter a password in order to lock or unlock the front panel, and also to establish a new or change an existing password.

Through the "Presets" selection (which as noted above is still accessible however "locked" the box may be) the default "System" selection is for "Password".

(Should the AP3 already be "unlocked", i.e. operating without password protection, or having been opened using a password already, this first selection will instead be "New Pass"; see below.)

Password selection (actually a "pass-number"), consists of two numbers between 0 and 255 entered on the mid and right-most encoders in true safe-cracker fashion. A selection of "000" "000" represents no passwording. The middle encoder is turned until the indicated number is correct, then the knob is "dobbied" and selection is repeated with the right-most encoder. Once the required password has been dialed up, if it is accepted, the number disappears and "UNLOCKED" appears on the middle display; should - heaven forbid - the password be wrong, "WRONG PASSWORD" will display embarrassingly for three seconds before disappearing, prompting for another attempt, scoundrel.

Tip: DOBBY (pronounced dah-bee) - means to quickly press and release an encoder knob.

New Password

Unless the AP3 is fully unlocked (by using the password, if in use) this feature will be inaccessible.

"New Pass" will appear in the left-most display, and the two 0-255 numbers are dialed in using the center and right-most encoders, "dobbying" to set the number. "REPEAT" will appear in the mid display for three seconds before blanking; the new password number is entered again in the same fashion. If the two numbers agree, "PASSWORD ACTIVE" will appear in the mid display; otherwise "MISMATCH" will indicate irritatingly for three seconds prior to the display blanking as it makes ready for another attempt.

Access

This defines the amount of control an unattended AP3's control surface is allowed when locked with a password. Dialing through with the center encoder brings up the following options:

"All" — All functions and features of the AP3's control surface are available.

"Presets" "Gains" — The input gain controls remain active and preset selection is accessible.

"Presets" — Preset selection available only.

"None" — No access, other than to "Password".

"Inputs" "Follow" "Presets" / "Inputs" "Stay" "Same"

One or the other of these will be displayed across the three alphanumerics. They may be cycled between by rotating the center encoder.

A serious operational concern is that a freshly selected preset may or may not have appropriate settings for the input selection and gains contained within it, and similarly that once an AP3 has good input and gain settings, just differing processing is all that is required of a preset. Contrarily, sometimes everything within the preset, input selection, gain and all, is necessary.

"Inputs Follows Presets" (default) when selected for an AP3 allows the device to accept all preset information, while in "Inputs Stay Same" the AP3 ignores input selection and gain settings within presets.

It should be noted that, regardless of this setting, any presets saved in the AP3 do contain the existent input settings.

Sample Rate

When using the AES digital input, the AP3 conveniently automatically follows the incoming sample rate, providing it is a valid value.

For analog inputs, though, there are options. One of four displays will show in the "System" sub-menu:

- **"SampRate" "Analog" "48kHz"**

  The AP3's sample rate when using an analog input will be an internally generated 48kHz.

- **"SampRate" "Analog" "44.1kHz"**

  Internally generated 44.1kHz.

- **"SampRate" "Analog" "Auto/48"**

  The clock will be internally generated 48kHz, *unless* a valid AES stream is being presented to the AES input; in that case the AP3 will follow that clock.

AES CLOCK ERROR
If AUTO/48 or AUTO/44.1 is selected and no digital input is present, the displays will flash "AES CLOCK ERROR"

- **"SampRate" "Analog" "Auto/44"**

  Internal 44.1kHz, unless a valid external AES clock is applied.

Since the item being varied is the last display, the right-most encoder cycles through the options.

In the event that the AES digital input is being used but the data stream is sensed to be invalid, the AP3 will mute, and "AES" "Clock" "Error" will flash infuriatingly across the three displays until such time as either (a) a valid stream is restored or (b) any button on the control surface is pressed. From then on, the error message will flash for three seconds every fifteen seconds, unless activity is sensed on the front panel.

If an "Auto" option (above) is selected for analog input clock source, and the AES source is invalid, the same "AES" "Clock" "Error" will appear; however, the AP3 will not however mute, since it will be operating off its own internal clock instead.

Sample Rate Error Messages

"AES" "CLOCK" "ERROR" — displays if AUTO is selected and no AES clock is present, or if the signal is INVALID.

"SAMPLE" "RATE" "MISMATCH" — displays if the sample rate connected to the AES input doesn't match the selected sample rate (AUTO/48 or AUTO/44.1).

IP Address

An AP3 needs telling who it is, from a networking point of view, so that its associated GUI knows how to find it, and what to call it. The device's IP address is entered in a similar "safe-cracking" manner to the password (above):

An IP address consists of four numbers between 0 and 255 (Vorsis will supply the necessary intervening "dots" free of charge); they are shown across the center and rightmost displays. These four numbers are set up one at a time, in left-to-right sequence, using the center control encoder; rotating the center encoder will cause the first number to "dither"; once the desired number is reached, the center knob is dobbied, whereupon the second number will dither - this should be dialed-in and dobbied, too. Likewise for the third and fourth numbers.

Tip: DOBBY (pronounced dah-bee) - means to quickly press and release an encoder knob.

Entry by dobbying of the last (fourth) number makes the overall displayed number the IP address by which that AP3 is known to the world.

Subnet / Gateway

Entered in exactly the same manner as "IPAddres"; these are sometimes necessary TCP/IP routing codes. "Gateway" is presently unused.

Changes made to IP address, subnet, and gateway take effect after a subsequent power-cycle.

Vorsis AP3 GUI

Chapter Contents

Vorsis AP3 GUI

Getting Started

A PC application program to control one or many AP3s comes 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 AP3 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\AP3"; a "Vorsis AP3" icon will appear on the desktop, double-clicking on which will start the GUI program. 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 AP3 GUI could be problematic.

Connecting the AP3 and the GUI

The AP3 and PC running the AP3's GUI program can be straightforwardly connected together over a standard Ethernet Local Area Network (LAN).

The AP3 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 AP3 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 AP3 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.

The AP3 does not accept DHCP (automatically assigned network addressing) and needs to be told its own network-unique address; this can be entered into the AP3 from its front panel ("System" / "IPAddress" - see Chapter 2). Most DHCP networks assign addresses starting at the bottom of the group 192.168.000.xxx; choosing an address high in that group, say192.168.000.200 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 255.255.255.000 (Subnet) and 255.255.255.255 (Gateway) will suffice for all but complex situations.

For security, the AP3 will not act as a “web device” or support open connection to the Internet. However, connection between a GUI and AP3s 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.

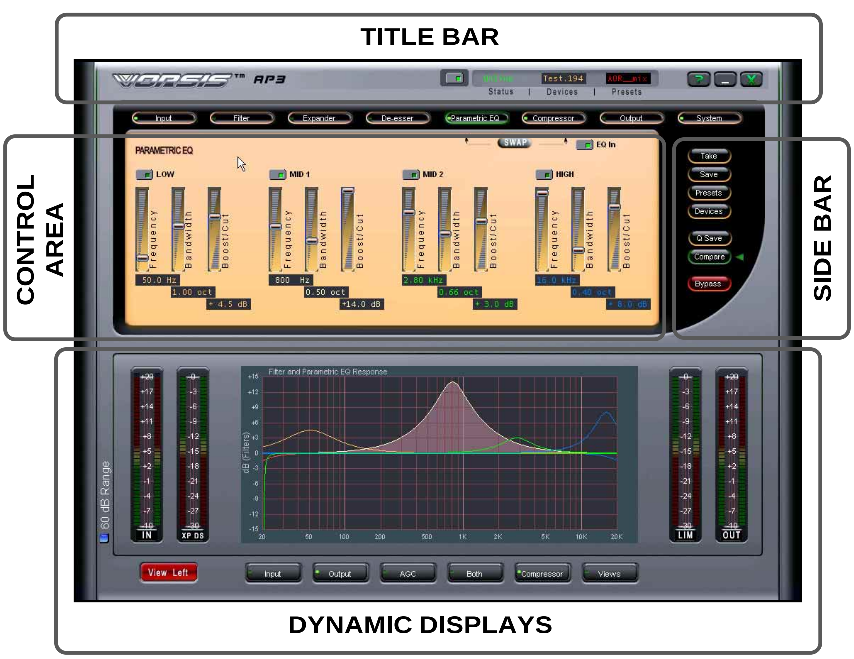
Once the AP3 has a valid address, the GUI can find it on the network. Double-click on “Devices” on the GUI; click “Add”; click “Browse” - most times the AP3’s IP address will promptly show in the appropriate field; enter a name for the AP3 if needed (say “Production”) and click “OK”. Alternatively, directly enter the new AP3’s IP address and its pet name, then “OK”. The GUI should immediately connect to the AP3, and “Online” will display. Once the GUI has been made aware of the AP3’s existence in this manner, it will always appear in the list of “AP3 Devices”, and be instantly accessible.

TIP: If “Trying” is continuosly displayed in the section of the title bar, then try:
• go Offline then Online, or
• cycle power on the AP3

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.



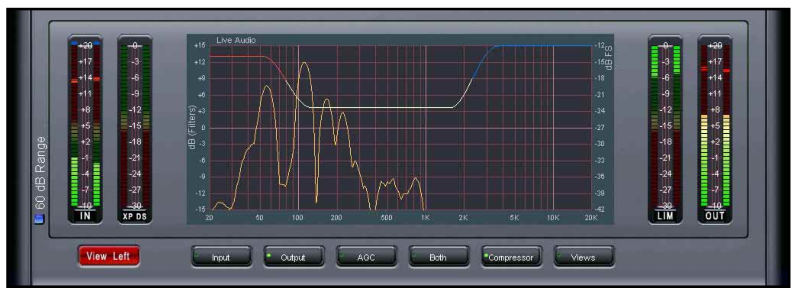
Dynamic Displays Region

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 is drawn.

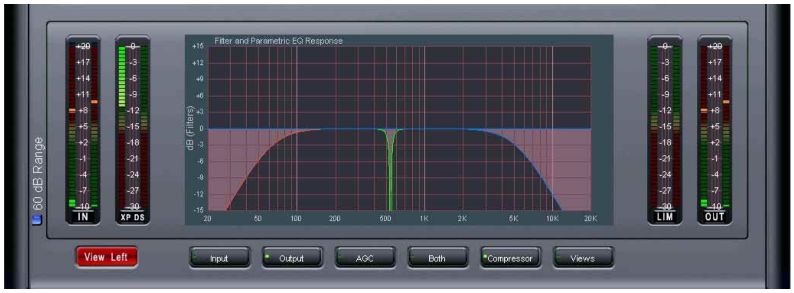
AGC — Displays the gain-reduction due to AGC action across the spectrum; since the AGC is three-band, a curve joining the LF, MF and HF gain reductions is drawn.

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 AP3 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 AP3 is being operated upon / displayed.

TIP: “A” and “B” correspond to the “Channel 1” and “Channel 2” labeling on the front of the AP3 unit.

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 reversed on the AP3 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, input gain control, balance/offset, symmetry decorrelator, and phase.

Symmetry

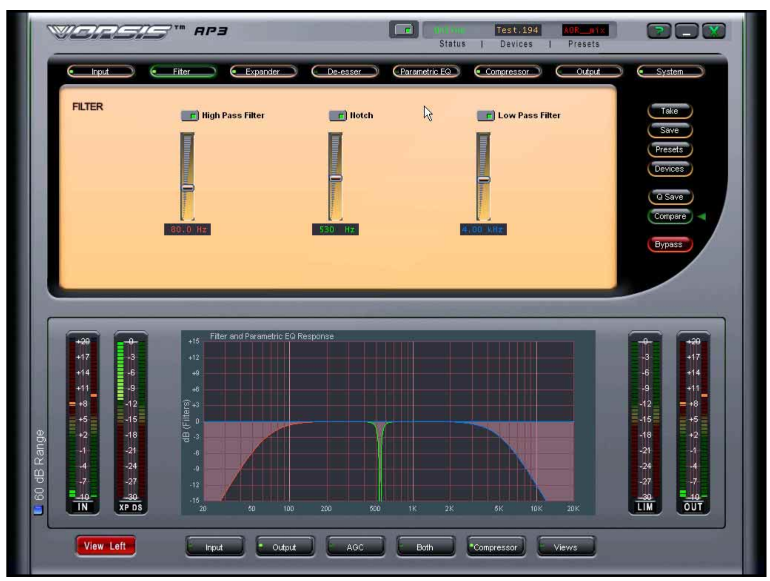
Used to reduce the "peak to average" ratio of the voice signal to allow a significant increase in apparent voice loudness in a dynamically controlled path (see Chapter 2).

Phase

Phase reversal of the signal path.

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 AP3 (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 off 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.

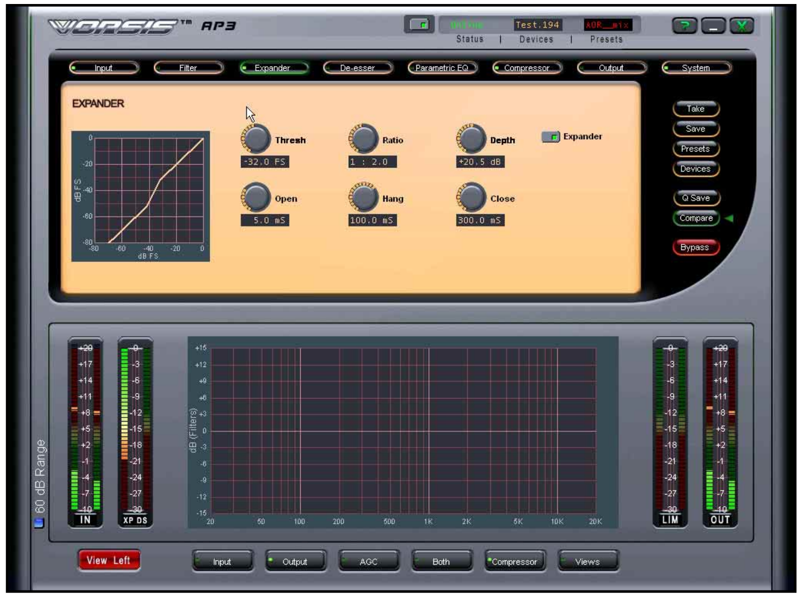
RATIO — Being the proportion of how many dB the input signal is attenuated for every dB it drops below the threshold. 1:3 indicates 18dB loss for 6dB drop in input signal level.

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 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.

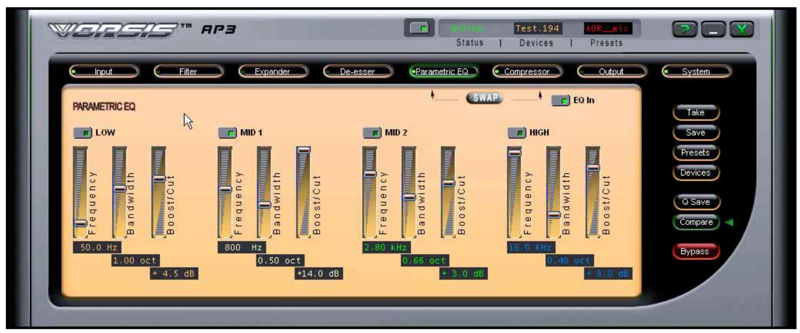
Regardless of the visual clues available thanks to the GUI, the most direct way of tuning in the de-esser is the old-fashioned way - by ear. Make the "Bandwidth" small, narrow; crank down the threshold (i.e. make it more sensitive), then carefully tune the "Frequency" until the undesired artifact is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

Parametric Equalizer

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



Compressor

The three-band dynamics section is the keystone of AP3. It 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 AP3 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 AP3'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 AP3 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 AP3'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:

TRESH — 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 AP3, 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 "Deutche 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.

Loud!

No pretences here about transparency or naturalness; the name of the game is Loud. Aggressive compressor settings, "Drive" high, high ratios, 5:1 and up, and short attack and release times. 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 making the compressor act 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.

The ear is most sensitive at mid-high 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.

The downside to these games is that highly Loud processing, although initially mightily impressive, becomes 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.

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 Atk” and “AGC Rls”, 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.

Limiter (available on the "Output" screen)

This is intended as a full-bandwidth "brickwall" limiter, a tool for hard-defining a maximum output level from the AP3, rather than an effects device. Protection, not production. Such limiters are necessary in applications where there is an inviolate maximum signal level in a following unit; transmitters, digital recorders, and HD / other bit-compression codecs are examples.

With an attack time set below 10mS, control is as a peak limiter; set below 2mS the peak control becomes absolute, with no overshoot, owing to the pre-sensing sidechain. Longer attack times are useful where a defined output level is desired, but there isn't a "hard" clip point; some PA systems, for example. Such allows a more relaxed feel to the sound.

Operationally, it is suggested that the limiter threshold ("Thresh") be set to a nominal level allowing for a reasonable yet not excessive amount of headroom (say, -10dBfs) and the limiter "Gain" adjusted until the desired maximum level for the downstream device is achieved. The compressor output level (i.e. the "Makeup" control on the compressor screen) should be adjusted such that the limiter just twitches and enters gain-reduction (as indicated on the GUI's "LIM" bargraph meters) on signal peaks.

Driving the full-bandwidth limiter hard invites back all the sonic artifacts one spent a lot of money on a multi-band compressor to avoid; however used as an incidental peak-control device as described above it allows maximum dynamic utilisation with no sonic detriment.

System

Sample-rate selection, stereo/mono selection, input settings protection from presets, security (passwords etc.), skin selection ("look" of the GUI).

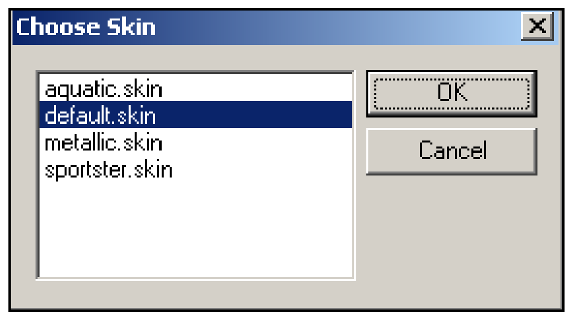
To a large extent, the selections here are of the "set-and-forget" variety, and mirror the System setup features and functions accessible through the AP3's front panel. There are five groupings to the System control screen:

SAMPLE RATE — Allowing selection of 44.1kHz or 48kHz internally generated sample rates, and the option of "Auto-Follow", under which the AP3 follows the sample rate of a valid applied AES/EBU signal.

STEREO — Through three buttons, this determines if the AP3 is going to operate as two discrete channels (“Dual Mono”), or if being made stereo, from which of A or B channels the common stereo parameters will be taken (“Stereo from A”, “Stereo from B”).

INPUTS — “Inputs follow presets”, if unchecked, allows the AP3 to ignore input source (Mic/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 AP3.

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 AP3, 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 AP3; 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.

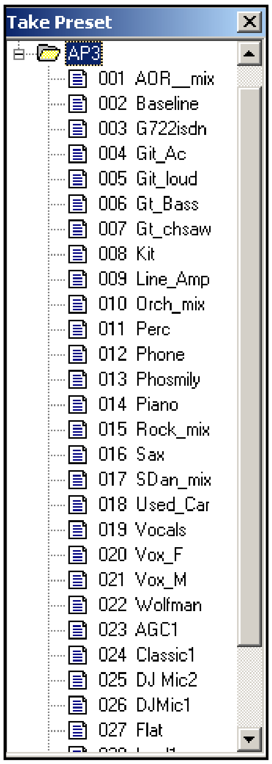


INTERFACE OPTIONS — Clicking “Skin” shows a selection of differing colors and styles available for the GUI. Selecting one and “OK” will cause the screen’s look to change in a few seconds.

Side Bar Region

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

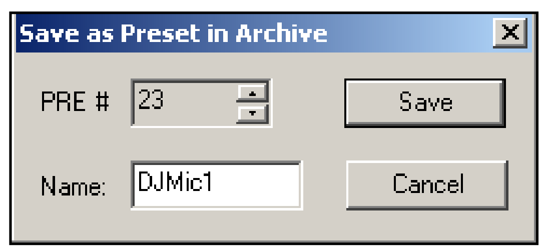
Take



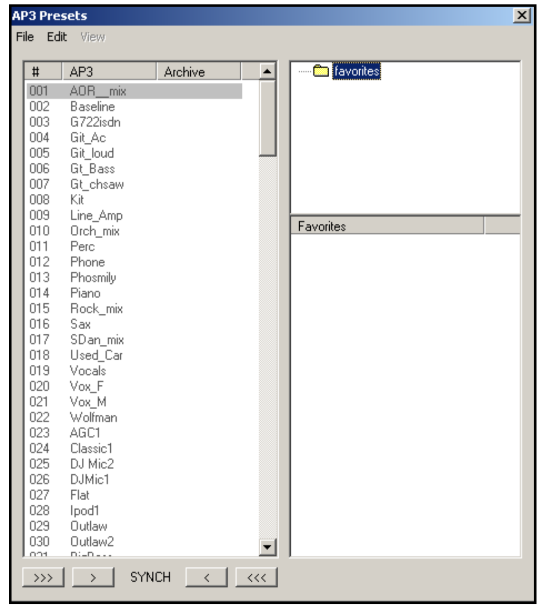
When left-clicked, a “Windows” box appears, showing folders that contain presets, typically “Favorites” and “AP3”, this last being presets already in place in the presently connected device. Opening a folder (if not already so) lists the presets in order. Double-clicking on a preset brings it immediately into use in the AP3. 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 AP3. 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.



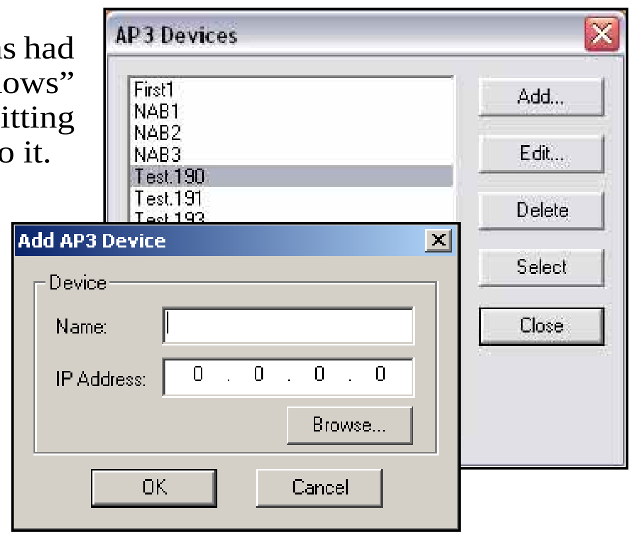
Presets



A three-panel “Windows” box appears. The large panel on the left has three columns, being the preset number, AP3 preset contents, and the GUI’s mirror archive contents for that AP3; 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 AP3 and its mirror archive in the GUI.

Devices



A list of AP3 devices with which the GUI has had cause to become familiar is shown in a “Windows” box. Highlighting the desired AP3’s name and hitting “Select” causes the GUI to attempt to connect to it.

In the event no AP3’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. “Browse” searches out likely victims connected to the same LAN as the GUI; this is the easy way if the AP3 has already been given an IP address.

Q Save (Quick Save)

This saves the immediate present AP3 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 AP3 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 AP3 GUI screen (in line with the "Vorsis AP3" 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 AP3 GUI is capable of controlling multiple AP3s ("Devices") and managing the "Presets" within them. Whether a connection is made ("Status"), and which device and which preset are presently under command are indicated. Double-clicking these indications will gain access to respective management screens.

Status

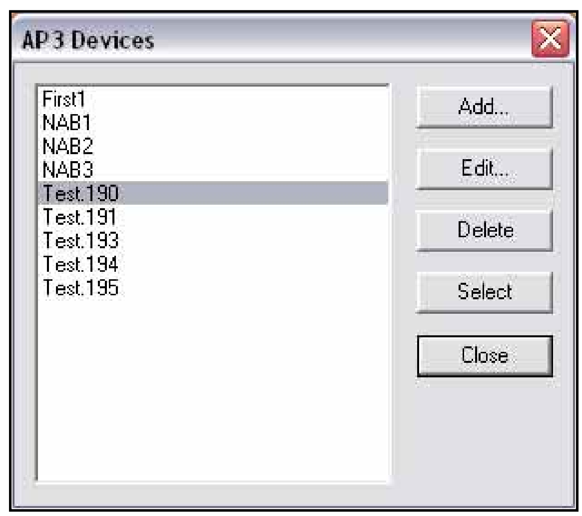
This indicates: "Online", meaning the GUI is in communication with and directly reflects an AP3; "Offline", meaning the GUI and AP3 are disconnected; "Trying", meaning the GUI is attempting to find the AP3 on the LAN and connect to it. An adjacent button toggles between "Online" and "Offline".

Devices

The name of the AP3 to which the GUI is connected (or with which it is attempting to connect) shows here in amber.

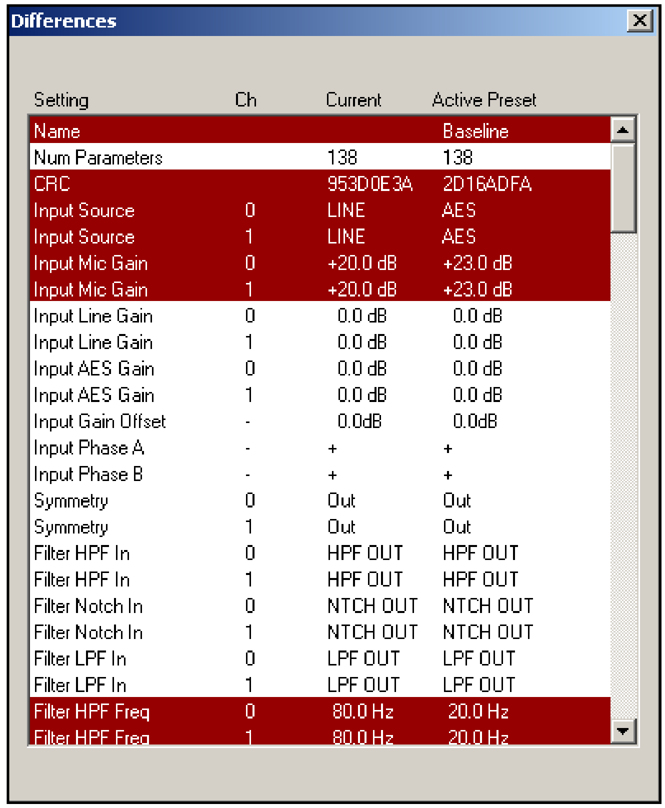
Double-clicking on the name area brings up the "Devices" "Windows" box. A list of AP3 devices with which the GUI has had cause to become familiar is shown in a "Windows" box. Highlighting the desired AP3's name and hitting "Select" causes the GUI to attempt to connect to it.

In the event no AP3'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. "Browse" searches out likely victims connected to the same LAN as the GUI.



Presets

The name of the last invoked preset is indicated here. Green indicates that the AP3'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 AP3 and the GUI mirror archive.


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

"SYNCH" allows the presets within the AP3 and the GUI to be made the same. The red color indicates a discrepancy between the actual current AP3 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 AP3, (b) the preset within the AP3 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:

a. Current settings correct:

- Save the current settings back into the originating preset.
- Copy the newly refreshed AP3 preset into the GUI archive.

b. Originating AP3 preset correct:

- Re-"Take" the preset into the AP3.
- 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 AP3 preset.
- "Take" the preset into the AP3.

Notes on "Online" and "Offline" Working

Most often operationally it will be required that the GUI act directly and instantaneously upon the AP3 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 AP3 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 AP3 is rendered "Online" again.

It is easy for the AP3 itself and the GUI to lose agreement; either the AP3 can have been adjusted using the hardware front-panel, or adjustments made on the GUI either "Offline" or when the AP3 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:

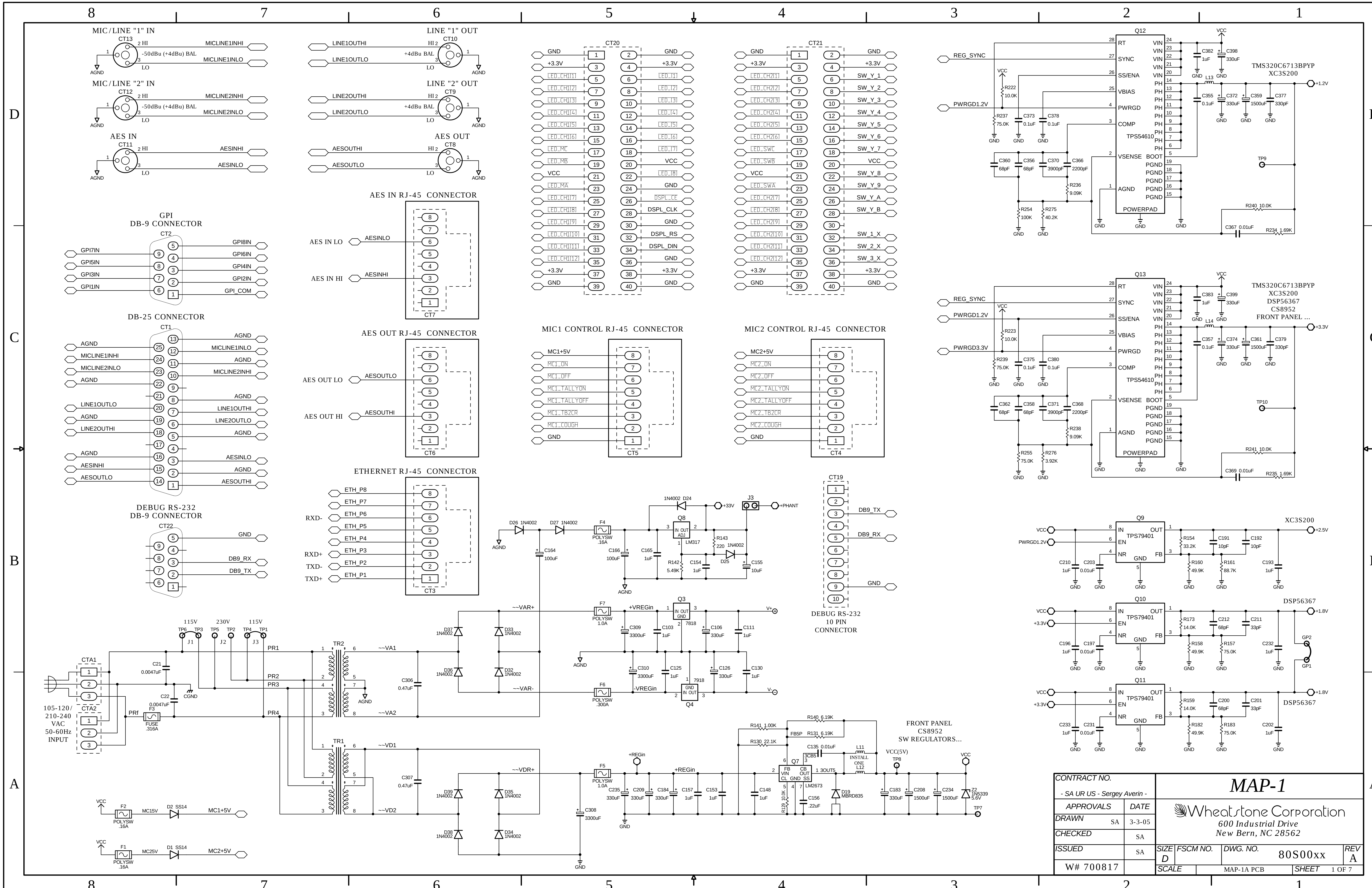
- "Online" - the double-clicked preset is immediately sent to the AP3 and becomes active.
- "Offline" - the double-clicked preset brings that preset onto the main control and display screen for perusal / editing. Any data entry / control which involves activity within the main graph will still operate upon the preset, but not on the AP3's audio.

I/O Schematic Drawings & Load Sheets

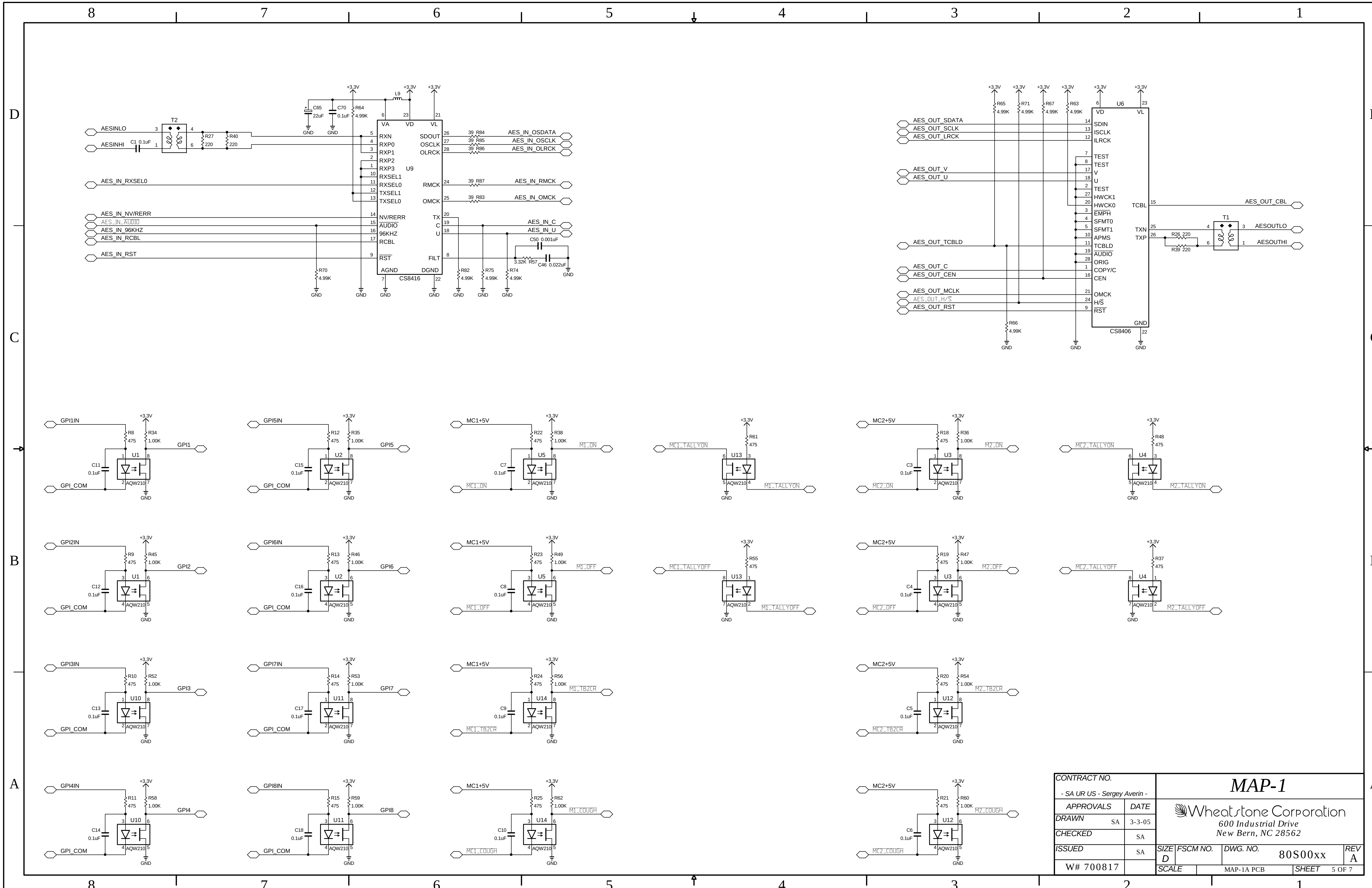
Chapter Contents

Mic Audio Processor (AP3)

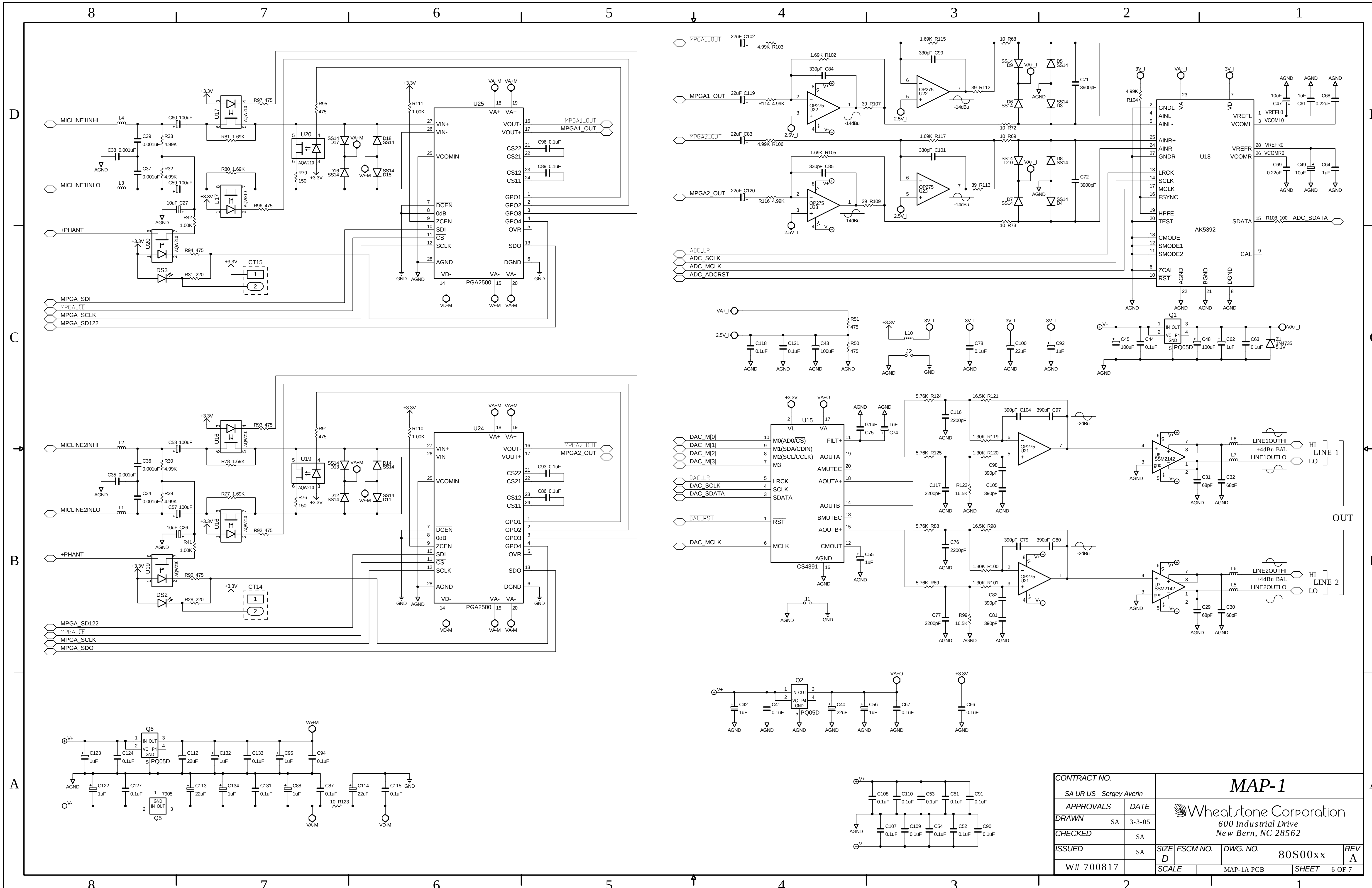
Processor Switch Card (PSW-1)



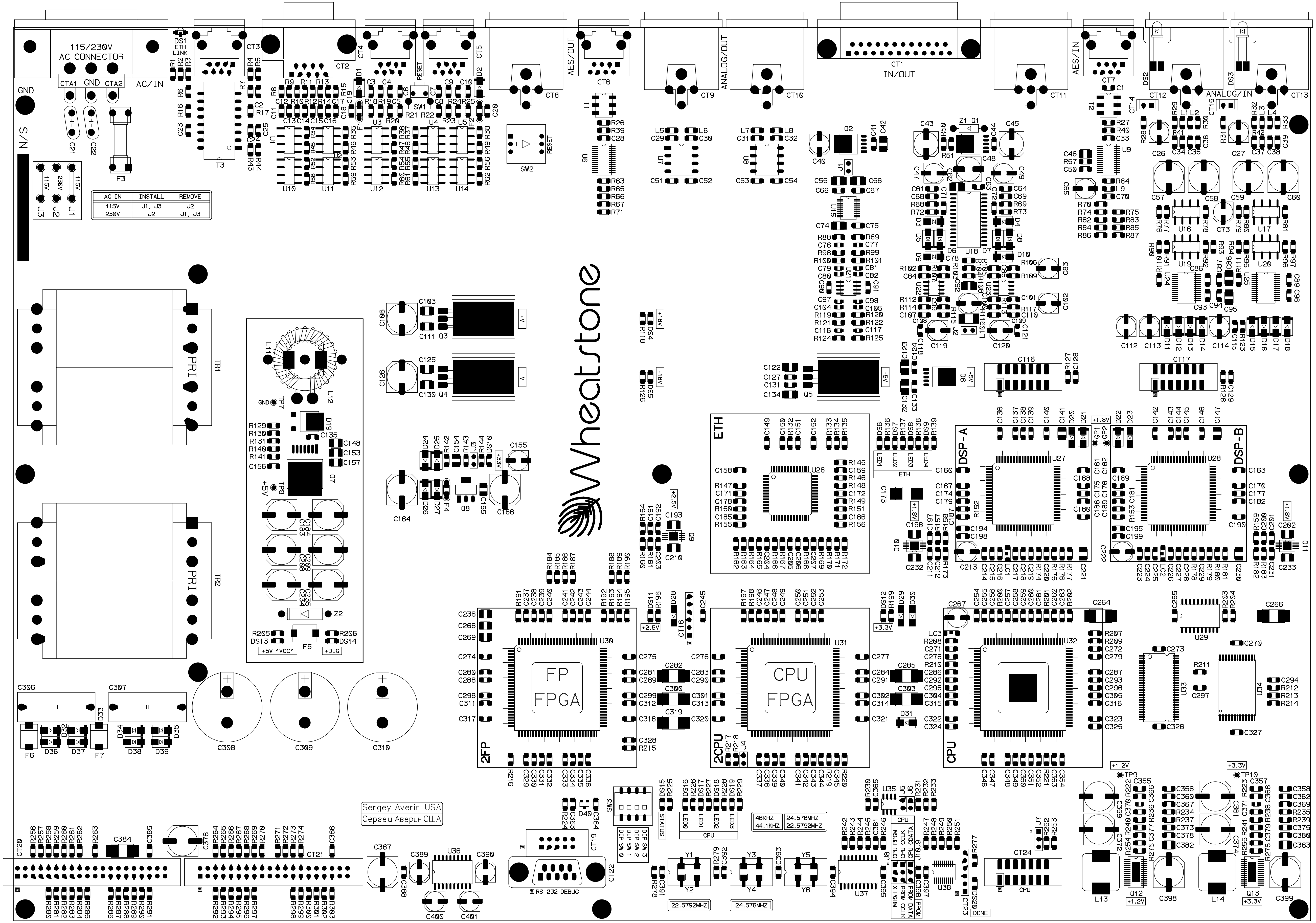
AP3 Mic Audio Processor Schematic



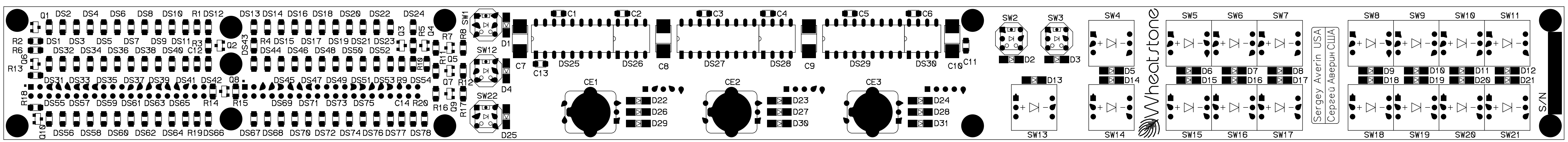
AP3 Mic Audio Processor Schematic



AP3 Mic Audio Processor Schematic



AP3 Mic Audio Processor - Load Sheet



PSW-1 Processor Switch Card - Load Sheet

Appendices

Appendix 1

Appendix 2

Appendix 3

Appendix 1

Contents

Using the AP3 or "Walk softly, and carry a Vorsis AP3."

Introduction

The Vorsis AP3 is a highly complex piece of audio processing gear. Typical industry hubris is such that most users will not get even close to reading this until panic ensues, or perhaps ever.

Nonetheless, it is strongly recommended that some low-stakes time is spent playing with the AP3 and its accompanying GUI software - which really makes operation of the unit easier - so becoming familiar with it. Such preparation will ensure that not only will there be good results under fire, but will definitely pay off in the long run by making for far better crafted presets and better sounds.

Lofty words dispensed with; here's the:

Five-Minute Processor

Power up the AP3. Connect input source and destination.

Stereo, or two channels of Mono? Press the "Channel 1", "Channel 2", or both together held for two seconds for Stereo.

Press "IN" button, or ensure it is unlit (and the AP3 is in bypass)

"IN" Switch
Lit = Processing
Dark = Bypass

Press "GAIN" button on front panel.

Press the "Page Up" (marked with up arrow) button until "Out Gain" appears in the left-most display.

Rotate the left-most encoder until "0.0 dB" shows on the left-most display.

Press the "Page Up" button until "Source" appears in the right-most display.

Rotate the right-most encoder until the required source is shown.

Adjust the left-most encoder until the input source material is peaking about +10dB on the input meter - between 0 and +20 is fine, but avoid clipping.

If necessary, readjust "Out Gain" to suit the destination.

The AP3 should now be passing audio and be roughly in accord with the environment's level structure.

Press the "PRESETS" button.

Cycle through the presets using the left-most encoder until one named appropriately to the program material shows; press the "TAKE" button until the light goes out.

Press the "IN" switch, if not already lit (and the box processing).

If necessary, go back to "In Gain" and "Out Gain" to adjust for levels.

Listen.

If unsure, peruse the presets some more and pick another to try.

The AP3 is now on-the-air! And probably well enough until it is possible to spend more than five minutes on optimizing the processing.

About Listening

There is lots of listening to be done setting up a box as complex and flexible as the AP3. It is critical, particularly when processing is about to be introduced into a chain which previously may have had none, or comparatively little. Going overboard in blazing glory, or underwhelming through timidity, are common errors with new processing, and best avoided by examining the new sounds as analytically and completely as possible in the contexts of the application and expected source program material.

It is ***firmly recommended*** that a pair of ***Good*** headphones be acquired, if not already owned. (“Good” means did-not-come-free-with-anything; a pair of “Good” headphones costs hundreds of dollars in the same way that a pair of “Good” loudspeakers costs thousands. “Good” means they don’t get lent out to anybody else. Ever. Especially not air staff.) A better investment cannot be made, and at minor cost in relation to the processing. It can be quite chastening how iffy most studio monitoring systems can sound in comparison to a pair of “Good” neutral headphones, and how much more detail one is capable of discerning. This is not to suggest all listening and setting up of the processor should be done with these headphones; in contrast, one should strive to listen to the results of presets under development on as many different monitoring systems and in as many different environments as possible, to ensure sonic transportability. Something may sound just grand in one place, but have - for example - the bass seem totally overblown in another. A reference pair of “Good” headphones will help avoid the easy excesses of a new processing system. Eventually, cross-experience will lead one to be able to reasonably anticipate a sound’s transportability simply from the headphones, in a manner impossible from more highly colored monitoring.

Processing settings are not once-and-for-all entities; as long as one may have spent fine-tuning a setup for a given application, deliberate or accidental revisiting is inevitable. Say, one day listening to station in the car, a sneaking need to tweak back the mid-band compressor a hair will surface. Or the realization that the expander on the main vocal may be set a tad high. Processor settings are in constant evolution to suit the subtly altering purposes and evolving sensibilities in their creator. Listen often and long, and be prepared to adjust settings to accommodate the results. To quote Cole Porter: Experiment!

Appendix 2

Contents

Parameters, Units and Ranges.

Approximately following the AP3's signal path, these are the values and ranges appropriate to each type of processing.

System Level

Headroom:	20dB
Nominal Operating Level:	-20dBFS

Input Gain

a. Microphone

Type:	Electronic Differential
Input Impedance:	> 2kohms at 1kHz
Optimum Source Impedance:	200 ohm
Phantom Power:	32V via 6.8kohm per leg
Maximum Gain:	70dB
Minimum Gain:	-10dB
Gain Increments:	1dB

b. Analogue Line Input

Type:	Electronic Differential
Input Impedance:	> 10kohm (bridging)
Optimum Source Impedance:	< 1kohm
Maximum Gain:	20dB (i.e. -20dBu input)
Minimum Gain:	-10dB (i.e. +10dBu input)

c. 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

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)

Peak Limiter

A zero-overshoot-capable peak limiter.

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

Appendix 3

Contents

For the most part there are no user-replaceable parts in the Vorsis AP3. 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 —AP3 PROCESSOR

COMPONENT	DESCRIPTION	*WS P/N*
MAP-1 LOADED CARD	PROCESSOR LOADED CARD ASSEMBLY	***"008413"***
PSW-1 LOADED CARD	SWITCH LOADED CARD ASSEMBLY	***"008413"***
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"***
NKK SWITCH	JB15 SWITCH W/BRIGHTER GREEN LED AND SILICON GASKET	***"510289"***
NKK SWITCH	JB15 SWITCH W/BRIGHTER YELLOW LED AND SILICON GASKET	***"510291"***
SWITCH	SINGLE POLE MOMENTARY SWITCH W/HOLES FOR LED, NO LED INSTALLED	***"510293"***
SWITCH CAP	WHITE SWITCH CAP	***"530004"***
LUMA BUTTON	GREEN LUMA-3 LIGHT PIPE BUTTON	***"530277"***
ENCODER	11MM ROTARY ENCODER WITH THREADED BUSHING	***"560002"***
ENCODER KNOB	11MM BLACK PUSH-ON KNOB	***"520105"***
ENCODER CAP	PLAIN BLUE CAP FOR 11MM COLLET KNOB	***"055200"***
DISPLAY	4 SEGMENT GREEN ALPHA NUMERIC DISPLAY	***"600016"***
MANUAL	OWNER'S MANUAL	***"008409"***