
FM - 2000

DIGITAL SPECTRAL PROCESSOR

TECHNICAL MANUAL



FM-2000 Digital Spectral Processor Technical Manual - 1st Edition

©2008 Vorsis

Vorsis
600 Industrial Drive
New Bern, North Carolina 28562
tel 252-638-7000 / fax 252-637-1285

Attention!

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

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



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

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

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





Making Audio Processing History

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

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

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

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

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

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

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

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

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

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

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

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

The Vorsis Vision

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

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

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

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

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

FM-2000 Technical Manual

Table of Contents

Chapter 1 – General Information

Introduction	1-2
Overview	1-4
Rack Mounting	1-5
Installation Tips	1-5
Where to Install the FM-2000	1-6
Analog STL	1-6
Analog Phone Lines	1-6
Digital STL	1-6
EAS—United States Emergency Alert System	1-7
Arbitron People Meter	1-7
Energizing	1-7
I/O Connections	1-8
Audio Inputs	1-8
Analog In	1-8
AES In	1-8
Audio Outputs	1-9
FM Path	1-9
Analog Out	1-9
AES Out	1-9
SCA In	1-9
TX Out	1-9
Headphones	1-10
Front Panel USB Port	1-10
Front Panel Ethernet Connection	1-10
Network Connections	1-10
Ethernet RJ-45	1-10
Typical Ethernet Cable	1-11
Typical Crossover Cable	1-11
General Purpose Interface (GPI/GPO)	1-12
GPI	1-12
User Presets	1-13
Example Schematic of General Purpose Inputs	1-14
GPO	1-15
Analog XLR Connections Pinouts Drawing	1-16
Digital XLR Connections Pinouts Drawing	1-17
BNC Connections Pinouts Drawing	1-18
Ethernet RJ-45/GPI DB-9 Pinouts Drawing	1-19

Chapter 2 - Features and Presets

Vorsis FM-2000 Quick Start Setup Guide	2-2
Ten Steps to Success	2-3
Achieving Your Special On-Air Sound	2-5
First a Few Words About Our Factory Presets	2-5
What Sound Are You Trying to Achieve?	2-5
About Bass	2-6
Isn't There Always a Caveat?	2-6
What Is Your Short Term Goal?	2-6
How Do I Make It Louder?	2-6
What Is Your Long Term Goal?	2-7
Clean and Loud	2-8
Can I Generate That "Sixties" Compression Sound?	2-9
The FM-2000 Factory Presets	2-10
A Quick Start (Factory Default)	2-10
Bass: <i>Loud, Loud n Wide, Punchy</i>	2-10
CHR: <i>Hot, Hotter, Medium</i>	2-11
Classic Rock: <i>I, II, III, IV</i>	2-11
Classical: <i>Big, Light, Medium</i>	2-11
Compressed: <i>Light, Loud, Medium, Wide</i>	2-12
Country: <i>Easy, Hot, Medium</i>	2-12
Dance: <i>Edgy, Energy, Smooth, Wide</i>	2-13
Folk: <i>Big, Medium, Soft</i>	2-13
Hot Stuff	2-14
Instrumental: <i>Big, Medium, Soft</i>	2-14
Jazz: <i>Big, Light, Smooth</i>	2-14
Latin: <i>Hot!, Medium, Mild</i>	2-15
News Talk: <i>Light, Loud, Medium</i>	2-15
Oldies: <i>Big, Medium, Soft</i>	2-15
Punchy	2-16
Radio Active	2-16
Sports: <i>Light, Loud, Medium</i>	2-16
Urban: <i>Heavy</i>	2-17
Future Vorsis Preset Updates	2-17

Chapter 3 - Vorsis FM-2000 GUI

Architecture of the FM-2000 GUI	3-5
Network Configuration	3-5
Getting Started	3-5
Connecting the FM-2000 and the GUI	3-5
Using a LAN Connection	3-5
Configuring Remote GUI	3-7
Connecting Directly Without a LAN	3-8
PC System Requirements	3-8
Wireless Connection	3-8
The Really Remote Connection!	3-8
VPN	3-9

The FM-2000 and Internet Security Concerns	3-9
About DHCP and FM-2000	3-9
FM-2000 Ports and Network Protocols	3-9
Using the GUI	3-10
Dynamic Displays Region	3-11
Frequency-Domain Graph	3-11
FM-2000 Graphical Interface Operation	3-12
First, a short tutorial	3-13
Bargraph Metering	3-14
Gain Reduction Meter	3-15
60dB Range	3-15
Display Signal Selection	3-15
Input	3-15
FM preDelay	3-15
Audio	3-16
Gain Reduction	3-16
Controls	3-16
L/R (Left/Right)	3-16
O-Scope	3-16
Control Area Region	3-17
Input Menu	3-18
Input Settings	3-18
Input Source	3-18
Input Signal Presence	3-18
Input Failsafe	3-18
Analog Gain	3-19
AES Gain	3-19
L/R Balance	3-19
Auto Mono Threshold	3-19
What is Auto Mono?	3-19
Phase Rotator	3-20
HPF - High Pass Filter	3-21
Why Would We Want To Do This?	3-21
VoiceMaster	3-22
Enable	3-23
Spatial Sense	3-23
Spectral Sense	3-23
Lim Drive	3-23
Pk (Peak) Ceiling	3-23
Lim Atk (Attack)	3-24
Lim RIs (Release)	3-24
AGC Atk (Attack)	3-24
AGC RIs (Release)	3-24
VoiceMaster Equalization	3-25
A Note Regarding VoiceMaster Equalization	3-26
Parametric Equalizer Menu	3-27
Enable	3-28
Automatic Gain Control and Compressor Menus	3-29
Drive	3-30
Makeup	3-30
SST/AGC/Comp Buttons	3-30
The Eight AGC Operating Modes: 0 through 7	3-31

SST Controls Overview	3-32
SST Drive	3-32
SST Gate Thresh.....	3-32
Ramp Rate	3-32
Window Size	3-33
AGC Controls Overview	3-34
Gated Modes	3-34
Ooze	3-34
Freeze	3-34
AGC Gate Threshold	3-34
Gate Delay	3-35
AGC Backoff	3-36
Band Coupling	3-36
Crossover Frequencies	3-37
Selecting Crossover Frequencies	3-37
AGC/Compressor Gain Reduction	3-39
FM Limiter Menu	3-40
31-Band Limiter	3-40
Multiband Enable	3-40
Multiband Drive	3-41
Lim/Clip (Limiter/Clipper) Drive	3-41
Multiband Knee	3-41
Vorsis Bass Management System (VBMS)	3-42
VBMS Drive	3-42
VBMS Style	3-43
VBMS Output	3-43
VBMS Frequency	3-43
Emphasis Selection	3-44
DeEmphasis	3-44
Look Ahead Limiter	3-44
FM Limiter Controls	3-45
Limiter Enable	3-45
L/R Linked	3-45
Limiter Thresh	3-45
Attack	3-45
Release	3-45
Delayed Release	3-46
Digital Output (dBFS)	3-46
Clip Style (Main Clipper)	3-46
FM Diversity Delay	3-47
Entering Delay Value	3-47
Special 31-Band Operating Settings — Wide and Timbral	3-48
31-Band Limiter Dynamics Screens	3-49
Thresholds Screen	3-49
Multiband Mode	3-50
Attack Time Screen	3-52
Release Time Screen	3-53
Coupling Controls Screen	3-54
Final Equalization Screen	3-55
VoiceMaster Equalization	3-56
A Note Regarding VoiceMaster Equalization	3-57

Stereo Encoder Menu	3-58
Stereo Width Limiter	3-58
Lim/Clip (Limiter/Clipper) Drive	3-59
Test Oscillator	3-59
Multiplex Process	3-60
Output Metering	3-60
Stereo Encoder	3-61
Pilot	3-61
SCA 1	3-61
SCA 2	3-61
Analog L/TX 1	3-61
Analog R/TX 2	3-61
Pilot Only	3-61
Pilot Phase	3-62
Multiplexer	3-62
Mask Filter	3-62
Analog—Stereo Coder Output Options	3-62
System Menu	3-63
System Settings Change With Preset Takes	3-63
Headphone Source	3-64
Headphone Control	3-65
Comp./AGC Bands	3-65
Remote Login	3-66
GUI Access	3-67
GUI Lock	3-67
Indicators/Temperature Readout	3-69
Skin	3-69
Events	3-70
Set Time	3-72
Backlight Dimmer	3-72
Side Bar Region	3-73
Preset	3-73
Save	3-73
Library	3-74
Locking Presets	3-75
Devices	3-76
Quick Save (QSave)	3-76
Title Bar Region	3-78
Status	3-78
Devices	3-78
Presets	3-79
Notes on “Online” and “Offline” Working	3-79
Accessing Menu Options	3-80
File Menu Items	3-80
Hardware Menu Items	3-80
FM-2000 Hardware Update	3-81
VBN/LBN Updating Order Sequence	3-85
Presets Menu Items	3-85

Chapter 4 - Schematic and Load Sheet Drawings

FM-2000 Signal Flow Diagram 4-2

Audio Processor (FM-2000)

Schematic 4-3

Load Sheet 4-8

Processor Switch Card (FMSW-2000)

Schematic 4-9

Load Sheet 4-10

Appendix

Parameters, Units and Ranges A-2

General Information

Chapter Contents

Introduction	1-2
Overview	1-4
Rack Mounting	1-5
Installation Tips	1-5
Where to Install the FM-2000	1-6
Analog STL	1-6
Analog Phone Lines	1-6
Digital STL	1-6
EAS—United States Emergency Alert System	1-7
Arbitron People Meter	1-7
Energizing	1-7
I/O Connections	1-8
Audio Inputs	1-8
Analog In	1-8
AES In	1-8
Audio Outputs	1-9
FM Path	1-9
Analog Out	1-9
AES Out	1-9
SCA In	1-9
TX Out	1-9
Headphones	1-10
Front Panel USB Port	1-10
Front Panel Ethernet Connection	1-10
Network Connections	1-10
Ethernet RJ-45	1-10
Typical Ethernet Cable	1-11
Typical Crossover Cable	1-11
General Purpose Interface (GPI/GPO)	1-12
GPI	1-12
User Presets	1-13
Example Schematic of General Purpose Inputs	1-14
GPO	1-15
Analog XLR Connections Pinouts Drawing	1-16
Digital XLR Connections Pinouts Drawing	1-17
BNC Connections Pinouts Drawing	1-18
Ethernet RJ-45/GPI DB-9 Pinouts Drawing	1-19



FM-2000 Digital Spectral Processor



General Information

Introduction

The Vorsis FM-2000 takes the power of the original AP-1000 and raises the bar to a whole new level! This second generation FM-2000 expands on technologies invented by the Vorsis design team, who approach audio processing with an attitude of “quality first.”

New and improved audio processing algorithms push the limits of what can be accomplished in an on-air processor. The result? Even more audio processing power while retaining complete user control over that power. No hidden controls. No secret “back door.” Every control is labeled for exactly what it does, and *everything* is brought out to the end user.

With the power and the tools required to create exactly the on-air sound you’ve been searching for, its audio clarity, on-the-dial power, and accuracy of sound is a combination not found in any other broadcast audio processor. And because the FM-2000 does not have its own sound coloration you are completely free to create any on air sound that you can dream of. Clean. Detailed. Loud. Or all three simultaneously.

The FM-2000 has 30% more DSP horsepower than the previous model, providing more than adequate resources for the new processing algorithms developed since the original AP-1000 was introduced. Nineteen DSP engines provide well over twice the computing power of the popular top-end audio processors, and an internal base sampling rate of 192 kHz processes your stations audio with extremely high precision.

A front panel LCD screen and touch-pad allows for total control of all system and processing parameters. The FM-2000 can also be remotely controlled from anywhere via a TCP/IP network connection and a Windows-based software application.

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



Wheatstone Corporate Headquarters

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

Wheatstone began in 1975 as the “Audioarts” company. By 1976, its product line had already grown to include high quality audio equalizers, electronic crossovers, and specialized compressor/limiters. In 1977 its first audio mixing control console was introduced. Wheatstone returned to its audio processing roots in 2005 when it created the Vorsis line of signal processing products and introduced the AP3 Digital Signal Processor for radio.

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

Overview



The FM-2000 digital audio processor is designed to process and output audio for conventional analog FM signals.

At the heart of the FM-2000 is a specialized DSP farm containing nineteen floating and fixed-point Digital Signal Processors. Advanced FPGA-based signal routing capability allows restructuring the processing chain as desired, and headphone monitoring can be patched to any processing section at will. Headphone monitoring is also possible for an audio source that is attached to the FM-2000 but not even on the air!

The most salient feature of the FM-2000 is its FM 31-band processing section. The FM side has the added feature of a 31-band distortion masked clipper that can be used alone or in combination with the 31-band limiter to achieve exactly the sound desired. This innovative design allows for far more flexible peak control and audio tailoring than the antiquated limiter algorithms having fewer bands that are found in other audio processors on the market today.

The 31-band limiter is equipped with fully adjustable thresholds and attack and release times for each of the 31 bands. Because the FM-2000 FM limiters operate much like the filter banks used in perceptual codecs its dynamic operation is nearly invisible to the ear. The sound of the signal path is quite flexible and can be carefully tailored to create exactly the on air sound desired.

The 31 bands used in the FM limiter allows tuning flexibility previously unheard of in any broadcast audio processor. The 31-band limiter section is followed by its own four-band parametric equalization stage that can be used to further fine tune the sound of the signal path.

The FM signal path is equipped with an adjustable diversity delay of up to ten seconds, corresponding to the requirements of the North American (iBiquity®) HD Radio system.

Rack Mounting

The FM-2000 is designed to fit into an industry standard 19" equipment rack, and requires three rack units (5.25 inches / 13.335cm) of vertical space.

The FM-2000 does not have top or bottom cover vent holes. Latent heat is vented out of the enclosure by natural convection through slots in the top of the rear panel. Cooler air is drawn into the unit through vertical slots positioned lower in the side panels.

There is one small fan internal to the unit to stir air inside the enclosure to prevent hot spots. It does not "remove" heat. Failure of the fan, in the unlikely event it should occur, will not compromise the operation of the FM-2000.

The FM-2000 may be mounted between other devices in the equipment rack; however, in accordance with good engineering practice it should not be mounted directly above devices that generate significant amounts of heat, such as power amplifiers or power supplies. If such a location is unavoidable, then it is advisable to utilize an extra 1RU blank rack panel between the FM-2000 and devices immediately above and/or below it.

WARNING! Under no circumstances should the FM-2000 unit be opened! The unit contains high voltage circuits that are hazardous and potentially harmful. The unit has no user-serviceable parts inside! If you have a problem the unit must be returned to Wheatstone Corporation for repair.

Installation Tips:

- Place any surge protection circuits as close as possible to the FM-2000 or other device being protected.
- Establish a low impedance common ground in your facility and try to route all grounds to that point.
- Choose the best power conditioning / UPS units that you can afford and suitable for your equipment - focus on the features and options you need. The better UPS products can prevent thousands of dollars in equipment damage — some even come with an external equipment damage warranty.
- *Unbalanced* audio connections to the FM-2000 should be made with shielded two conductor cable such as Belden 8451 or 9451 as if connecting a balanced source. At the unbalanced source's output connect the + Output to the HI input wire and connect the source GND wire to the LO wire. Connect the shield at the FM-2000 end only.
- For digital audio connections always use a good quality digital audio cable with a characteristic impedance of 110 ohms.

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



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

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

Where to Install the FM-2000

The recommended location for the FM-2000 is at the transmitter site if a discrete STL is involved in the signal path. A transmitter site installation will enable the use of the built-in stereo encoder, which will *always* allow more precise control of modulation peaks.

If you choose to locate the FM-2000 at the studio and use an STL to send the program material to the transmitter there are several issues that need to be addressed.

Analog STL:

Older analog discrete left/right STL’s can suffer from an inability to control high frequency audio peaks because of inadequate bandwidth in their IF circuits, or inadequate low frequency performance which will result in tilt. They also do not usually have identical audio group delay in both channels. On the other hand some composite STL designs may suffer from bounce in their AFC loops when handling processed low frequency material and this can rob modulation capability.

Analog Phone Lines:

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

Digital STL:

If using a Digital STL that employs audio compression, then it is highly recommended that the FM-2000 be placed at the transmitter site. This is primarily because the encoding schemes used in such STL’s will not accurately pass the well-defined peak levels created by the FM-2000.

The FM-2000 can also be placed at the studio end of a compressed STL with at least two caveats:

- You will not be able to use the composite clipper in the FM-2000. While most digital exciters offer a composite clipper function, they are typically quite crude in their operation by comparison to what is inside the FM-2000 and are *never* the optimum choice when sound *quality* is important.

- Compressed (data reduced) STL's may not perform well when competitively processed audio is presented to their inputs. This is primarily because when dense audio is presented to these codecs they have fewer "opportunities" to remove redundant audio information and mask that removal. Therefore, codec operation may be much more obvious — potentially more so than what might be expected if the processing is located *after* the codec where the masked artifacts are simply unmasked by the increased "gain" due to processing.

TIP: If using the FM-2000 at the studio be certain that the pre-emphasis and clippers in the stereo generator at the transmitter site are properly set up to complement the settings in the FM-2000. This will prevent gross distortion and large modulation overshoots.

EAS – United States Emergency Alert System

If the EAS generator is placed before the FM-2000, the levels of the signaling tones and FSK data may be disturbed by the processing. Therefore it is highly recommend that the EAS unit be placed *after* the processing.

If the EAS unit must be installed prior to it, the FM-2000 should be configured so that when the EAS tones are being sent the FM-2000 is switched to a bypass mode. This can be accomplished by using the FM-2000's GPI interface and by assigning a special preset to the EAS function that has internal processing defeated and audio levels calibrated to provide proper modulation levels during EAS transmissions.

Arbitron People Meter

The FM-2000 favorably passes the data watermarking scheme used in the Arbitron rating service People Meter technology regardless of the aggressiveness of the audio processing performed.

Energizing

Assuming the Vorsis FM-2000 Digital Audio Spectral Processor is correctly rackmounted, you may now energize it. There is no power switch. The AC line input voltage may be between 90 and 260 VAC, 50 or 60Hz. Power consumption is under 100VA.

Aggressive AC input filtering is utilized at the AC input of the FM-2000; however it is always advisable to use external surge protection and/or an uninterruptible power supply (UPS), especially where AC power quality is questionable, such as at a remote transmitter site.

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

Use of a UPS (uninterruptible power supply) is a good idea and will protect the FM-2000 from short duration power interruptions which may cause it to reboot. During boot up, audio is interrupted for approximately 28 seconds.

I/O Connections



All audio input and output, control, Ethernet, and power supply connections are made via various connectors mounted on the FM-2000's rear panel.

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

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

Two RJ-45 connectors are provided for Ethernet connections, with a third RJ-45 Ethernet connector provided on the front panel (see page 1-10). Any of these connectors can be used to connect a Windows® PC running the Vorsis FM-2000 GUI (see Chapter 3). The other two connectors can be used for peripheral equipment.

Two DB-9 connectors provide 8 GPI inputs (for selecting the first 8 presets) and 4 GPO outputs.

The pinout drawings on pages 1-16 through 1-19 summarize all wiring connections.

Audio Inputs

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

Automatic audio failover from one input to the other is a user selectable option. Failover from AES3 to analog is instantaneous and based on invalid or missing bits in the AES3 stream, or after 30 seconds of “silence” (level below -48dBFS). Failover from analog to AES3 is based on the same silence sense algorithm responding to audio below -48dBFS for more than 30 seconds.

Analog In—XLR-F

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

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

AES In—XLR-F

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

Audio Outputs

FM Path

Output audio is available as:

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

Analog Out—XLR-M

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

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

AES Out—XLR-M

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

SCA In—BNC

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

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

TX Out—BNC

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

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

Headphones

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

Front Panel USB Port

A USB port is provided on the front panel as a convenience to those who may not prefer the use of the front panel track pad for navigating the GUI, and instead wish to use a mouse of their own preference.

USB PORT NOTE!

The front panel USB port will accept a standard two button USB mouse (although three button or wheel types may work as well). However it is not configured for use with any other USB-equipped devices such as memory sticks, USB Flash Drives, CDROM drives, or wireless devices. Connecting such devices to the USB port may not cause harm to the FM-2000 or to the connected device, but such devices are unsupported at this time and may produce unpredictable results.



Front Panel Ethernet Connection

As previously mentioned, this is one of the three Ethernet connections that can be used equally as the GUI computer connection or for connections to peripheral equipment.

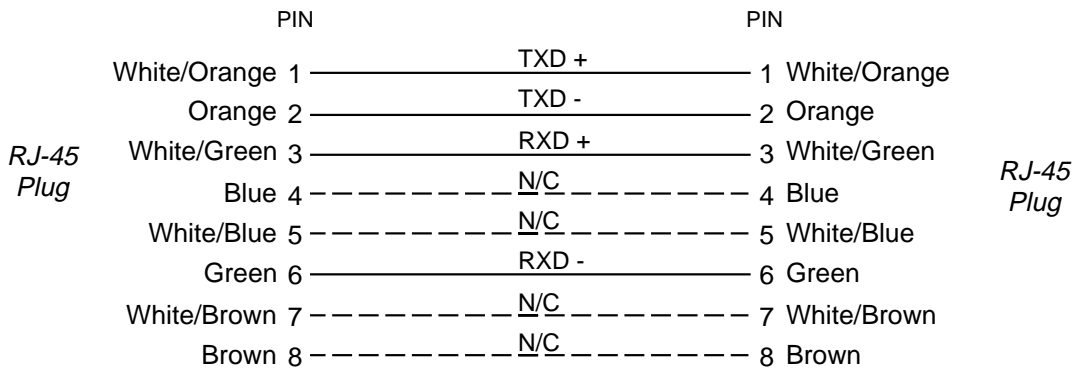
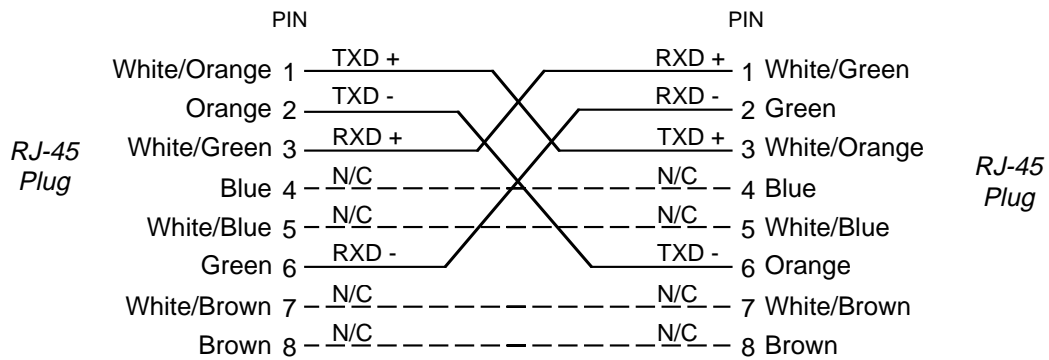
Network Connections

The FM-2000 contains its own five port 100Base-T Ethernet switch. The three ports available on the outside of the product (two on the rear panel and one on the front) are reserved for user connections. The Ethernet switch is equipped with auto-sensing on all ports and therefore any port will accept either a straight-through or crossover cable.

Networked systems are normally connected to the FM-2000 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 FM-2000 with no network in between, it is normal to use a crossover cable.

Ethernet—RJ-45

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

TYPICAL ETHERNET CABLE**TYPICAL Crossover CABLE**

General Purpose Interface (GPI/GPO)

The FM-2000 is equipped with eight General Purpose Input (GPI) control input ports on one female DB-9 connector, and four General Purpose Output (GPO) tally outputs on a second female DB-9 connector. All GPI and GPO connections are optically-isolated to prevent external ground loops and prevent dangerous voltages from being introduced into the FM-2000.

GPI

The GPI interface provides eight separate General Purpose Inputs, sharing a common return. The connector is a female DB-9 connector with the following pin assignments. Note that all pins are isolated from FM-2000 internal circuitry!

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

The GPI input circuits are optoisolated by devices having an input/output isolation of 1,500VAC. These inputs are expected to work very well in high RF environments and will not cause ground loop issues when connected to external equipment.

Each of the GPI inputs is current limited by a 475 ohm resistance located inside the FM-2000's GPI control circuitry. The LED device inside the optoisolators is rated at a maximum forward current of 50mA and a normal operating current of 5mA.

Before connecting external circuitry to the GPI inputs you should first ascertain whether it will be necessary to utilize additional out-board series resistance in your application!

The following table will serve as a convenient reference to determine if an external resistance is required in your application and if so, what value that resistance should be. Note that in all applications up to and including 48VDC a one-half watt resistor power rating is sufficient.

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

User Presets

The FM-2000 comes with a variety of factory presets and can hold a total of 80 in its onboard memory. You can design and save additional presets within the FM-2000's onboard memory and can store an unlimited number of presets on the PC that is hosting the remote control GUI software. The factory pre-sets can be readjusted and then saved to new names in order to create your own unique and individual sound.

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

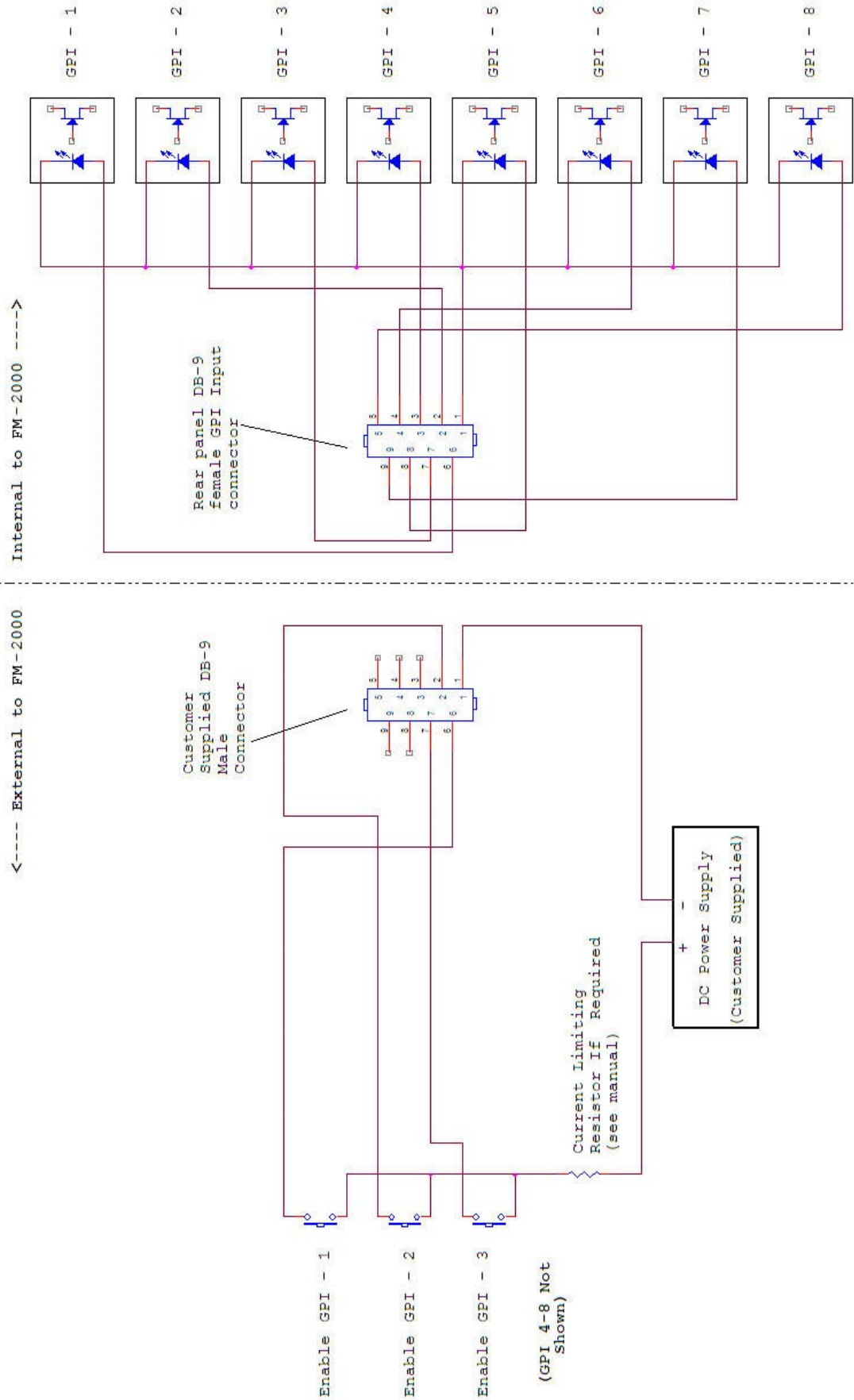
NOTE: User presets stored in locations 1 through 8 are logically assigned to the remote GPI function! Two examples of presets that might be stored here are:

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

Preset storage is typically as follows:

- Presets 1 - 8 are user presets that can be quickly selected from the GPI connector.
- Factory presets are installed starting at Preset 9, with Preset 9 being a "bypass" preset. The factory presets cannot be written over or deleted.
- The number of installed factory presets is subject to change, but is typically in the range of 40 to 50.
- User presets, other than the GPI selectable presets 1 - 8, are stored above the highest factory preset. The number of available slots for user presets depends on how many factory presets were installed.
- User presets may be locked by the user (see Chapter 3). Factory presets, on the other hand, are locked at the factory. Whereas a user can unlock user-locked presets, the same user can not unlock factory presets.

Example Schematic of General Purpose Inputs



GPO

The FM-2000 provides four General Purpose Output (GPO) circuits that may be used in a variety of ways. Like the GPI circuits, the GPO utilizes optoisolators in order to provide high electrical isolation from the outside world.

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

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

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

Pin 1 - N/C

Pin 2 – GPO 1 Return

Pin 6 – GPO 1 Source

Pin 3 – GPO 2 Return

Pin 7 – GPO 2 Source

Pin 4 – GPO 3 Return

Pin 8 – GPO 3 Source

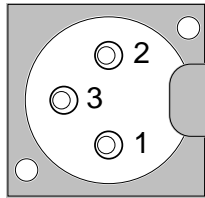
Pin 5 – GPO 4 Return

Pin 9 – GPO 4 Source

Analog XLR Connections

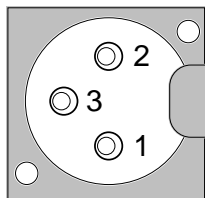
Analog In - XLR-F

XLR LT-F



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

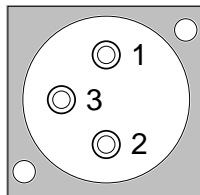
XLR RT-F



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

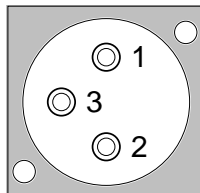
Analog Out - XLR-M

XLR FM LT-M



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

XLR FM RT-M

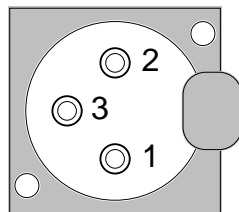


PIN 1 XLR FM RT SH - LINE FM RT OUT SH
 PIN 2 XLR FM RT HI - LINE FM RT OUT HI
 PIN 3 XLR FM RT LO - LINE FM RT OUT LO

Digital XLR Connections

AES In - XLR-F

XLR-F



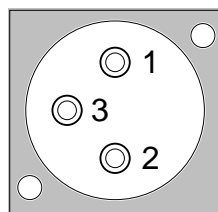
PIN 1 XLR SH - AES IN SH

PIN 2 XLR HI - AES IN HI

PIN 3 XLR LO - AES IN LO

AES Out - XLR-M

XLR FM-M



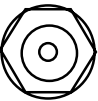
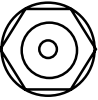
PIN 1 XLR FM SH - AES FM OUT SH

PIN 2 XLR FM HI - AES FM OUT HI



PIN 3 XLR FM LO - AES FM OUT LO

BNC Connections

BNC - SCA

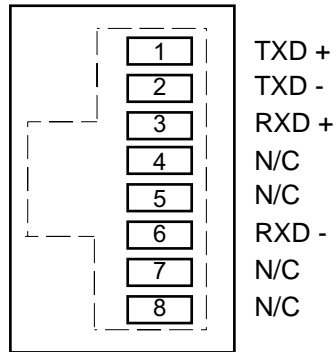
- 1  PIN 1 BNC 1 HI - SCA 1 IN HI - CENTER PIN
PIN 2 BNC 1 SH - SCA 1 IN SH - SHELL
- 2  PIN 1 BNC 2 HI - SCA 2 IN HI - CENTER PIN
PIN 2 BNC 2 SH - SCA 2 IN SH - SHELL

BNC - TX

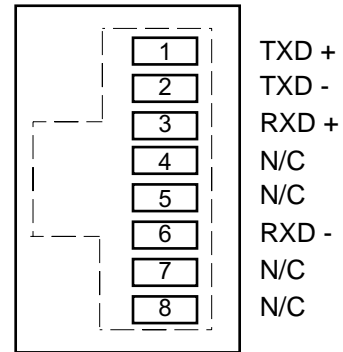
- 1  PIN 1 BNC 1 HI - TX 1 OUT HI - CENTER PIN
PIN 2 BNC 1 SH - TX 1 OUT SH - SHELL
- 2  PIN 1 BNC 2 HI - TX 2 OUT HI - CENTER PIN
PIN 2 BNC 2 SH - TX 2 OUT SH - SHELL

Ethernet - RJ-45

RJ-45 ETH1

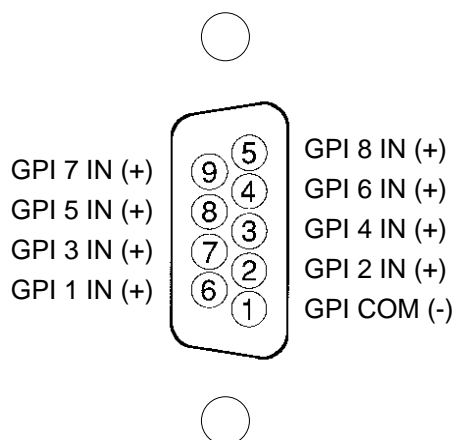


RJ-45 ETH2

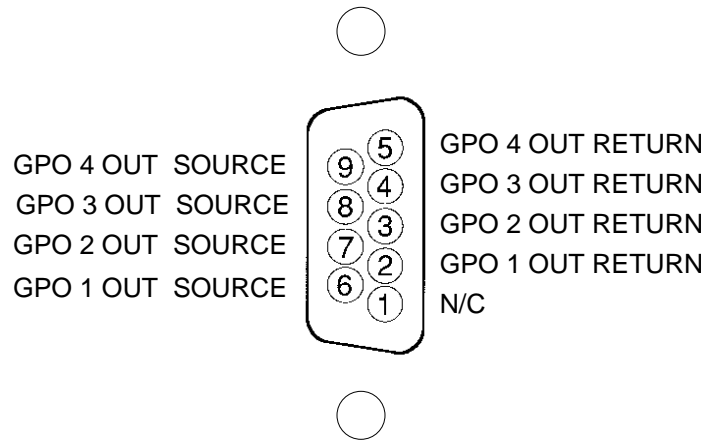


General Purpose Interface - DB-9

GPI - DB-9



GPO - DB-9



Features and Presets

Chapter Contents

Vorsis FM-2000 Quick Start Setup Guide	2-2
Ten Steps to Success	2-3
Achieving Your Special On-Air Sound	2-5
First a Few Words About Our Factory Presets	2-5
What Sound Are You Trying to Achieve?	2-5
About Bass	2-6
Isn't There Always a Caveat?	2-6
What Is Your Short Term Goal?	2-6
How Do I Make It Louder?	2-6
What Is Your Long Term Goal?	2-7
Clean and Loud	2-8
Can I Generate That "Sixties" Compression Sound?	2-9
The FM-2000 Factory Presets	2-10
A Quick Start (Factory Default)	2-10
Bass: <i>Loud, Loud Wide, Punchy</i>	2-10
CHR: <i>Hot, Hotter, Medium</i>	2-11
Classic Rock: <i>I, II, III, IV</i>	2-11
Classical: <i>Big, Light, Medium</i>	2-11
Compressed: <i>Light, Loud, Medium, Wide</i>	2-12
Country: <i>Easy, Hot, Medium</i>	2-12
Dance: <i>Edgy, Energy, Smooth, Wide</i>	2-13
Folk: <i>Big, Medium, Soft</i>	2-13
Hot Stuff	2-14
Instrumental: <i>Big, Medium, Soft</i>	2-14
Jazz: <i>Big, Light, Smooth</i>	2-14
Latin: <i>Hot!, Medium, Mild</i>	2-15
News Talk: <i>Light, Loud, Medium</i>	2-15
Oldies: <i>Big, Medium, Soft</i>	2-15
Punchy	2-16
Radio Active	2-16
Sports: <i>Light, Loud, Medium</i>	2-16
Urban: <i>Heavy</i>	2-17
Future Vorsis Preset Updates	2-17

Vorsis FM-2000 Quick Start Setup Guide

We know that you're probably in a hurry to get on the air with your new FM-2000. If you have little technical expertise with audio processor fundamentals, the following eleven point checklist will get you up and running quickly and without having to know a lot of techie stuff.

Please check out the various chapters of the FM-2000 Operating Guide if you need some additional information. If for some reason you find that you're really stuck, you can always call our friendly technical support folks at (252) 638-7000.

The photo below shows the rear panel of the FM-2000 while the drawing below it more clearly shows the location of the various connectors associated with an installation:



FM-2000 Rear Panel Connections

Ten Steps to Success

1. Install the FM-2000 in the equipment rack using at least two rack screws. If only two screws are used they *must* be installed in the bottom holes of the rack ears to prevent undue stress on the front panel and possible bending.
2. Connect local AC power to the unit. There is no power switch and power is applied to the unit as soon as you connect it to a suitable source of AC power. The power supply is universal and accepts line voltages from 90VAC to 260VAC, 50 or 60Hz.
3. Connect the Analog or Digital audio inputs as appropriate for your installation.
4. Click on the Input Menu tab on the front panel GUI and select the input source to match the Input type you connected to the rear panel. If your audio source is already providing an audio feed, you should now see meter activity on the FM-2000 front panel Input and Output meters.
5. While still in the Input menu and observing the Input meters on the left side of the front panel, adjust the Input Gain control (AES or Analog as required) until the input meters are peaking around -12dBFS with normal program levels being applied.



Note that for best overall performance, the Input of the FM-2000 should be calibrated so that it reaches 0dBFS when the signal chain prior to the FM-2000 also reaches 0dBFS, or digital full scale.

6. Make the audio output connections that are appropriate for your installation.
7. If using the Analog outputs pre-emphasized, navigate to the Stereo Encoder screen and select the checkbox for Analog L/R. Then adjust the TX1 and TX 2 controls to set the correct Left and Right output levels respectively.
8. If using the Analog outputs de-emphasized navigate to the Stereo Encoder screen and select the checkbox for Analog L/R deemph (de-emphasized). Then adjust the TX 1 and TX 2 controls to set the correct Left and Right output levels respectively.
9. If using the AES Outputs, connect those to the equipment following the FM-2000 and adjust the AES Output level appropriately.
10. If using the Composite outputs associated with the rear panel BNC connectors, navigate to the Stereo Encoder screen and select the checkbox for Analog MPX. Then adjust the TX 1 and TX 2 controls to set the correct modulation levels for your system.



When Analog MPX is selected, the Analog Left/Right outputs are also transmitting composite stereo, in balanced audio form while the BNC outputs are unbalanced. If you have an exciter that provides a balanced composite input (most do, even though the input is a BNC) the best possible signal to noise performance will be obtained by using the FM-2000 balanced outputs.

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

→ Cable shield to XLR Pin #3

→ Cable center conductor to XLR Pin #2

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

You're now on the air!

We ship every Vorsis Audio Processor with the Quick Start preset as the factory default (preset 10). This preset provides balanced competitiveness and provides a good starting point for a variety of program formats. We've adjusted it to have a rich and balanced sound and with reasonable loudness.

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



From time to time we'll be adding new presets to our Vorsis website for you to download and play with. Presets can easily be installed into your FM-2000 using the Preset management features within the remote GUI.

Achieving Your Special On-Air Sound

First a Few Words About Our Factory Presets

Factory presets are great starting points for many applications, but we'll be the first to admit that even with fifty presets in the product and another fifty or so that are available for download from the Vorsis website as optional Preset Packages, our presets simply might not be appropriate for every situation. Why?

Here's a simple math problem to illustrate our case:

$$A \times B \times C \times D = \text{NPPF (Number of Perfect Factory Presets)}$$

Where:

A = the number of different program formats that exist

B = the number of different sounding markets there are

C = the number of opinions about the station's sound

D = the number of adjustments that your competitors will make once you install your new Vorsis audio processor*

* also commonly known as "moving target syndrome"!

The product of $A \times B \times C \times D$ represents the number of presets that an audio processor manufacturer would need to provide in their products in order to perfectly match *every* possible programming + market + opinion + competitive combination. As you must imagine, this is certainly *not* a trivial number of presets!

So... what we do at Vorsis to help you, the customer, mitigate this issue is spend a lot of time, literally thousands of collective hours, very carefully crafting our factory presets so that they will get you closer to your desired on air sound quicker than if you had to build even one of the presets from scratch.

We try our very best to provide many good starting points for you in our factory presets. We're also always listening to our customers when they tell us what their competitive needs are. This is why you'll see ongoing advancements in our Vorsis product line and why you'll also periodically see new FM-2000 presets offered for download on our Vorsis website.

What Sound Are You Trying to Achieve?

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

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

The truth is, unless you're just now getting around to replacing an old audio processor that still uses 6386 variable μ p triodes, you won't be getting all four of the above differences in mass quantities just by buying and installing a new audio processor box!



6386 Variable μ p Triod

About Bass

The FM-2000 contains our specialized Vorsis Bass Management System (VBMS) that is tasked with extracting and revealing low frequency nuances in the audio programming that are simply not heard on the air with any other audio processor.

Isn't There Always a Caveat?

Of course! The majority of radio listeners don't have the quality of monitoring equipment that is found at most radio stations. Therefore great care is advised when adjusting the FM-2000 to accentuate the extremes of the audio spectrum.

Program energy at the frequency extremes is typically several dB below that in the more easily heard midrange, and because of the nuances of human hearing, very low and very high frequencies take a *lot* more modulation power to become audible to the average listener on the average receiver. Over-boosting of bass or treble energy that can be heard by only a small percentage of the station's audience can (will!) result in either reduced loudness or higher distortion for the majority of the station's listeners.

What Is Your Short Term Goal?

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

Almost as important as the station playing a listener's favorite record when they happen to tune in, a station that's *LOUD* carries with it a certain "authority" that can make a listener stay tuned long enough to see what's next. And though high loudness on FM doesn't technically increase coverage area of the station like it does for AM, greater loudness certainly does help mask noise at the fringes of the FM coverage area.

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

How Do I Make It Louder?

The FM-2000 is equipped to satisfy any competitive situation. In general terms, in order to just be louder (we'll cover that first — it's actually a bit easier) you should:

- Increase the density in the five band AGC by operating it with faster attack and release times
- Increase the Drive to the 31-band limiters
- Carefully use some equalization in the parametric section to gently boost frequencies that the ear is most sensitive to (1kHz to 4 kHz)

- Drive the clipper section harder to further increase the amount of RMS energy being generated by the processing
- Operate the 31-band section with faster attack and release times, in fact sloping them slower for the lows, faster for the highs (see Proportional Drag in the tutorial section for the GUI, page 3-14); this can create additional perceived loudness
- Intentionally over modulate like the other guys in the market are probably doing (we don't advocate this, but it *does* happen)

We haven't gone into the details of the 31-band clippers yet because they can be very deceptive. Because they operate instantaneously and in very narrow frequency bands they can generate some mighty impressive loudness without pumping or other dynamic artifacts and without any of the distortion typically generated by other, more "broadband" clipper schemes. The FM-2000 sets you free to be as loud as your market conditions demand.

Be aware that too much clipping depth carries with it the risk of driving away your long-term listeners, especially the females!

What Is Your Long Term Goal?

If the long term goal isn't to be the loudest radio station on the planet you can still be competitively loud while also sounding better, cleaner, brighter, and punchier than other stations in the market.

This is because when loudness isn't the first priority your options open up very quickly! Being "cleaner" than the other stations on the dial (while also having the right programming!) is thought to increase time spent listening. But be aware that being significantly cleaner also carries with it a perception of not being as loud. This is even though in acoustical and electrical RMS terms, the station may in fact be "loud." Why?

The easiest way to explain this is to remember how a boombox sounds when the volume is cranked up to 11 — it sounds loud, doesn't it? In electrical terms it probably isn't all that loud, but the distortion created by its amplifier being overdriven creates an association in the brain that says "it must be loud because it's distorted."

The opposite is true too — if your station has no perceptible distortion at all when compared to "the other guy" across town — then it might carry the perception that your station can't possibly be as loud. We're not saying don't be clean — cleaner is always better. Just don't be fooled by this psychoacoustic anomaly!

If you're not in an all-out loudness war you can use tools like equalization more freely than a station that has pulled out all the stops loudness-wise, and you'll have more freedom to craft the on air sound you really need or want. This is because equalization eats up headroom, and if you're trying to be really loud like they are, you need all the headroom you can get, and therefore less EQ can be used.

Clean and Loud

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

- Use slower attack times in the AGC sections
- Use slower release times in the AGC sections
- Use slower attack times in the 31-band limiters
- Leave the 31-band release times fairly fast (<100ms)
- Increase the drive to the final clipper (but not too much!)
- Increase the drive to the composite processor and use the Mask Filter
- Use a fairly low AGC Super Low to Low crossover frequency — perhaps around 80Hz
- Use a fairly high Mid High to High band crossover — perhaps 6 or 7kHz
- Be gentle with any mid-frequency equalization. In fact, partially subduing the range (no more than 1dB) between about 600Hz and 2.5kHz can subjectively clean up otherwise harsh-sounding material. To do this, use the parametric equalizer configured for post AGC/Compressor operation with a center frequency of around 1.2kHz, a bandwidth of about 1.5 octaves, and a cut of no more than about 1 to 2 dB.
- Another trick is use a small amount of EQ cut at 180-200 Hz. This will open up the sound on car radios which typically have a bit of muddy sound in this region. Use the parametric equalizer either pre or post AGC and set it for:

EQ Frequency of 180Hz to 200Hz

EQ cut of about -3dB

EQ Bandwidth of about 0.2 octaves.

The above combination will somewhat “relax” the sound because the AGC sections are operating slower, slightly lowering their average output levels and also because of the slightly reduced levels in the ear's most sensitive region due to the midrange equalization. If the “car radio” EQ is used the sound can be significantly opened up on such receivers while not losing too much in the way of loudness.

You'll note in the above recommendations that the drive has been increased to the clippers. The increased clipping depth, albeit only momentary because of the longer AGC attack times we recommended, will increase the subjective loudness on larger signal peaks. This will restore a sense of loudness and punch and create a feeling of dynamics and loudness and do it without generating the same kind of distortion and other artifacts that dense compression and higher duty cycle clipping would create.

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

Can I Generate That “Sixties” Compression Sound?

Of course! The FM-2000 can create virtually any desired on-air sound!

The way to do this is to purposely drive the multiband section harder and then trade off the slower acting AGC against the faster time constants of the compressors.

This is accomplished by operating the AGC Backoff control at negative numbers greater than about -3.0dB. In doing this the bulk of the gain control will be accomplished using the faster compressor time constants. If careful to not run the Backoff control too far negative, the AGC will provide a “platform” that the compressor can work against, preventing rapid suck-up of background noise that might be undesirable.

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

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

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

The FM-2000 Factory Presets

The following is a generic description of the Factory Presets shipped with the Vorsis FM-2000 audio processors to point out the general differences between them.

Each preset was named and tuned so that the dynamics control of the audio path through the five band AGC creates a desired foundation sound.

Each preset was arrived at through *many* hours of listening with appropriate program material and each is our best personal assessment of the tradeoffs between what the DSP algorithms are capable of and what a typical radio listener might find appealing.

Feel free to experiment with the presets, as each one has a unique sound. Also, don't fall into the trap of thinking that (for instance) the "Country" presets are only good for "Country" music. The preset that sounds best with your format is the one to use, either as is, or as a foundation sound to be personalized for your station.

Without further fuss about them, and in alphabetical order.....

A Quick Start (Factory Default)

This preset (preset number 10) achieves a balanced overall texture, fairly flat dynamic frequency response, and moderate loudness. Suitable for a wide range of program material, it is the factory default preset and can be used to get on the air quickly without embarrassment in most markets. It utilizes the multiband limiter operated in "Hard" mode. Both the main clipper and composite processors are utilized to generate reasonable loudness.

Bass: *Loud, Loud Wide, Punchy*

LOUD borrows many of the same settings from the Compressed Loud preset, but has a more relaxed sound due to less depth of dynamic frequency response corrections being allowed. The Super Low crossover has been set to 80Hz, allowing the SL band to bring up very low frequency material. The SL band's attack and release have been adjusted to pass the feel of the bass without overloading the sections following the AGC. A moderate amount of main clipping is performed in addition to moderate composite processing.

LOUD WIDE is basically the Dance Energy preset with the AGC section's Sum and Difference channels offset in drive and makeup gain to present a more widened stereo sound field. The sound field is wide, but should not create or exacerbate multipath issues. If you like the sound of WIDE but do experience increased multipath, you can try reducing the STEREO WIDTH limiter setting to 70% in the Stereo Encoder menu.

PUNCHY has further relaxed time constants than the Loud preset, primarily in the attack times of the five band AGC. The preset allows a moderate amount of multiband limiting with staggered time constants (slower on lows, faster on highs) and utilizes a moderate amount of drive in both main clipping and composite processing.

CHR: *Hot, Hotter, Medium*

HOT combines energetic qualities with more activity in the multiband limiter, deeper dynamic frequency response correction in the multiband AGC, and moderate amounts of multiband limiting. A small amount of final peak control is accomplished by both the main clipper and composite processor.

HOTTER combines even more energetic dynamics with deeper activity in the multiband limiter, deeper dynamic frequency response correction in the multiband AGC, and moderate amounts of multiband limiting. A moderate amount of the final peak control is accomplished by utilizing both the main clipper and composite processor.

MEDIUM combines some less energetic qualities of HOT with a little less activity in the multiband limiter. There is also less deep dynamic frequency response correction in the multiband AGC, and a bit less multiband limiting. Final peak control is accomplished by sharing this task with both the main clipper and composite processor.

Classic Rock: *I, II, III, IV*

I creates a balanced texture with moderate loudness. It offers a moderate amount of short term dynamics control as well as moderate levels of dynamic frequency response correction. Preliminary peak control is accomplished by small amounts of multiband limiting and clipping, with primary peak control being done by the main clipper. A small amount of composite processing is used in this preset.

II is an energetic but smooth preset utilizing approximately the same levels of short term dynamics control and dynamic frequency response correction as I, but relies more heavily on limiting in the multiband section. Primary peak control is accomplished by the main clipper. A small amount of composite processing is also used in this preset.

III combines the more energetic qualities of II, with more activity in the multiband limiter, deeper dynamic frequency response correction in the multiband AGC, and moderate amounts of multiband limiting. Final peak control is accomplished by both the main clipper and composite processor.

IV is a variation of III above with a different set of multiband AGC parameters to give it a different “feel” from the other Classic Rock presets. Final peak limiting is performed by both the final clipper and composite clipper.

Classical: *Big, Light, Medium*

BIG is an even more energetic version of MEDIUM, allowing larger corrections in short term dynamics to be made as well as a small amount (~3dB) of correction of dynamic frequency response. Final limiting is by small amounts of multiband control, clippers defeated, “Timbral” enabled, and with final peak limiting done by a combination of a small amount of main clipping and a small amount of composite processing. BIG has a larger on-air presence loudness-wise, but still retains the subjective cleanliness of MEDIUM.

LIGHT is a soft preset and maintains a very flat dynamic frequency response. It alters long term dynamics only, leaving short term dynamics untouched. It utilizes look-ahead limiting in lieu of clipping as the final peak control method and therefore will be very clean but not competitively loud.

MEDIUM is a more energetic version of LIGHT, allowing small corrections to short term dynamics. Like LIGHT, it has very flat dynamic frequency response and uses the look-ahead final limiter. The primary difference is that this preset also utilizes a very small amount of composite processing to slightly boost perceived loudness without generating noticeable forms of distortion.

Compressed: *Light, Loud, Medium, Wide*

LIGHT is tuned to be compressed but not aggressively so. It also uses the Expander section to gently subdue background noise when the audio level falls. A moderate amount of dynamic frequency response correction is used, as well as moderate amounts of drive to the multiband limiter section in order to increase density. Clipping is kept to a minimum. Additional “compressed feel” is accomplished by using the look-ahead limiter. Final peak control is by the main clipper only. No composite processing is used.

LOUD is more compressed than Medium. A large amount of dynamic frequency response correction is used, as well as moderate amounts of drive to the multiband limiter section, which increases density. Clipping drive is deeper. Like the Medium preset (below), a more “compressed feel” is accomplished by using the look-ahead limiter. Final peak control is by a main clipper with moderate composite processing added for increased modulation density.

MEDIUM is tuned to be more compressed but still not aggressively so. A moderate amount of dynamic frequency response correction is used, as well as moderate amounts of drive to the multiband limiter section to increase density. Clipping is increased over the Light version. Additional “compressed feel” is accomplished by using the look-ahead limiter. Final peak control is by a main clipper with a small amount of composite processing added.

WIDE is tuned almost identically to the Medium preset, but uses offsets in the Sum and Difference channels to achieve a wider stereo sound field. If you like the sound of WIDE but experience increased multipath on your station, you can try reducing the STEREO WIDTH limiter setting to 70% in the Stereo Encoder menu.

Country: *Easy, Hot, Medium*

EASY creates a balanced texture with moderate loudness. It offers a moderate amount of short term dynamics control as well as moderate levels of dynamic frequency response correction. Preliminary peak control is accomplished by small amounts of multiband limiting and clipping, with primary peak control being done by the main clipper. No composite processing is used in this preset.

HOT combines the energetic qualities of Medium (below) with more activity in the multiband limiter, deeper dynamic frequency response correction in the multiband AGC, and moderate amounts of multiband limiting and clipping, with a small amount of final peak control accomplished by both the main clipper and composite processor.

MEDIUM is an energetic but smooth preset utilizing approximately the same levels of short term dynamics control and dynamic frequency response correction as Easy, but relies more heavily on limiting in the multiband section. Clipping is approximately the same density as Easy, with primary peak control accomplished by the main clipper. A small amount of composite processing is also included in this preset.

Dance: *Edgy, Energy, Smooth, Wide*

EDGY uses the same crossover frequencies as Energy (below), but with additional dynamic frequency response correction and deeper compression. With some material, particularly that which has been compressed by a perceptual codec, the program can sound a little “edgy.” We didn’t find the sound particularly annoying or negative, just different. So we kept this preset and just named it “Edgy.”

ENERGY is compressed like Smooth (below), but has a greater degree of dynamic frequency response correction and lower Super Low and higher High band crossover points. This results in a tight feel on the low end with sheen on the high end.

SMOOTH is an obviously compressed, level-feeling preset with good dynamics control and average dynamic frequency response correction. Peak control is accomplished by small amounts of multiband limiting running at slower time constants. Main peak control is by the main clipper and composite processor.

WIDE is essentially the Energy (above) preset with the AGC section’s Sum and Difference channels offset in drive and makeup gain to present a more widened stereo sound field. We took care to make the sound field wider, but not so wide as to create problems for stations with multipath issues. If you like the sound of Wide but experience increased multipath on your station, you can try reducing the STEREO WIDTH limiter setting to 70% in the Stereo Encoder menu.

Folk: *Big, Medium, Soft*

BIG is tuned similarly to Medium (below), but has deeper gain control in the multiband section with multiband limiting as the preliminary peak control. The “Wide” algorithm is also used in the multiband section to maintain a more favorable texture with complex instruments. Final peak control is via main clipper with a small amount of composite processing dialed in.

MEDIUM trades off some absolute cleanliness for additional loudness by using the multiband limiters instead of look-ahead limiting as preliminary peak control. Final peak control is via the main clipper with no composite processing used.

SOFT is very similar to the Classical Light preset but with additional short term dynamics control dialed in. It relies on look-ahead limiting as the primary peak control mechanism and therefore will not be competitively loud in large markets.

Hot Stuff

HOT STUFF combines energetic qualities with deep activity in the multiband limiter, deeper dynamic frequency response correction in the multiband AGC, and moderate amounts of multiband limiting. A moderate amount of the final peak control is accomplished by utilizing both the main clipper and composite processor. This preset has some similarities to the CHR Hotter preset.

Instrumental: *Big, Medium, Soft*

BIG is energetic and smooth, utilizing about the same levels of short term dynamics control and dynamic frequency response correction as Country Easy. It relies more heavily on limiting in the multiband section, and final clipping is approximately the same density as in Country Easy, with primary peak control accomplished by the main clipper. A small amount of composite processing is also used in this preset.

MEDIUM is tuned for additional loudness by using more drive to the multiband limiters and instead substitutes main clipping as preliminary peak control. Final peak control is via the main clipper with no composite processing used.

SOFT is a “gentle” preset and maintains a very flat dynamic frequency response. It alters long term dynamics only, leaving short term dynamics untouched, giving a sense of “depth” to the music. It utilizes look-ahead limiting in lieu of clipping as the final peak control method, and therefore the preset will be very clean but not quite as competitively loud.

Jazz: *Big, Light, Smooth*

BIG is energetic but smooth and utilizes the same levels of short term dynamics control and dynamic frequency response correction as Country Easy. It borrows some Sum/Difference tricks from the Wide presets, and it relies more heavily on limiting in the multiband section. Primary peak control is accomplished by the main clipper with a small amount of composite processing added to increase loudness.

LIGHT is tuned similarly to Country Easy, but has a lower High band crossover point in order to prevent unnatural dynamic high frequency equalization. Multiband AGC action is fairly gentle with little contribution from the multiband compressors. Preliminary peak control is via the multiband limiters. No main clipping is used in this preset; however a small amount of composite processing is utilized.

SMOOTH is a more aggressively compressed preset with good medium-term dynamics control and an average amount of dynamic frequency response correction. Peak control is done by small amounts of multiband control running at somewhat slower time constants, with main peak control being done by the main clipper and a small amount of composite processing.

Latin: *Hot!, Medium, Mild*

HOT! combines energetic qualities of the CHR Hot preset with more activity in the multiband limiter, deeper dynamic frequency response correction in the multiband AGC, and moderate amounts of multiband limiting. A small amount of final peak control is accomplished by both the main clipper and composite processor.

MEDIUM combines some less energetic qualities with less deep activity in the multiband limiter, less deep dynamic frequency response correction in the multiband AGC, and moderate amounts of multiband limiting. A moderate amount of the final peak control is accomplished by utilizing both the main clipper and composite processor.

MILD combines some less energetic qualities of Medium with a little less activity in the multiband limiter. There is also less deep dynamic frequency response correction in the multiband AGC, and a bit less multiband limiting. Final peak control is accomplished by sharing this task with both the main clipper and composite processor.

News Talk: *Light, Loud, Medium*

LIGHT is tuned to be compressed but not aggressively so. A small amount of dynamic frequency response correction is used, as well as moderate amounts of drive to the multiband limiter section to increase density. Clipping is minimal. A more “compressed feel” is accomplished by using the look-ahead limiter, and final peak control is by the main clipper with no composite processing.

LOUD borrows many of the same settings from the Compressed Loud preset, but has a more relaxed sound due to less depth of dynamic frequency response corrections being allowed. The Super Low crossover has been set to 120Hz, preventing the SL band from bringing up very low frequency material. A moderate amount of multiband limiting is used to further increase program density. Primary peak control is performed by the main clipper, in addition to moderate composite processing.

MEDIUM has been tuned to be more compressed than Light. A larger amount of dynamic frequency response correction is permitted than in Light, and it also uses moderate amounts of drive to the multiband limiter to increase density. Clipping is minimal. A more “compressed feel” is accomplished by using the look-ahead limiter, and final peak control is by the main clipper, with additional composite processing added for loudness.

Oldies: *Big, Medium, Soft*

BIG uses some of the Sum/Difference channel attributes of the Dance Wide preset, but also leans more heavily into the multiband limiter section and multiband AGC/compressors. Preliminary peak control is via the multiband limiters, with a small amount of final clipping permitted. Primary peak control is via moderate amounts of main clipper drive as well as a small amount of composite processing.

MEDIUM is tuned similarly to the Dance Smooth preset and uses more gain control in the multiband AGC and multiband limiters. Primary peak control is via the main clipper with a small amount of composite clipping added to give it a competitive boost

SOFT is tuned similarly to the Country Easy preset, but has a lower High band crossover point in order to prevent unnatural increases of high frequency noise and hiss on older recordings. Preliminary peak control is via the multiband limiters. No main clipping is used in this preset; however a small amount (0.5dB) of composite processing is utilized.

Punchy

PUNCHY has further relaxed time constants similar to the Bass Punchy preset, primarily in the five band section's attack times. This preset allows a moderate amount of multiband limiting with sloped time constants (slower on lows, faster on highs) and utilizes a moderate amount of drive in both main clipping and composite processing.

Radio Active

RADIO ACTIVE combines settings from the CHR Hot, Dance Energy, and Hot Stuff presets to create an energetic, fun to listen to, very alive on-air sound. The preset uses generous activity in the multiband AGC, deep offsets in the crossover frequencies, generous amounts of stereo enhancement and moderate multiband limiting. A carefully chosen amount of final peak control is accomplished by utilizing both the main clipper and composite processor.

Sports: *Light, Loud, Medium*

LIGHT is tuned to be compressed but not aggressively so. A small amount of dynamic frequency response correction is used, as well as moderate amounts of drive to the multiband limiter, which increases density. Clipping is minimal. The look ahead limiter is used and final peak control is by a main clipper with no composite processing.

LOUD borrows many of the same settings from the Compressed Loud preset, but allows more dynamic frequency response corrections. The Super Low (SL) crossover has been set to 120Hz, preventing the SL band from bringing up very low frequency material. A moderate amount of multiband limiting is used to further increase density. Primary peak control is performed by the main clipper, in addition to light composite processing.

MEDIUM is tuned to be more compressed than Light. A larger amount of dynamic frequency response correction is used than in Light, and it uses moderate amounts of drive to the multiband limiter to increase density. A more "urgent" and authoritative feel is accomplished by using the look-ahead limiter driven moderately hard. Final peak control is by a main clipper with some additional composite processing.

Urban: *Heavy*

Urban HEAVY is similar to the Dance Energy preset with the AGC section's Sum and Difference channels offset in drive and makeup gain for a widened sound field to give a sense of space to the preset. There are also faster release times used in the compressors (but not the AGCs), further increasing short-term density. Clipping is carefully controlled in the main clipper by offsetting the thresholds in the multiband limiter and final clipper sections.

Future Vorsis Preset Updates

From time to time there will be additional presets posted on the Vorsis website that were created by the Vorsis development team and/or submitted by our customers. These presets will be contained in "Preset Packages" and named appropriately for the Vorsis product that they are compatible with.

As of this writing there have already been two revisions to the Factory presets for most Vorsis processors. The latest presets contain presets that have been finely tuned during actual field use and with our customers' best suggestions.

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

Cautionary note: Although we make every attempt to ensure preset compatibility across versions of the same model, because new features are added to the product all the time a perfect match cannot always be guaranteed.

Vorsis FM-2000 GUI

Chapter Contents

Architecture of the FM-2000 GUI	3-5
Network Configuration	3-5
Getting Started	3-5
Connecting the FM-2000 and the GUI	3-5
Using a LAN Connection	3-5
Configuring Remote GUI	3-7
Connecting Directly Without a LAN	3-8
PC System Requirements	3-8
Wireless Connection	3-8
The Really Remote Connection!	3-8
VPN	3-9
The FM-2000 and Internet Security Concerns	3-9
FM-2000 Ports and Network Protocols	3-9
Using the GUI	3-10
Dynamic Displays Region	3-11
Frequency-Domain Graph	3-11
FM-2000 Graphical Interface Operation	3-12
First, a short tutorial	3-13
Bargraph Metering	3-14
Gain Reduction Meter	3-15
60dB Range	3-15
Display Signal Selection	3-15
Input	3-15
FM preDelay	3-15
Audio	3-16
Gain Reduction	3-16
Controls	3-16
L/R (Left/Right)	3-16
O-Scope	3-16
Control Area Region	3-17
Input Menu	3-18
Input Settings	3-18
Input Source	3-18
Input Signal Presence	3-18
Input Failsafe	3-18
Analog Gain	3-19
AES Gain	3-19
L/R Balance	3-19
Auto Mono Threshold	3-19

What is Auto Mono?	3-19
Phase Rotator	3-20
HPF - High Pass Filter	3-21
Why Would We Want To Do This?	3-21
VoiceMaster	3-22
Enable	3-23
Spatial Sense	3-23
Spectral Sense	3-23
Lim Drive	3-23
Pk (Peak) Ceiling	3-23
Lim Atk (Attack)	3-24
Lim Rls (Release)	3-24
AGC Atk (Attack)	3-24
AGC Rls (Release)	3-24
VoiceMaster Equalization	3-25
A Note Regarding VoiceMaster Equalization	3-26
Parametric Equalizer Menu	3-27
Enable	3-28
Automatic Gain Control and Compressor Menus	3-29
Drive	3-30
Makeup	3-30
SST/AGC/Comp Buttons	3-30
The Eight AGC Operating Modes: 0 through 7	3-31
SST Controls Overview	3-32
SST Drive	3-32
SST Gate Thresh	3-32
Ramp Rate	3-32
Window Size	3-33
AGC Controls Overview	3-34
Gated Modes	3-34
Ooze	3-34
Freeze	3-34
AGC Gate Threshold	3-34
Gate Delay	3-35
AGC Backoff	3-36
Band Coupling	3-36
Crossover Frequencies	3-37
Selecting Crossover Frequencies	3-37
AGC/Compressor Gain Reduction	3-39
FM Limiter Menu	3-40
31 Band Limiter	3-40
Multiband Enable	3-40
Multiband Drive	3-41
Lim/Clip (Limiter/Clipper) Drive	3-41

Multiband Knee	3-41
Vorsis Bass Management System (VBMS)	3-42
VBMS Drive	3-42
VBMS Style	3-43
VBMS Output	3-43
VBMS Frequency	3-43
Emphasis Selection	3-44
DeEmphasis	3-44
Look Ahead Limiter	3-44
FM Limiter Controls	3-45
Limiter Enable	3-45
L/R Linked	3-45
Limiter Thresh	3-45
Attack	3-45
Release	3-45
Delayed Release	3-46
Digital Output (dBFS)	3-46
Clip Style (Main Clipper)	3-46
FM Diversity Delay	3-47
Entering Delay Value	3-47
Special 31 Band Operating Settings — Wide and Timbral	3-48
31 Band Limiter Dynamics Screens	3-49
Thresholds Screen	3-49
Multiband Mode	3-50
Attack Time Screen	3-52
Release Time Screen	3-53
Coupling Controls Screen	3-54
Final Equalization Screen	3-55
VoiceMaster Equalization	3-56
A Note Regarding VoiceMaster Equalization	3-57
Stereo Encoder Menu	3-58
Stereo Width Limiter	3-58
Lim/Clip (Limiter/Clipper) Drive	3-59
Test Oscillator	3-59
Multiplex Process	3-60
Output Metering	3-60
Stereo Encoder	3-61
Pilot	3-61
SCA 1	3-61
SCA 2	3-61
Analog L/TX 1	3-61
Analog R/TX 2	3-61
Pilot Only	3-61
Pilot Phase	3-62
Multiplexer	3-62
Mask Filter	3-62

Analog—Stereo Coder Output Options	3-62
System Menu	3-63
System Settings Change With Preset Takes	3-63
Headphone Source	3-64
Headphone Control	3-65
Comp./AGC Bands	3-65
Remote Login	3-66
GUI Access	3-67
GUI Lock	3-67
Indicators/Temperature Readout	3-69
Skin	3-69
Events	3-70
Set Time	3-72
Backlight Dimmer	3-72
Side Bar Region	3-73
Preset	3-73
Save	3-73
Library	3-74
Locking Presets	3-75
Devices	3-76
Quick Save (QSave)	3-76
Title Bar Region	3-78
Status	3-78
Devices	3-78
Presets	3-79
Notes on “Online” and “Offline” Working	3-79
Accessing Menu Options	3-80
File Menu Items	3-80
Hardware Menu Items	3-80
FM-2000 Hardware Update	3-81
VBN/LBN Updating Order Sequence	3-85
Presets Menu Items	3-85

Vorsis FM-2000 GUI

Architecture of the FM-2000 GUI (Graphical User Interface)

The FM-2000 GUI allows you to adjust the processor as well as view in real time how it is reacting to the audio passing through it. The FM-2000 GUI has been very carefully designed to put parameters never more than two mouse clicks away. This is a radical change from competing products where constant navigation through numerous layers of menus and screens is required.

Network Configuration - Vorsis FM-2000 GUI

The Vorsis GUI PC software allows you to control an unlimited number of FM-2000's via a standard Ethernet local area network (LAN). Utilizing TCP and UDP protocols, the GUI can be used to control FM-2000 devices from anywhere with an Internet connection, including those behind firewalls and through VPN tunnels.

Getting Started

The FM-2000 GUI is a Windows™ based program and is designed to be intuitive and straightforward in use. It is supplied as a self-installing program executable that may be installed from any directory on the computer. The resulting files will be installed under “\Program Files\Vorsis\FM2000.” After installation, select the shortcut in the Windows Start menu to start the GUI program.

To configure the FM-2000 network connection a basic familiarity with Windows navigation techniques is assumed. If terms such as “left-click,” “right-click,” “double-click,” and “drag-and-drop” seem alien, operating the FM-2000's front panel or remote GUI could be a bit problematic.

Connecting the FM-2000 and the GUI

Using a LAN Connection

The FM-2000 and the PC running the FM-2000 GUI program can be straightforwardly connected together over a standard Ethernet Local Area Network (LAN).

The FM-2000 should be installed, powered-up, and verified to be operating normally. A CAT5 Ethernet cable connected to the 100baseT LAN should be inserted into the appropriate socket on the FM-2000. Likewise, the GUI should be installed on the desired PC and that PC verified to be working correctly with the LAN.

Configure the desired hardware TCP/IP address for the FM-2000 by using the front panel's "Right Click\File\Settings" navigation to get to the IP address page see below:

FM-2000's Front Panel Network Settings

On this page you will see the following four entries (default settings shown):

GUI IP Address - This is the reserved address that the front panel (not the remote GUI on the PC!) uses to communicate with the FM-2000 system board. This address should ALWAYS be configured to be one number *higher* than the IP address that you'll assign to the FM-2000's "Hardware" address, but in no case should it end in "0" or "255."

Hardware IP Address - This is the address of the FM-2000's DSP farm and is the address that the remote GUI on another computer will use in order to connect to the FM-2000 over your network.

Subnet Mask - The subnet mask entry must be appropriate for the network being used. Please see your local friendly IT person if this is a bit confusing.

Gateway - This is the gateway address that the FM-2000 will use to reach connections outside of its own internal network. If you are on a local segment of LAN or connected to the FM-2000 by a direct connection, the Gateway address is not usually necessary.

After determining the IP address that you wish the FM-2000 to use, enter that address in the FM-2000's Hardware IP address field using the navigation buttons. Please clear the entire IP address field before entering new data since this ensures there will be no hard to see leading or training blanks in the address.

Next, using the same sequence as above enter a new GUI IP address by adding “1” to the IP address that you entered for the hardware in the above step.

Example:

If you entered 192.168.0.200 as the hardware address, then the correct GUI address will be 192.168.0.201. Note that the GUI address is *not* utilized externally by the FM-2000, so there are no concerns whatsoever about traffic to/from this address external to the FM-2000 itself. However, this address will appear on your network so this should be kept in mind and this address reserved for the FM-2000.

Enter the correct Subnet Mask and Gateway in the same manner as the IP addresses were entered. Then click the “OK” box. When prompted to reboot the FM-2000 for the changes to take effect, click “OK” and then interrupt power to the unit for approximately five seconds. Reapply power, and once the FM-2000 has rebooted and come back to life, the new IP address will be valid.

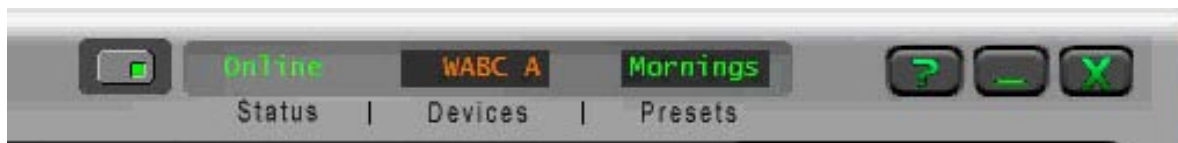
Configuring Remote GUI

Now that the FM-2000 has a valid network address the remote GUI must be configured so that it can find it on the network. Install and then start the remote GUI. In the remote GUI, click Devices>Add... to bring up the Edit Device form. Fill in the pet name you’ll be giving to the unit. In the IP Address field below the name fill in the “HARDWARE” address that you just gave the unit in prior steps and then click OK.

Highlight the new FM-2000 device in the Devices list and click on “Select.”

Place the GUI online by clicking on the button to the left of the Status message - it should turn green when the FM-2000 connects, indicating that the GUI is now communicating with the FM-2000.

Once the GUI has been made aware of the FM-2000’s existence in this manner, it will always appear in the list of “FM2000 Devices,” and be instantly accessible – from anywhere.



FM-2000 GUI Status Bar

When connected to an FM-2000 the status bar will indicate “Online” and the “Devices” text will show the name of the FM-2000 that it is currently communicating with. The status bar will also display the name of the preset that the FM-2000 you’re connected to is currently running.

When the Preset name is Green it represents a preset (factory or user) that has been recalled and is running with no changes made to it. If the Preset name is scrolling in Red it means that the preset has had changes made or is a work in progress not yet saved.

Connecting Directly Without a LAN

You can work without a LAN by connecting the FM-2000 and the PC Ethernet ports together with a standard Ethernet cable - either a straight through or crossover cable may be used because the five-port 10/100 Ethernet switch inside the FM-2000 has auto-sensing ports.

Note: The controlling PC and the FM-2000's network settings must be configured to place them both on the same subnet! This is also why the front panel GUI and remote hardware GUI must be one IP address from each other – although the internal subnet can be any one that is valid, both local and remote devices must be on the same subnet in order to communicate.

PC System Requirements

Constraints for running the GUI are not extraordinary — the PC should be preferably at least 1GHz in speed, running Windows 2000 or XP. Screen resolution should be at least 1024 x 768 pixels in order to get the full benefit of the graphics used. You need about 15MB of space on the hard drive for the GUI software and presets. The PC's LAN interface (NIC) should be 100baseT (100MHz) capable for best performance.

Wireless Connection

Many users will want to make adjustments to the FM-2000 while in a remote location such as their home, office, or car. This can be accomplished in a variety of ways.

The GUI can be run on a portable computer connected to the FM-2000's LAN by a wireless (802.11b as a minimum) device, as long as the FM-2000 itself "sees" a 100baseT network. This can be done by connecting the FM-2000 to a readily available Wireless Access Point and then connecting to the FM-2000 through this access point from a wireless interface on a laptop computer.

A fairly high quality connection is required in order to have full advantage of the real-time graphics displays on the remote GUI. Low speed connections will still function, but graphics performance may be degraded.

The Really Remote Connection!

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

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

We will leave the myriad other ways in which the FM-2000 can be remotely controlled "wirelessly" up to your imagination and the rapid changes in wireless technology!

VPN

Another way to connect remotely to the FM-2000 is via VPN. A VPN, or Virtual Private Network, is a private data tunnel between computer devices. It behaves just as if the local and remote computers were sitting beside each other — devices connected via VPN act like they are on one big LAN regardless of where they are physically located. VPN tunnels can be configured both with and without data encryption. Setting up a VPN is not difficult, but certainly beyond the scope of this manual.

The FM-2000 and Internet Security Concerns

The FM-2000 does not act as a “web device” or “web server,” nor does it support open ports to the Internet. Therefore there is no worry that someone could “hack” into the FM-2000 and use *it* as a pathway to the rest of the network to which it is connected.

About DHCP and the FM-2000

The FM-2000 does not utilize DHCP — Dynamic Host Configuration Protocol (or automatically assigned network addressing) — and needs to be assigned a “STATIC” network-unique IP address.

Most DHCP servers assign addresses starting at the bottom of the group 192.168.0.xxx. Therefore choosing an address high in that group, say 192.168.0.200, as a static IP address will likely keep it out of the way of the busiest DHCP addressing. However, it’s always wise to check with your local friendly network administrator to determine what static IP address is open and safe to use.

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

FM-2000 Ports and Network Protocols

The FM-2000 uses both TCP and UDP protocols to communicate with the remote GUI.

TCP is used from GUI to FM-2000 because its high reliability ensures that control changes sent to the FM-2000 will be received.

UDP is used from FM-2000 to sent and update metering and other real time data.

TCP uses port 55897, while UDP uses port 60001, and, if 60001 is busy, tries the next highest port until an idle one is found.

Both TCP and UDP ports must be open for any firewall-based connectivity.

Using the GUI

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

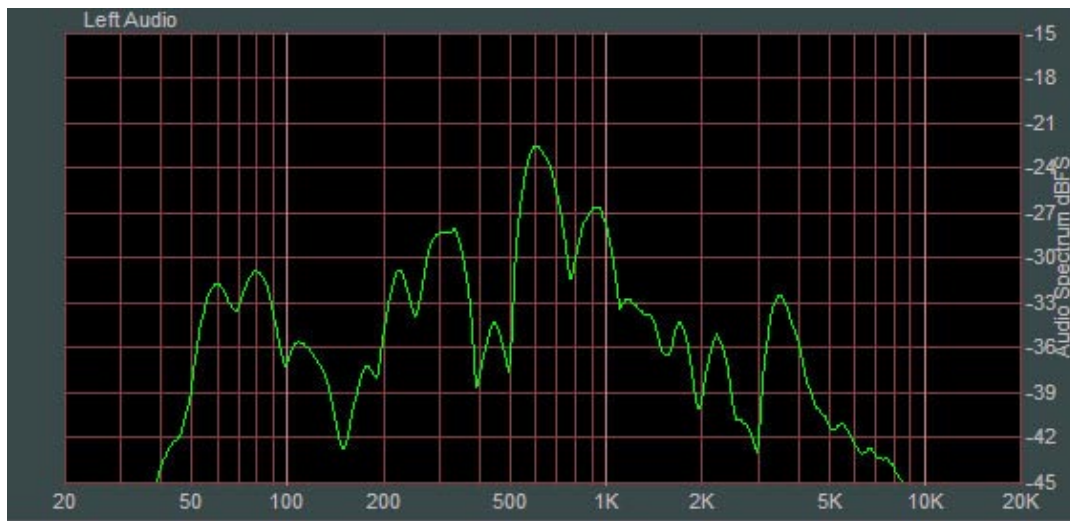
Descriptions of the features and functions of the GUI will follow different discrete regions of the screen; it will begin with the most eye-catching region — the Dynamic Displays — and then we'll explain the Control Area above it, followed by the Side Bar buttons, and lastly the Title Bar.



Dynamic Displays Region

Frequency-Domain Graph

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



FM-2000 Audio Spectrum Display

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

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

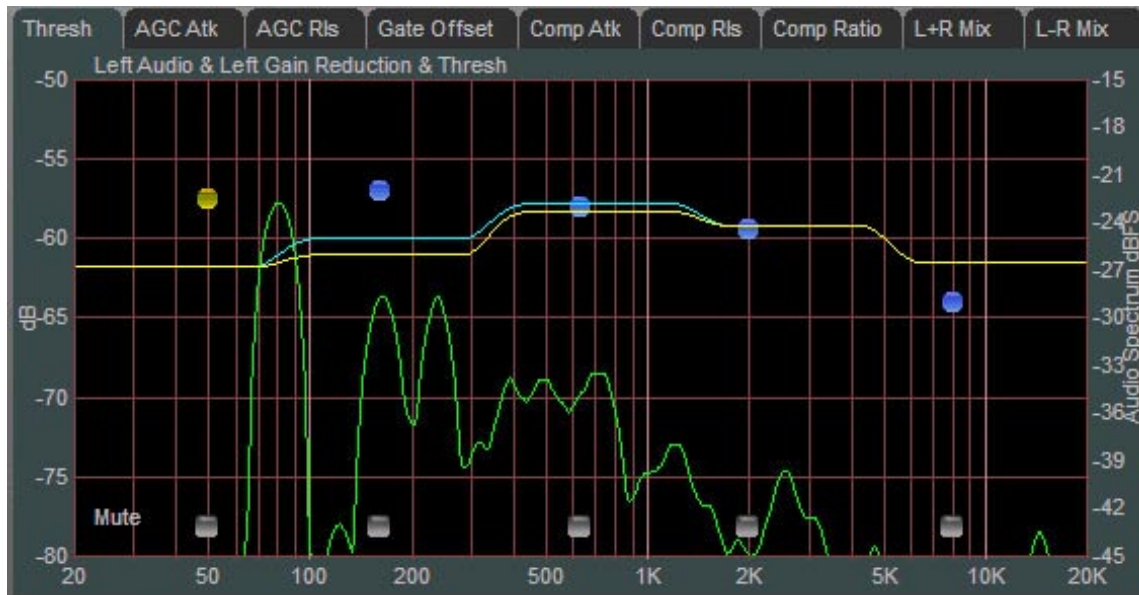
FM-2000 Graphical Interface Operation

Select the FM Lim (or perhaps AGC/Comp. as shown below) from the buttons at the top of the Control Area (see page 3-10) and then make sure there is a check mark in the Controls checkbox at the bottom of the Dynamic Displays area.

We've worked very hard on the design of the Vorsis graphical user interface to make it intuitive, friendly, and easy to navigate and interpret. Each of the processing section's screens has a graphical area where not only visual representations of processing activity are shown, but in some cases individual adjustments of different sections of the processing can be made.

All of the operating parameters of the five band AGC and the 31-band FM limiter can be individually adjusted by manipulating the positions of various "Blue Dots" on that sections' graphical screen.

Each of the blue dots on the screen represents the center frequency of a band. By double clicking a blue dot (to highlight it and turn it Orange) and then dragging it with the mouse a particular parameter for each band can be adjusted.



Example Graphical Control Screen

There are other ways to set parameters, too. In fact *all* of the controls on the graphical screen work the same way, and each parameter to be adjusted is selected by clicking on the tabs at the top of the screen. Once you've learned one screen, you'll know how to make adjustments in any other!

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

Group Adjusters

UnGroup Adjusters

Group All Adjusters

UnGroup All Adjusters

Flatten

Proportional Drag

Flat Drag

First, a short tutorial....

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

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

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

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

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

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

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

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

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

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

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

To review:

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

UnGroup Adjusters — disconnects any **blue** dots that were grouped by “Group Adjusters.”

Group All Adjusters — gangs all the dots, turning them **blue**, *regardless* of their previous highlighting state.

UnGroup All Adjusters — ungangs *all* the adjustment dots, turning them **orange**, regardless of any previous highlighting.

Flatten — removes any variance to adjacent dots not being horizontal to each other.

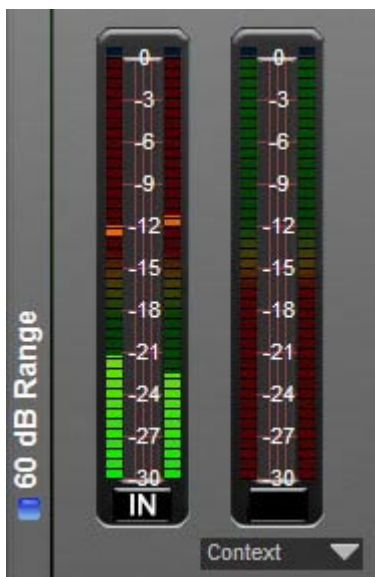
Proportional Drag — allows the dragging of adjustment dots to create slopes.

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

Bargraph Metering

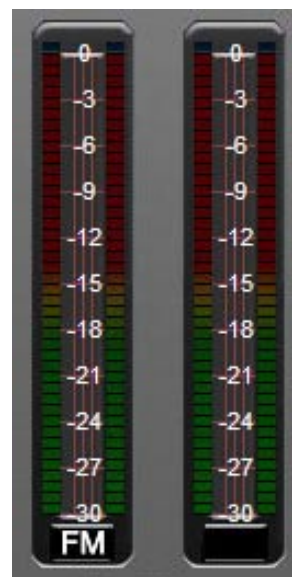
Adjacent to the graph to the left and right, and sharing a 30dB vertical range (0-30dB of gain reduction, 30dB signal level meter range), are a group of vertical bargraph level indicators for signal levels and gain reduction.

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



Left of Graph

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



Right of Graph

Left pair - FM Output Levels (see page 3-60)
Right pair - Unused

Gain Reduction Meter (Left of Graph)

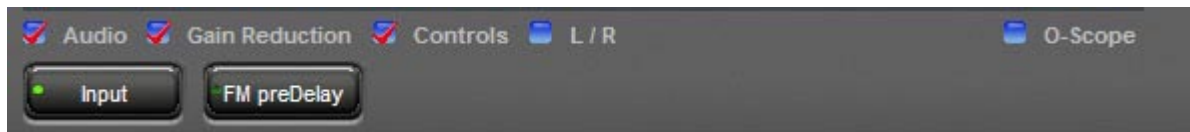
The actual function of this meter is selected in the drop down list below it. You can select one of the self-explanatory functions, or allow the meter function to change depending upon the processing section being viewed in the Control Area Region by selecting “Context” from the list.

60dB Range

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

Display Signal Selection

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



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

Input

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

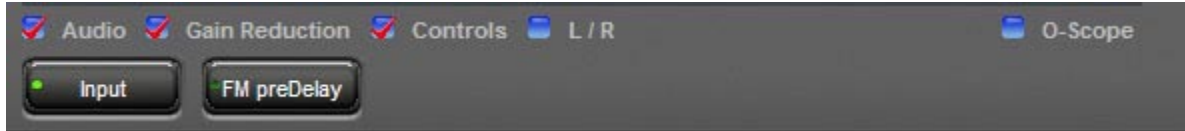
The FFT¹ is widely used in test equipment in the audio industry because of its extremely precise measurement capability. Thanks to the extremely efficient DSP chips utilized in the FM-2000, we are able to offer the same exotic measurement capability in an audio processor. Thus what you see on our graphics displays is as precise as current technology permits.

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

FM preDelay

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

Just above the two large buttons, and below the Graphical display, is a row of several checkboxes. We will describe these in order, starting with the left-most one.



Audio

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



When the Input button is illuminated, the FFT shows the audio spectrum of the Input audio.



When the FM preDelay button is illuminated, the FFT shows the audio spectrum of the output of the FM processing chain prior to the diversity delay.

Gain Reduction

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

Controls

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

L/R (Left/Right)

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

O-Scope




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

Control Area Region

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



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

-  place the cursor over the desired control, and use the mouse’s scroll wheel to “increase” or “decrease” the control’s position, or
-  click on the control, and slide the mouse to move the control - up or right increases the control’s indicated position, down or left decreases it. When using the mouse in this fashion (as opposed to using the scroll wheel), the resolution of the control variation can be increased — made finer — by holding down the keyboard’s Ctrl key while adjusting the control as above, or
-  in most cases you can double click on the text that indicates the setting value (for example, the window that reads X.XdB under the Line Gain knob on the Input Section) and a box will pop up where you can enter an exact value and click OK.

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



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

INPUT MENU

Input Settings

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



Input Screen

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

This screen also has the operating controls for Vorsis' exclusive VoiceMaster algorithm, which ensures pristine voice quality on the air regardless of the competitive state of the radio station.

- ☞ All controls can be adjusted with (1) the FM-2000 front panel track pad, (2) a USB mouse connected directly to the FM-2000 front panel, or (3) by the Windows-based remote GUI.

Input Source

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

- ☞ The currently selected input is shown by a red checkmark.

Input Signal Presence

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

Input Failsafe

In the event the currently selected input source fails and the "Input Failsafe" checkbox is checked, an alternate source of audio connected to the other input can automatically be put on the air.

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



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

Analog Gain

The Analog audio input gain can be adjusted by clicking on the knob and dragging the mouse or touchpad to the right or left. The Line Gain level can be adjusted over a +/- 24dB range.

AES Gain

The AES/EBU Digital audio input gain can be adjusted by clicking on the knob and dragging the mouse or touchpad to the right or left. The AES Gain can be adjusted over a +/- 24dB range.

L/R Balance

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

Auto Mono Threshold

The Auto-Mono threshold can be enabled by clicking on the checkbox for it and the level at which mono blend occurs can be adjusted with the Auto-Mono threshold knob.

What is Auto Mono?

Auto Mono is a feature that automatically and continuously analyzes the L+R/L-R ratio of the incoming program material to determine if the content in the L-R difference channel is valid. If iAuto Mono is enabled and the input signal is not true stereo, then that information is discarded and the output of the processor is pure mono.

This feature is most useful in formats that play music such as Oldies, Solid Gold, or other program that was primarily mastered in mono or that is intended to be mono but might contain small left/right channel balance errors. These errors result in information in the difference channel that has no meaningful purpose and as such that information may be reduced or eliminated with no detriment to the audio.

Most oldies music was originally mastered in mono but in recent years many of those songs may have been ripped into playout systems using audio compression schemes such as MPEG Layer II or MPEG 3 (MP3). Although the songs may have been originally recorded in mono, they were likely ripped into the playout system while it was in stereo mode, which means there could be uncorrelated stereo information in the L-R which is basically “junk.”

The Auto Mono circuit has been designed to be unobtrusive in its action while removing such “junk.” Therefore it may typically be left enabled so that it has an opportunity to correct any material that may pass through the FM-2000 that requires the difference channel to be cleaned up.

A typical operating threshold setting for the Auto Mono circuit with most program material is usually between -20dB to -30dB. Higher settings (less than -20dB) may cause unnecessary falsing on certain material. Lower settings (below about -30dB) may allow suspect material to pass uncorrected. The control may be adjusted over the range of -40.0dB to 0.0dB.

Phase Rotator

Human voice is usually quite asymmetrical in nature, which means that it is usually “peakier” in one polarity than the other. What the Phase Rotator algorithm does is fix these asymmetrical peaks. The way it works is by phase shifting the signal harmonics relative to their fundamental frequencies. This action reduces the peak to average ratio and enables an increase in apparent loudness with minimal audible detriment. We do this because processing the audio for a symmetrical medium like FM is greatly simplified if the waveforms are symmetrical.

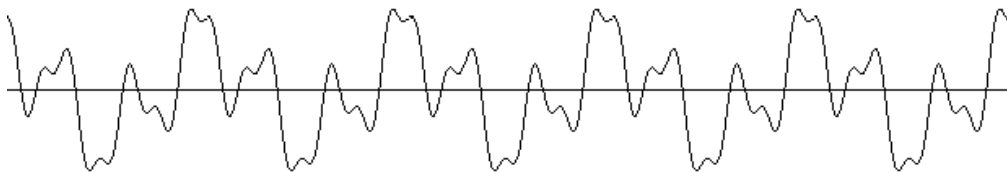
Ordinarily, human voice looks like the waveform below. Notice how the width and amplitude of the upper and lower signal peaks are different? This is ‘asymmetry’, and makes the processing of voice waveforms more difficult.



Human voice before “Phase Rotator” algorithm

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

No information is lost and, more importantly, no harmonic or intermodulation distortion is created by the Phase Rotator. The only thing that happens is that the signal’s phase relationships are rearranged to make the waveform energy more uniform in polarity.



Human voice after “Phase Rotator” algorithm

HPF - High Pass Filter

The High Pass Filter is used to remove inaudible and unnecessary subsonic energy from the audio signal prior to it being processed. By removing this energy, processing is cleaner and modulation energy is not wasted by transmitting sounds that would not be perceptible to a listener. The High Pass Filter is also useful in removing subsonic energy that could upset the operation of equipment later in the audio chain, such as the AFC loop in the FM exciter.

The High Pass Filter has three operating modes:

Off – the High Pass Filter is not in operation.

Stereo – When checked, the High Pass Filter operating mode is stereo and both left and right audio channels are treated equally by the high pass filter frequency.

Sum/Diff – If the High Pass Filter is *not* operating in Stereo mode (Stereo unchecked) and either High Pass Filter checkbox is checked, then the High Pass Filter will operate on the Sum and Difference signals separately. This means that the sum (mono) information and difference (stereo) information may have entirely different high pass filter settings.

Mode is selected as follows:

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

Why Would We Want To Do This?

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

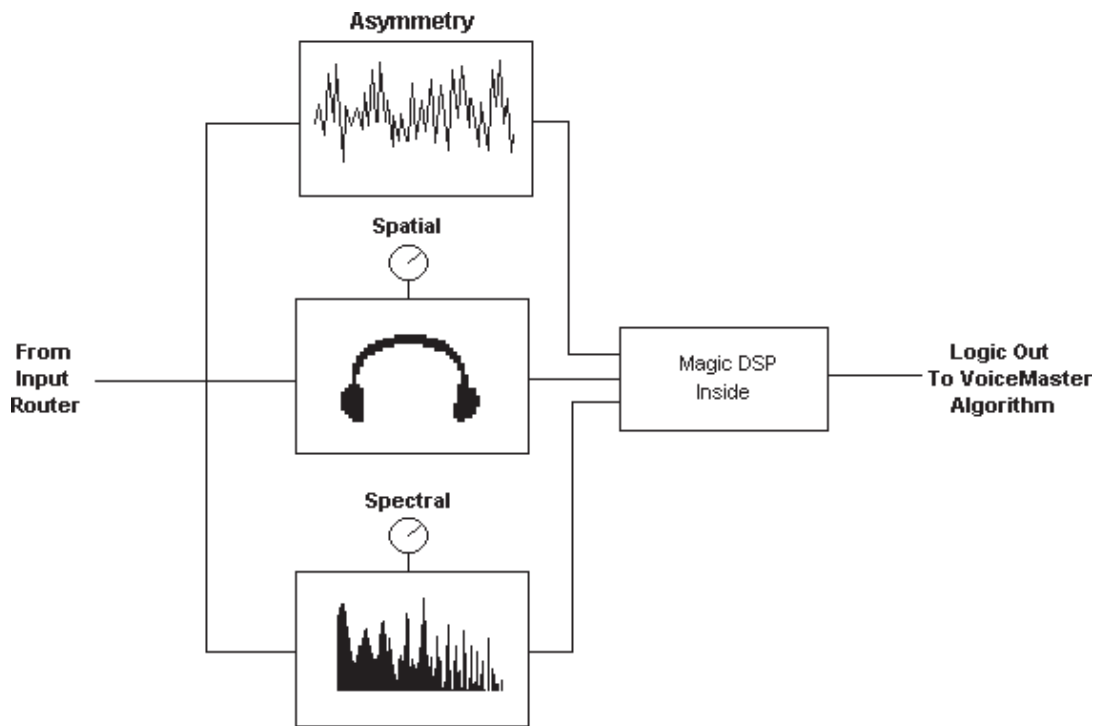
The sum (mono) channel, on the other hand, carries most of the bass energy in the majority of program material, and therefore it cannot utilize as high a cutoff frequency as the difference channel without seriously impacting the bass feel of the program.

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

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

VoiceMaster

The FM-2000's exclusive VoiceMaster algorithm is quite unique in that it uses several different measurement schemes to make decisions about whether the input program material is voice. This assures successful voice detection with a minimum of "falsing." The block diagram below shows the three essential detector algorithms that make up the fundamentals of the VoiceMaster algorithm.



VoiceMaster - Basic Logic Diagram

The sensitivity of two of the detectors is adjustable over sensible ranges to enable the algorithm to be tailored to the individual station's air chain and its electrical performance (left/right channel amplitude and phase balance) under monophonic programming conditions (Asymmetry operation is automatic).

All three detectors are utilized to make decisions about whether or not the incoming program material is mono and if it is also voice. Satisfaction of any single detector (or pair of detectors) does not constitute a logical assurance that voice is present. All three detectors must be satisfied before the algorithm decides that isolated voice program is present.

Because all three detectors must be satisfied before the output of the VoiceMaster algorithm commands the processing to make changes in how it handles program energy, there must be a mechanism to "tune" the algorithm for each individual station's audio chain behavior. The controls that affect how VoiceMaster reacts to voice and other programming are relatively simple to adjust.

Enable

Turns the VoiceMaster voice detection algorithm on and off. When the Enable box is unchecked, no correction for voice energy will be made.

Spatial Sense

Adjusts the sensitivity of the algorithm to the amplitude and phase behavior of the left and right stereo input channels when monophonic program energy is being detected. The more negative the setting, the less the difference energy in the L-R that is required in order to satisfy the detector. The control may be adjusted to settings between -24dB and -60dB. A good starting value is between -25.0dB and -30.0dB.



Spectral Sense

Adjusts the sensitivity of the algorithm to program energy falling inside the typical voice frequency band when compared to frequencies outside of that band. The more negative the setting the less spectral differences between the inside and outside of the human voice frequency band will be required in order to satisfy the detector. The control may be adjusted to settings between 0dB and -24dB. A good starting value is between -2.0dB and -6.0dB.

Lim Drive

Adjusts the amount of drive to the special VoiceMaster limiter whenever the VoiceMaster algorithm has determine that voice is present within the incoming program. The range of this control is from -12dB (no processing by the VoiceMaster limiter) to 0dB (aggressive processing). A good starting value is between -3.0dB and -1.0dB.

Pk (Peak) Ceiling

Adjusts the maximum peak amplitude of voice energy relative to the clip threshold of the main clipper. What this means is that when the control is set to “0dB” the peak amplitude of voice that is processed by the VoiceMaster limiter exactly matches the clip threshold of the main clipper when the VoiceMaster Lim Atk. (Limiter Attack) control is set to its 0.1mS setting. No clipping of voice will occur with these settings. A good starting value is between -3.5dB and -1dB.

When the Peak Ceiling control is set to a negative number, that number then represents how much lower (in dB) the peak amplitude of voice exiting the VoiceMaster limiter will be relative to the main clipper threshold. In fact, when the control is set to *any* negative number, it is virtually assured that voice energy will *never* be clipped by the main clipper.

When the Peak Ceiling control is set to any number above “0,” then voice energy “may” be clipped by the main clipper according to how much higher than “0” the Peak Ceiling control has been set. For instance, if the Peak Ceiling control has been set to “+1.0 dB,” then the voice peaks exiting the VoiceMaster limiter will drive the main clipper harder by *exactly* that “1.0dB.”

What this means is that if the station requires absolutely pristine voice with no hint of processing distortion due to clipping, then the Peak Ceiling control should be adjusted to a setting of “0dB” or below.

If on the other hand some amount of clipping is desirable on the station in order to give announcer voice an “edge” of excitement (or perhaps some extra loudness), then the control may be adjusted to an amount above “0dB.”

The range of this control is from -6dB to +6dB.

Lim Atk (Attack)

Adjusts the VoiceMaster limiter attack time, with faster attack times (smaller numbers) acting more aggressively on voice peaks. The control can be adjusted between 0.1 milliseconds (100 microseconds) and 33.0 milliseconds. A good starting value is 10-15 milliseconds.

Lim Rls (Release)

Adjusts the VoiceMaster limiter’s release time, with faster release times (smaller numbers) acting more aggressively on voice. The control can be adjusted between 33.0 milliseconds and 200 milliseconds. A good starting value is between 50 and 100 milliseconds.

AGC Atk (Attack)

Adjusts the VoiceMaster AGC attack time, with faster attack times (smaller numbers) acting more aggressively on *average* voice energy. The control can be adjusted between 100 milliseconds and 1.0 seconds. A good starting value is 100 to 150 milliseconds.

AGC Rls (Release)

Adjusts the VoiceMaster AGC release time, with faster release times (smaller numbers) acting more aggressively on *average* voice energy. The control can be adjusted between 330 milliseconds and 3.00 seconds. A good starting value is between 500 and 700 milliseconds.

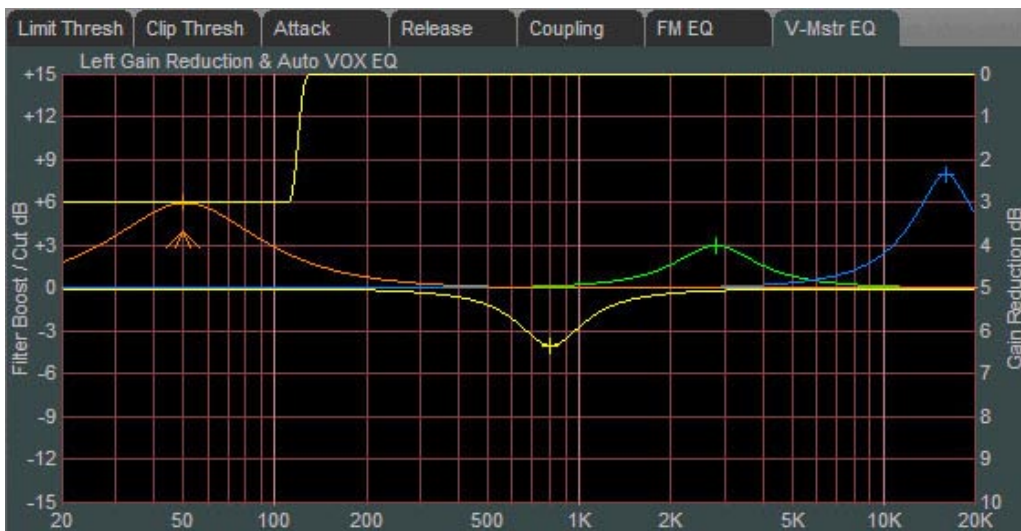


VoiceMaster Equalization

In some applications it might be desirable to add some tasteful equalization to the VoiceMaster signal path in order to enhance certain characteristics of live voice. This can easily be done by using the special VoiceMaster equalization screen located on the FM Lim menu under the V-Mstr EQ tab. See the following graphic for where the controls are located.



Below is a closer look at the equalization section of the graphical display.



The VoiceMaster equalization controls may be adjusted just as the other parametric equalizers are adjusted — using the mouse to craft the desired equalization curve shape.



Double-clicking an EQ will select it, or you can right click on the graphical area and a four-choice menu (shown above) will appear, allowing you to choose which of the four parametric equalizer sections you'd like to adjust. All four sections of the VoiceMaster equalizer are identical and each may be adjusted/moved to any part of the 20Hz to 20kHz frequency range (including overlapping them!).

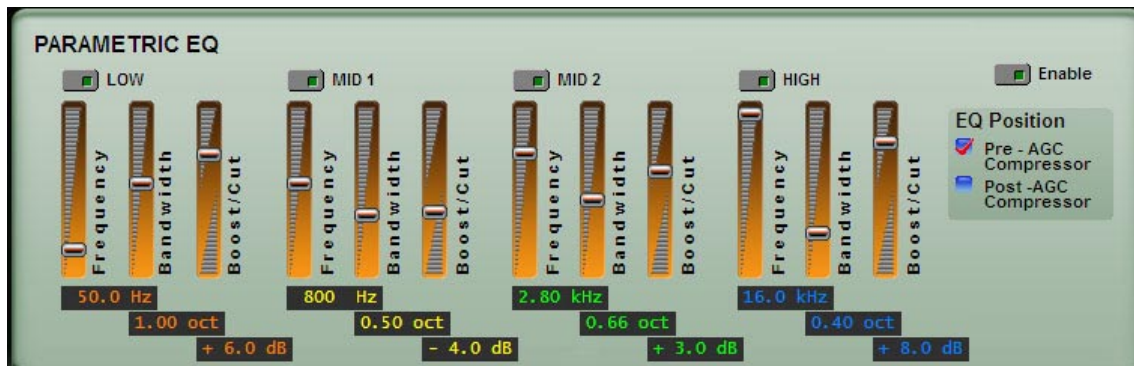
A Note Regarding VoiceMaster Equalization

The equalization for the VoiceMaster signal path is located just prior to the FM limiter/clipper. What this means is that some *care* is required when equalization boost is used.

Equalization boost of more than 3dB can be troublesome, although equalization cuts are *always* safe to make because they do not add to the pre-clipper headroom requirements.

Because the VoiceMaster's "Peak Ceiling" control is located after equalization, it can be used to tame the following clipper's behavior with live voice if more than a dB or so of VoiceMaster equalization has been dialed in.

PARAMETRIC EQUALIZER MENU



Parametric Equalizer Screen

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

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

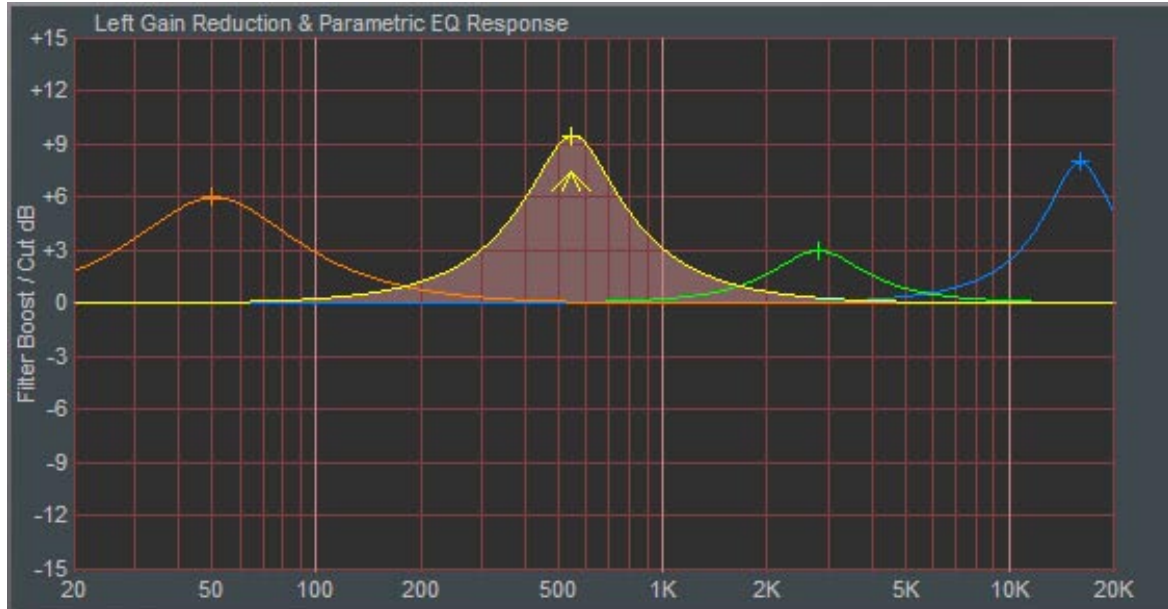
The position of the parametric equalizer in the signal chain can be selected to be prior to the multiband AGC/Compressor, or after it. The effect created by inserting the equalizer into each position is different.

When the parametric equalizer is used ahead of the multiband AGC, equalization changes predispose the AGC to operate on those changes. In the case of equalization boost, the compressors operate on the boost to do two things: (1) hold back some of the boost dialed in according to the threshold and compression ratio settings, and (2) allow more boost to occur at those frequencies when the AGC band is adding gain to the signal chain.

For instance, if equalization is applied pre-compressor and a fair amount of low and high frequency equalization is added, the compressors will operate in a way that makes them behave to create an automatic loudness contour — as compression levels go down, equalization boost goes up.

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

The second method of adjusting the parametric equalizers is by directly manipulating the actual curves on the graphical screen below using the mouse (the Controls checkbox must be checked to see the curves).



Parametric Equalizer Display

(the shaded area depicts overall response of combined sections)



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



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



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

Enable

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

AUTOMATIC GAIN CONTROL and COMPRESSOR MENUS

The FM-2000 has three sections dedicated to dynamics control, SST, AGC, and Compressor, and each is inextricably linked by DSP algorithms to logically connect them. They will be covered collectively in one chapter rather than discussed separately.

Although the SST, AGC, and Compressor sections may be separately enabled or disabled, the Compressor algorithm is *always* running (even if it's just in the background) so that certain control signals are available to the SST and AGC sections.



Five Band AGC / Compressor Screen

Unique to the FM-2000 (and AP-2000) model is a special algorithm called SST, which stands for Sweet Spot Technology. The SST incorporates control signals from the input audio, the multiband crossover, and the AGC and Compressor behaviors in order to derive a special control signal that is used to force the multiband AGC/Compressor to operate within a specified gain reduction window. By doing so, the medium and long term “sound” of the processor is quite stable as program elements and incoming audio levels wander about.

Without the contribution of the SST, the AGC and Compressor sections would not be able to maintain as consistent a spectral and dynamics texture over widely ranging signal input conditions. In fact, they would have little advantage over the technology used in other audio processors which all suffer from the inability to manage *long-term* spectral consistency.

A “broadband AGC” is not used prior to the AGC and Compressor because such devices know *nothing* about the output of the multiband section and it is *this* output that determines the overall texture of how a radio station sounds on the air. By using a multiplicity of control signals the SST is able to calculate a correction signal that exactly manages the behavior of the multiband AGC and compressor in order to create a consistent signature sound across a very wide variety of input sources and audio levels.

The automatic gain control (AGC) itself operates in five frequency bands, and its medium-term operation is a special derivative of the type of program-related processing occurring in the Compressor algorithms. While the SST section operates in Left/Right mode, the AGC and Compressors operate on the program content’s Sum and Difference — the highest signal, whether its in the sum or difference channel, determines the amount of processing to be applied to both channels.

Drive

The Drive control adjusts the amount of signal into the AGC/Compressor, primarily determining the amount of compression that can be achieved. Driving harder (higher numbers) results in a more consistent on air sound, with an increased capability to pull up low passages in program material. Less drive creates a more “gentle” on air sound and has less capability to bring up low passages. The Drive control simultaneously adjusts the sum and difference channels. The range of this control is from -80.0dB to +6.0dB.

L-R Trim — The List box below the Drive control contains the trim adjustment for the Difference, or L-R, channel, and may be adjusted +/- 6dB referenced to the Drive control. In other words, if the Drive control is set to +4.0dB and the L-R trim to -5.0dB, the difference channel drive is actually -1.0dB.

Makeup

The control adjusts the overall output level of the AGC/Compressor into the following stages. It simultaneously adjusts the Sum and Difference channels.

The AGC/Compressor is a feed-forward design and therefore the gain removed by compression must be “made up” afterwards in order to return it to a normal level. This is why the Makeup Gain control has such a wide range (-20.0dB to +48.0dB) and also why it is “normal” to see makeup gain settings in the 20 to 30dB range.

Note: The “Makeup Gain” control in the FM-2000 does not operate like a control having a similar name in another audio processor. That control would be better called “Fast Release” because of what it actually does, which is quickly add gain when the input signal falls at a certain rate compared to the amount of gain reduction taking place at that instant.

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

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

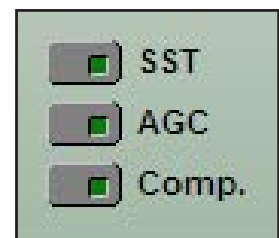
SST/AGC/Comp Buttons

These buttons determine the presence of the SST, AGC, and Compressor sections in the output.

SST Button — When illuminated, the SST (Sweet Spot Technology) is enabled and is managing the behavior of the five (or three) band AGC.

AGC Button — When illuminated by clicking on it, indicates that the output of the AGC section is in the audio path. The AGC operates on the relatively long-term variations in incoming program material and has the most contribution to the function of long-term audio level control.

Comp Button — When illuminated by clicking on it, indicates that the output of the Compressor section is in the audio path. The Compressor operates on the relatively short-term variations in incoming program material and has the most contribution to overall program loudness.



The Eight AGC Operating Modes: 0 through 7

0. SST Out*, AGC Out, Compressor Out

When all modes are off there is no processing by the AGC section. Extremely “gentle” formats might benefit from this mode if the FM limiter section is left operating and readjusted accordingly.

1. SST In, AGC Out, Compressor Out

The SST is the only dynamics modifier operating. While it is extremely gentle in operation its very long “attack” time may make it unsuitable for certain applications.

2. SST Out*, AGC In, Compressor Out

The AGC is operating and generally controlling the medium-term dynamics of the audio passing through this section.

3. SST In, AGC In, Compressor Out

The SST and AGC are operating and generally controlling the medium- and long-term dynamics of the audio passing through this section.

4. SST Out*, AGC Out, Compressor In:

Signal processing by Compressor only with relatively fast time constants. The sound of the processing will be obvious with most program material.

5. SST In, AGC Out, Compressor In:

Signal processing is by the SST and Compressor only, operating with relatively slow and fast time constants respectively.

6. SST Out*, AGC In, Compressor In:

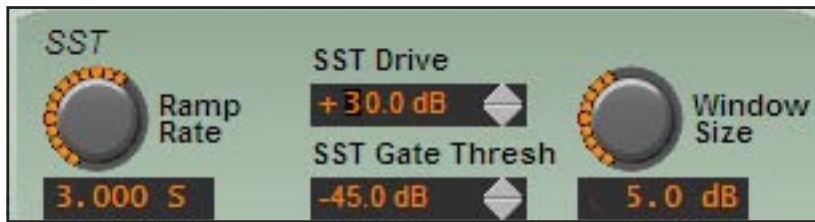
Signal processing by the AGC and Compressor only, with no benefit from the SST which operates on very long variations in program level.

7. SST In, AGC In, Compressor In:

This is the normal (and preferred!) operating mode of the AGC/Compressor section of the FM-2000. In this mode audio is conditioned by the SST to correct long-term level variations, the AGC corrects medium-term level variations, and the Compressor operates to control medium-short-term level inconsistencies.

* When SST is ‘out’ a static gain of -12dB is set by algorithm.

SST Controls Overview



There are primarily four controls associated with the SST algorithm:

SST Drive

The SST Drive control adjusts the amount of signal fed to the algorithm that measures the dynamic behavior of the five band AGC. The higher the SST Drive setting the deeper the amount of automatic gain reduction in the algorithm – gain reduction in the SST is necessary in order to allow it to operate over the widest possible program dynamic range.

A typical amount of SST gain reduction that affords the best tradeoffs in operation will be somewhere between -12 and -15dB. Note that there is no harm in generating more or less indicated gain reduction as long as one understands that more gain reduction affords greater long term control of audio levels and less gain reduction affords less. The overall control range is +30.0dB to +40.0dB.

SST Gate Thresh

The SST algorithm has been provided its own Gate Threshold so that its operation may be optimized when it is combined with the five band AGC. Typically the SST Gate Threshold will be set several dB higher (less negative) than that of the five band AGC. The overall control range is -70.0dB to -6.0dB.

The SST Gate Threshold control sets the audio *input* level where SST correction is turned off and its gain is frozen.

Ramp Rate

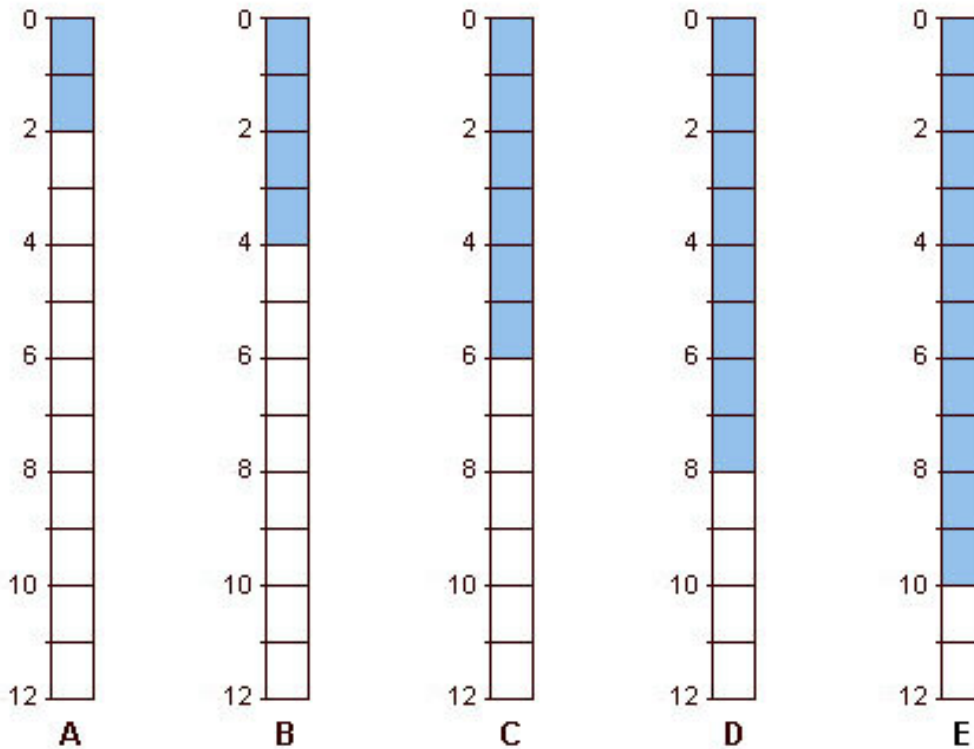
In order for the SST to behave with all types of program requirements it must have its speed of correction tailored for the application, and this is what the Ramp Rate control does. The control is calibrated in units of time, and therefore describes the rate at which the SST causes more or less activity to occur within the five band AGC/Compressor.

A good typical starting value for SST Ramp Rate is 3.000 seconds. The full control range is 500.0mS to 7.000 seconds.

Window Size

Because the SST is responsible for maintaining the operation of the five band AGC in its “Sweet Spot” the size of that sweet spot needs to be able to be set for different kinds of program requirements. See the graphic below for some examples:

Window Size Control Setting Relationships

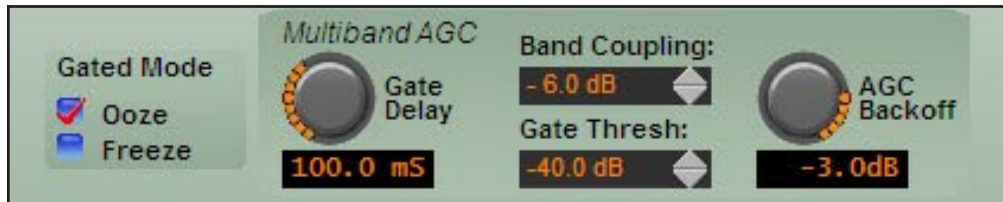


The size of the shaded areas in the above graphic serve to demonstrate the size of the error (in dB) permitted to occur within the five band AGC before the SST corrects it. It is important to note that the SST has no control over allowing the five band error to go positive — that is specifically handled by the five band AGC itself. As the shaded areas show, the larger the Window Size, the larger the “slop” allowed by the SST before correction takes place.

Very small Window Size settings can cause the SST to continuously try to correct for minute behavioral errors within the five band AGC. Likewise, very large Window Size settings will allow large AGC errors to go uncorrected. Settings at either extreme, while probably not doing anything wrong to the audio, are not particularly useful.

The usual setting of the Window Size control for the majority of program content will be between 4.5dB and 7.5dB. As indicated by the graphic, the full range of the control is 0dB to 12dB.

AGC Controls Overview



Gated Modes

The “Gated Mode” controls determine how the AGC behaves in the absence of audio.

Ooze

When “Ooze” is selected and the audio input falls below the AGC Gate Threshold, rather than the gains of the bands “freezing” they will slowly return towards 0dB. This setting may be useful for formats that contain material with very wide dynamic range or for certain speech-based formats where the “Gate Delay” setting may not be appropriate.

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

Freeze

When “Freeze” is selected and the audio falls below the AGC Gate Threshold value the gains of the five bands will “freeze” at their current values and will hold at those values until audio is present that is above the AGC Gate Threshold. This prevents the increasing of gain in the absence of audio which would likely increase background noise.

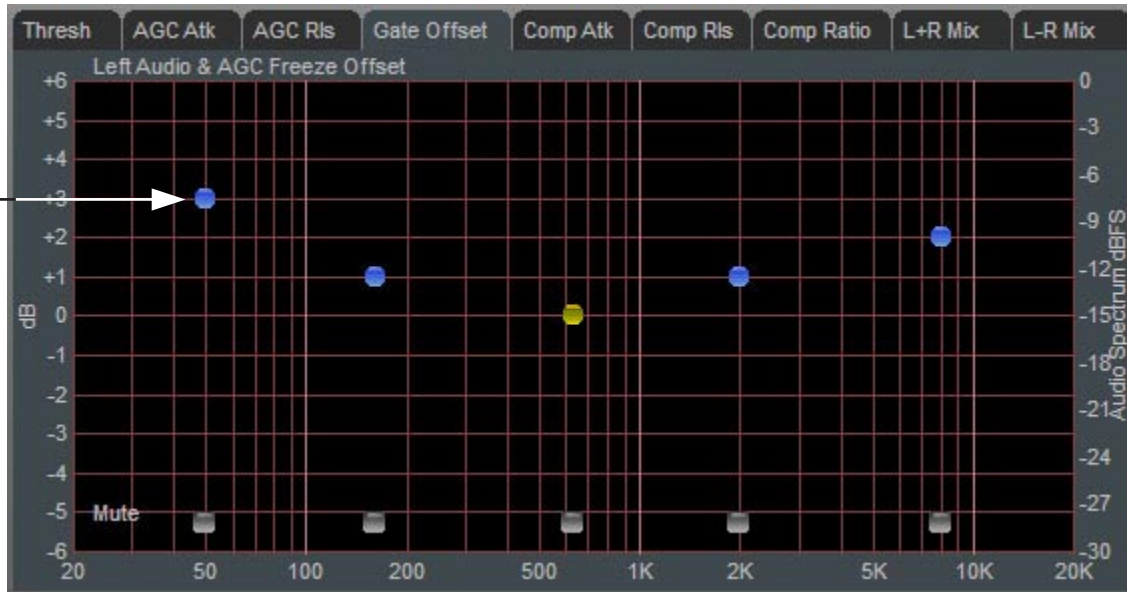
Note that the time between reaching the AGC Gate Threshold and the actual gain Freeze is determined by the setting of the Gate Delay control.

AGC Gate Threshold

The AGC Gate Threshold control sets the audio level below which the audio gain reduction values are held until audio returns. This control can be set to OFF or adjusted over the range of -79.0dB to -20.0dB.

In each of the five bands there is also a Gate Threshold Trim control allowing offsets of +/-6dB to be made to each band referenced to the main Gate Threshold setting. This allows precise Gate thresholds on a band by band basis as appropriate for the program format requirements.

Example: Suppose the AGC Gate Threshold control is set to -40.0dB and the controls for the individual bands are set as in the example below:



Five Band AGC Gate Offset Controls

In this example, Band 1's Gate Threshold has been offset by +3db, which means its actual gate threshold is now -37dB, or 3dB less sensitive than band 3 (which is still set at the "0" reference line). This will cause band 1 to need a higher audio level before it will "un-gate" and allow the gain to increase. Setting the gate this way prevents unnatural increases in low frequency gain that might be undesirable. The Gate Thresholds may be offset this way in any of the bands as the application warrants.

Gate Delay

The Gate Delay control sets how long the gating circuit waits after the signal has fallen below the Gate Threshold before it modifies the release time according to the "Gated Mode" setting. The Gate Delay is adjustable from 50 milliseconds to 500 milliseconds, with values on the order of 50 milliseconds being useful for typical programming environments.

When Gate Delay is adjusted for longer periods of time beyond about 100 milliseconds, partial release of all bands is afforded when the input signal falls below the actual Gate threshold. This can be useful for maintaining a more natural drop in signal level during program fades.

The controls in the Gate section are quite subjective and therefore should be adjusted entirely by the requirements of the program format. There is no right or wrong setting of the controls unless those settings cause undesired, unexpected, or unwanted behavior of the processor during low level program passages.

AGC Backoff

The AGC Backoff adjusts the processing *balance* between slower AGC action and faster compression.

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

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

The setting of the Backoff control allows the audio compression texture to be modified from a gentle, almost unobtrusive control (Backoff between “0” and about minus 2.0dB), to very obvious compression (Backoff control at “-3.0dB” or greater negative numbers) creating a sound that is similar to vintage compressors and limiters from the 1960’s.

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

Band Coupling

The Coupling control has one specific purpose – to keep the gains of bands 1, 2, 4, and 5 from exceeding the gain of band 3 when those “outer” bands are “ungated.” This prevents unnatural gain rush-up by those bands when such behavior would be undesirable. Without the “Coupling” feature the gains of the bands at the frequency extremes can increase their gain and cause a spectrally incorrect sonic mixture at the output of the five band section.

Band 3 was chosen as the “master” band for the Coupling control because that band is in the approximate center of the audio spectrum.

The Coupling control may be adjusted over the range of “0dB” to “-30dB.”

At a setting of “0dB” bands 1, 2, 4, and 5 can *never* have more gain than band 3. Such a setting is useful in formats such as Classical and Jazz that typically would not wish the spectral “correction” caused by the outer bands increasing their gain but still need those bands to reduce their gains when appropriate.

The setting of “-30dB” essentially turns the Coupling mechanism off — the outer bands are then allowed to have more gain than band 3 - up to 30dB more gain (which is a lot!).

In typical programming situations the best setting of the Coupling control will likely be somewhere between -3.0dB and -8.0dB.





If grander settings of the Coupling control (greater than about -10.0dB) are necessary in order for the stations’ programming to sound more consistent on the air it might be advisable to investigate the cause and correct it, because this implies either poor frequency response in one or more parts of the signal chain or grossly poor quality program material coming from some source.

Crossover Frequencies



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

There are four crossover frequencies to select.

-  The Super Low to Low band crossover frequency is selected using the red list box. Frequencies between 40.5Hz and 120Hz may be selected.
-  The Low to Low Mid crossover frequency is selected using the yellow list box. Frequencies between 149Hz and 334Hz may be selected.
-  The Low Mid to High Mid crossover frequency is selected using the green list box. Frequencies between 817Hz and 1.59kHz may be selected.
-  The High Mid to High band crossover frequency is selected using the blue list box. Frequencies between 2.52kHz and 7.55kHz may be selected.

Selecting Crossover Frequencies

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

The upper and lower range of crossover frequency settings is calibrated so that they are most useful with typical program material. Note that the crossovers that affect Band 2 and Band 4 have been made variable in such a way as to allow pseudo-three band operation or operation of those bands with extremely narrow bandwidths which may be useful for some programming requirements.

The table below serves as a starting point for most formats:

Program Format	SL to L	L to LM	LM to HM	HM to H
Classical/Jazz	120 Hz	250 Hz	1.0 kHz	3.00 kHz
Country/AC	102 Hz	204 Hz	817 Hz	4.49 kHz
Dance/Hip Hop	80 Hz	180 Hz	1.16 kHz	5.50 kHz
Modern Rock	93.6 Hz	216 Hz	1.30 kHz	4.62 kHz
Oldies/CHR	111 Hz	250 Hz	1.37 kHz	4.00 kHz
Classic Rock	83.4 Hz	193 Hz	1.59 kHz	6.17 kHz
Talk	120 HZ	250 Hz	917 Hz	3.00 kHz
Sports	120 HZ	281 Hz	817 Hz	2.52 kHz

SL = **Super Low Band**

L = **Low Band**

LM = **Low Mid Band**

HM = **High Mid Band**

H = **High Band**

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

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

For Rock music a slightly lower Super Low to Low crossover of 80Hz can put some extra “gut” into the feel of the music - likewise with Dance or music of other formats that contain punchy, deep bass. There is normally not a *significant* amount of fundamental energy below about 60Hz - 80Hz in most program material.

Lowering the Super Low to Low band crossover down to even lower frequencies, perhaps to 70Hz or even 60Hz, can definitely put some “solid bottom” in the audio with certain music formats. This setting can be inappropriate with other types of program material — particularly human voice — so some care and perhaps even experimentation is advised!

If excessive brilliance is an issue the High Mid to High crossover can be reduced slightly down to around 5kHz or perhaps lower.

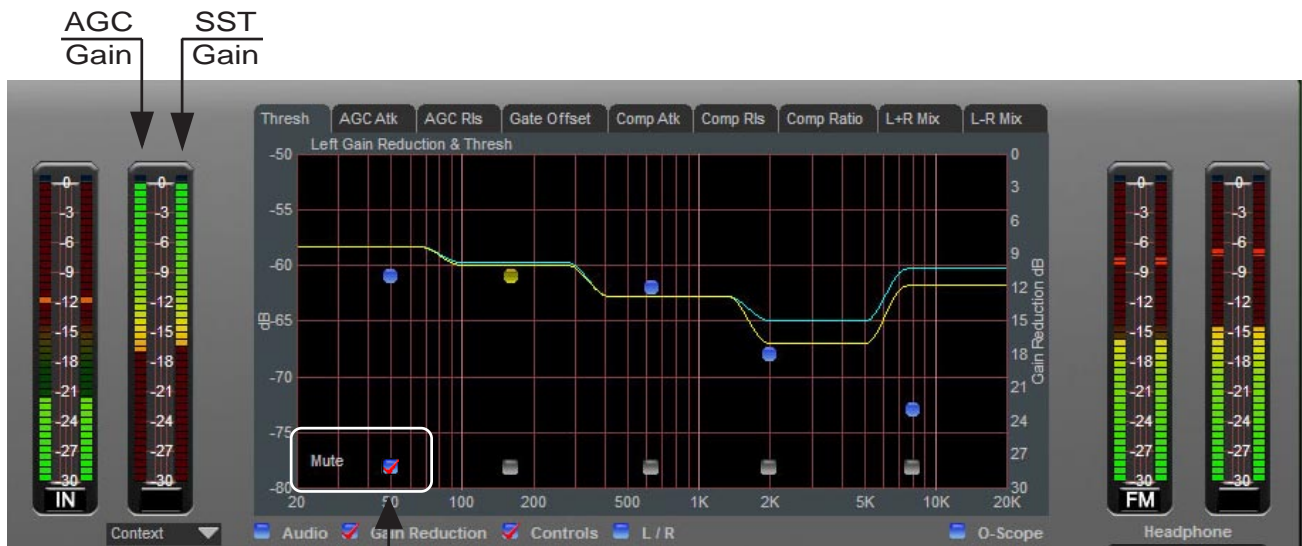
To achieve the opposite effect, that is, to impart a “shimmer” to the highs, one might instead move the High Mid to High crossover up in frequency to around to 7kHz. There is generally no sensible reason to increase the Mid High to High crossover to settings above 7kHz on FM.

For more middle of the road music and for Classical and Jazz the Super Low to Low band crossover may be moved *up* to around 120Hz. The High to High Mid crossover may then be moved *down* slightly to perhaps around 2.7kHz. These settings prevent the high band from taking too much gain and making the audio shrill, while the higher than normal Low to Low Mid crossover prevents too much low frequency extension on music that just doesn’t sound natural with it.

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

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

AGC/Compressor Gain Reduction



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

In the above graphic, the meter pair above the radio button with “Context” listed in it are displaying total gain reduction of the five band AGC in the left meter, and total gain control generated by the SST in the right-hand meter.

Mute - When checked, the band output is turned off. This is useful when you want to hear what one or more bands are doing and allows the dynamic action of bands that are still on to be heard. Note that a band’s mute status is not saved with a preset, meaning that if a band is in mute mode it is always in mute mode until manually removed from that mode.

FM LIMITER MENU



FM Limiter Screen

31-Band Limiter

The FM-2000 contains the Vorsis exclusive 31 band FM final limiter algorithm with 31 bands of limiting in addition to 31 bands of distortion masked clipping.

The multiband clippers are individually distortion masked and can perform up to 20dB of clipping before objectionable artifacts appear. One caveat of 31 bands of clipping is that while *incredible* increases in “electrical” and acoustical “on-air” loudness can be created, the sound can be fatiguing if the clippers are overdriven. Therefore it is probably inappropriate for most program material to drive the clippers *continuously* above 6 to 8dB of indicated clipping.

Instead, we recommend operating the clippers in a manner that allows them to only take care of short transients that the 31-band limiter “intentionally” misses — for best musicality we recommend that the limiters not be operated with extremely short attack and release times. Please open several of our factory presets and observe the settings in the FM multiband limiter sections to see what we mean.

The 31-Band Limiter has been designed to be fully adjustable in order to accommodate a wide variety of program material and competitive needs. It also contains the Vorsis Bass Management System (VBMS) and interoperates with it to intelligently control bass energy. Each of the 31 bands is individually adjustable for threshold, attack time, and release time. In addition, adjustments can be made to the transfer function (knee hardness) for the limiters and the VBMS.

Multiband Enable

This checkbox determines whether or not any processing is applied by the 31-band multiband limiter and clipper. When the 31-band section is not enabled, audio peak control is handled solely by the main clipper, look ahead limiter (if selected), and the composite processor in the FM stereo generator.

Multiband Drive

Determines the amount of input signal applied to the 31-band limiter. Less drive creates a more gentle sound while still controlling peaks, while higher settings make the sound more aggressive through increased short term density. Very high drive settings that result in more than 5-6dB of continuous gain reduction are typically unnecessary with most formats.

The Drive control is calibrated in percent instead of dB because its setting is completely relative to the settings in the five-band AGC section. Normal settings of the Drive control will usually range between 80% and 100%. Full adjustment range is 0% to 100%.

Lim/Clip (Limiter/Clipper) Drive

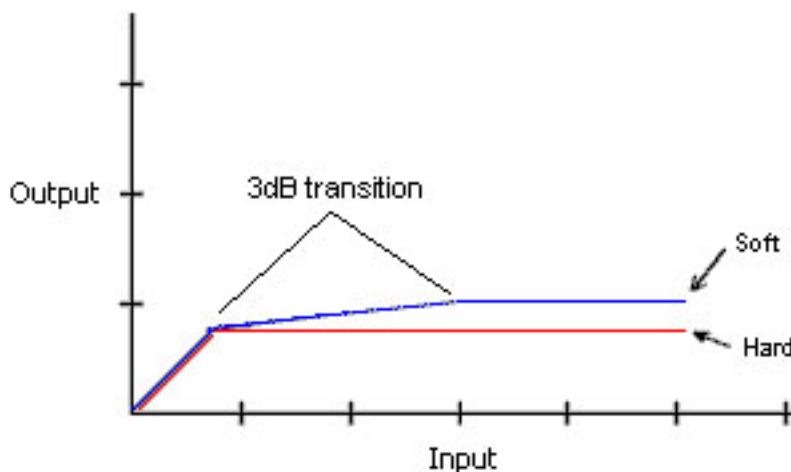
This controls the absolute peak output level of the 31-band section and is adjustable ± 42 dB. The resolution of this control is 0.05dB, fine enough for the most exacting output level adjustments.

Multiband Knee

The multiband knee list box offers two choices for how the multiband limiter's compression knee operates:

Soft – In the Soft mode amplitude limiting begins 3dB below the setting of the threshold controls and the limiter's compression ratio becomes progressively higher as the final limit threshold is approached, where it becomes infinity to one.

Hard – In this mode the limiter's compression ratio is infinite and limiting begins precisely at the threshold setting with absolutely no increase in level above that threshold.

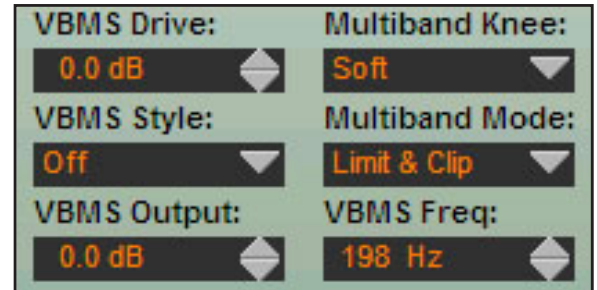


Comparison of Hard and Soft Limiter Modes

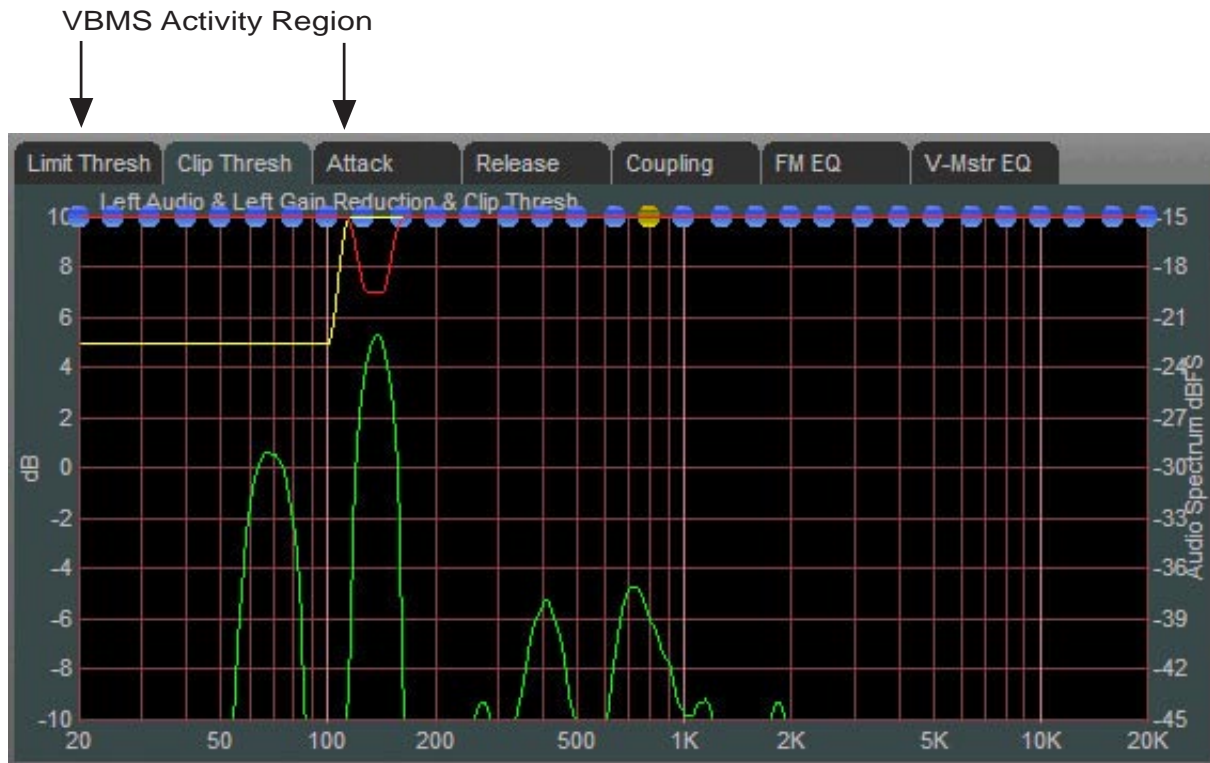
Vorsis Bass Management System (VBMS)

The Vorsis Bass Management System processes extremely low program frequencies in an artistic and natural program - dependent manner without causing dynamic gain changes and undesirable intermodulation artifacts. Its primary purpose is to allow percussive instruments to sound natural while still being well controlled in peak amplitude.

The Vorsis Bass Management system uses special DSP algorithms to control low frequency program energy and allows bass punch to be set for the type of program material. It offers two styles, plus an OFF setting which, when selected, bypasses the VBMS algorithm altogether.



The VBMS offers a visual indication of program induced bass control which can be viewed in the 31-Band limiter graphic as a solid yellow line extending from the VBMS Frequency turnover setting downward to 20Hz, similar to that shown below.



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

VBMS Drive

Adjusts the amount of drive to the VBMS algorithm and is adjustable over a range of +/- 10dB. Higher drive settings cause bass energy within the selected VBMS bandwidth to spend more time above VBMS threshold, which increases the amount of low frequency energy.

VBMS Style

Two VBMS styles plus an OFF setting are available.

OFF - the VBMS is out of the circuit.

Hard - has a transfer function occupying approximately 1dB of transition amplitude. The VBMS action in this mode is moderately abrupt and generates low order harmonics. In a post-filtering process that borrows from the technology of our Timbral mode we mathematically calculate and reduce all bass harmonics except the third and fifth.

Soft - setting operates with a transfer function occupying approximately 1.75dB of transition amplitude. The VBMS action is less sharp than the hard setting, generating even fewer harmonics. When the Soft setting is enabled we mathematically calculate and reduce all energy except the third harmonics.

The two settings may appear to be almost identical because of the small amount of difference in transfer functions. However, because the transition slope is larger on the Soft setting there is a *significant* difference in the way the two clipper styles sound on actual program material.

VBMS Output

Adjusts the output contribution from the VBMS algorithm and may be thought of as a “low bass” mix control.

Generally speaking, the VBMS Drive and VBMS Output may be adjusted to taste. However, care should be exercised in the setting of the VBMS Output control because settings above approximately +3dB may cause the peak amplitude of bass to interfere with the main clipper distortion masking scheme, which may increase distortion on some program material.

Like VBMS Drive, VBMS Output is adjustable over a range of +/-10dB.

VBMS Frequency

The VBMS section’s frequency of operation is adjustable from 60Hz to 300Hz, with program energy not passing through the VBMS algorithm time aligned for phase coincidence regardless of the Frequency setting.

In general, lower frequency settings produce a “heavier” sound, while higher settings result in less coloration while still favorably controlling bass energy excursions. This control should be adjusted to personal taste and program format.

Emphasis Selection

Pre-emphasis is selectable as 50 or 75 microseconds, or Flat (OFF) as shown in the graphic on the right:

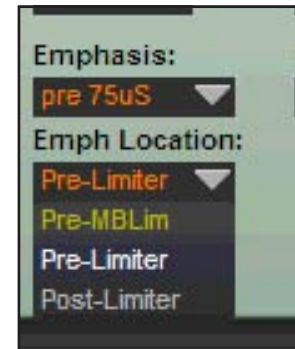
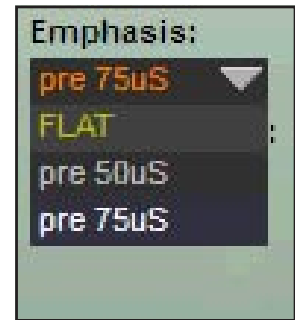
When pre-emphasis is enabled it may be placed in one of three locations in the signal chain:

Pre-MBLim – In this position, pre-emphasis selected by the Emphasis selection box is placed *before* the 31-band limiter/clipper section. In this mode the 31-band section may be operated as a frequency conscious limiter tailored to the pre-emphasis curve if that type of operation is desired.

Pre-Limiter – In this position the pre-emphasis is *after* the 31-band section but *before* the Look Ahead limiter and/or main clipper.

Post-Limiter – In this position the pre-emphasis is *after* the 31-band section and Look Ahead limiter and *before* the main clipper.

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

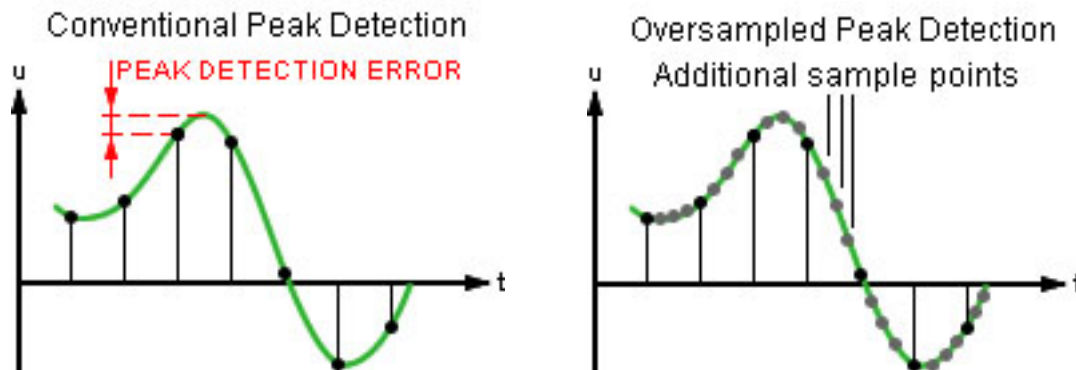


DeEmphasis

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

Look Ahead Limiter

The FM output may be peak controlled by a special high-speed, feed forward limiter with a 0.5 millisecond lookahead time. The control path, and not the audio path, is oversampled in this limiter in order to achieve extremely precise peak control. Because the primary goal of oversampling is better peak control through finer definition of the audio waveform, oversampling of the audio signal itself is not only unnecessary, but wastes DSP resources. The graphic below visually illustrates why control signal oversampling is effective.



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

FM Limiter Controls

The graphic below shows the controls available for the look ahead limiter and the text below it describes each Limiter control according to its function.



FM Look Ahead Limiter

Limiter Enable

When this checkbox is checked the Look Ahead limiter is enabled.

L/R Linked

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

Limiter Thresh

Adjusts the relative threshold of the look ahead limiter compared to the setting of the Lim/Clip Drive control, and is adjusted until the onset of limiting is reached with normal program material as indicated by the FM limiter gain reduction bargraph. The range of this control is +/-6dB.

Attack

Adjusts the attack time of the look ahead limiter. Note that attack time settings from 0.2 milliseconds through 0.5 milliseconds allow the limiter to “look ahead” in time and react to signal peaks *before* they arrive at the limiter’s gain control algorithm.

Attack time settings greater than 0.5 milliseconds, and up to the limit of 100 milliseconds, cause the limiter to operate conventionally. For example, if the attack time control is set to 1.0 milliseconds, the actual attack time is 0.5 milliseconds because the 0.5 millisecond look ahead is *always* in circuit and its value must be subtracted from any adjustment of 0.5 milliseconds or greater in order to know the actual attack time.

Release

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

Slower settings (longer release times) cause the peak limiter to recover more slowly from limiting; release times set too slow may allow undesirable signal ducking to occur.

Conversely, fast release times increase signal density, and release times that are too fast result in increased intermodulation distortion which can “muddy” the sound.

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

Delayed Release

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

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

Digital Output (dBFS)

Adjusts the precise peak audio signal level present at the FM Limiter of the FM-2000. The resulting peak output levels in dBFS are indicated by the front panel and remote GUI bargraph meters. This control can be set to OFF or adjusted over the range of -79.95dB to +12.00dB.

Clip Style (Main Clipper)

Three final clipper styles are offered in the Main Clipper to enable the end user to create the on-air sound desired. All settings offer excellent peak control because the main clipper is followed by an overshoot calculator that removes peaks that, due to their frequency and energy content, would cause the main 15kHz lowpass filter to overshoot. The Main Clipper drive level is determined primarily by the setting of the Lim/Clip (Limiter/Clipper) Drive control as well as the amount of activity occurring in the 31-band section.

Off – defeats all clipping in the main FM limiter, noting that the overshoot calculator is always active and may act like a non-band limited/non-distortion cancelled “clipper” if it is overdriven with test tones or program material.

Hard – the Hard clipper style has an infinite ratio with no increase in peak levels once its threshold is achieved. This is the least “forgiving” of the clip styles as far as clipper drive levels are concerned and adds brightness and density to the program.

This clip style is the “cleanest” of the three styles because unlike the other two (see below), below the clip threshold no clipping occurs. The other reason this is a “cleaner” clipper is because the distortion spectrum created by this style is more easily manipulated by the distortion cancellation scheme.

The Hard style creates predominately harmonic distortion up to approximately 0.5dB of drive, depending on the incoming program material.

Firm – The Firm clipper style has a transfer function that allows progressively higher ratios over approximately a 0.5dB transition region. It is a softer sounding clipper than the Hard style and is also more forgiving in drive level but can sound grainy on some material because the distortion spectrum is richer and not as easily managed by the distortion cancellation scheme.

The Firm style creates both harmonic and intermodulation distortion with the balance leaning towards harmonic rather than intermodulation.



Round – The Round style has a transfer function that allows progressively higher ratios over approximately a 1.05dB transition region. This is most forgiving of the three clipper styles as far as drive levels are concerned and contributes “roundness” to the audio without sounding overly harsh. However, it is the least “distortion cancelled” of the three styles simply because of the richer harmonic spectrum created by its action.

The Round style creates both harmonic and intermodulation distortion with the balance leaning more towards intermodulation. This is the warmest and “thickest” sounding of the three clipper styles.

FM Diversity Delay

The FM-2000 contains an internal digital delay to compensate for the corresponding delay in the HD Radio codec. The delay section is capable of providing up to ten seconds* of compensating delay in steps of 100 microseconds.

* Future software will enable Diversity Delay settings beyond ten seconds.



The **Coarse** delay is set first to align the analog and HD signals as closely as possible, and then the Fine delay control is used to trim the delay to precisely compensate for that in the HD channel. Generally it is a good idea to set the Fine delay to midrange before adjusting Coarse delay. Resolution of the Coarse delay is 50 milliseconds.

The **Fine** control covers a range of 50 milliseconds in 0.1 millisecond (100 microsecond) steps. If the Fine control does not set the increment exactly to where it is desired, double clicking on the delay value text box will open it for manual entry of the delay time.

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

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

Entering Delay Values

You can double click on the delay value window to directly enter a numeric delay value in *milliseconds*. This is useful when you know roughly how much delay is required or wish to try various values quickly. You may also hold down the *Shift* key on your keyboard while adjusting a delay slider to effect faster adjustment rates, or hold down the *Ctrl* key to achieve finer control.

Special 31-Band Operating Settings — Wide and Timbral

The 31-Band FM limiter section operation varies according to the settings of the “Wide” and “Timbral” checkboxes, offering four different operating modes.



- If the “Wide” and “Timbral” checkboxes are both unchecked the 31-band limiters operate with all bands completely independent.
- If only “Wide” is selected, the 31-band section operates as a pseudo-10-band limiter. That is, ten “center bands” across the spectrum are selected and the immediate neighbors above and below are ganged to it.
- If only “Timbral” is selected, the 31-band section operates in a “mathematically coupled” mode so that gain reduction occurring at a signal’s fundamental frequency also causes gain reduction to occur at its 2nd and 3rd harmonics. This operating mode is extremely complex in mathematical terms, and has a more natural sound for music formats such as Jazz and Classical because the gain of upper harmonics of solo instruments stays more favorably balanced with gain reduction occurring at the fundamental frequencies.
- If both “Wide” and “Timbral” are selected the 31-band section operates as a pseudo-ten band limiter with the Timbral algorithm running on top of it.

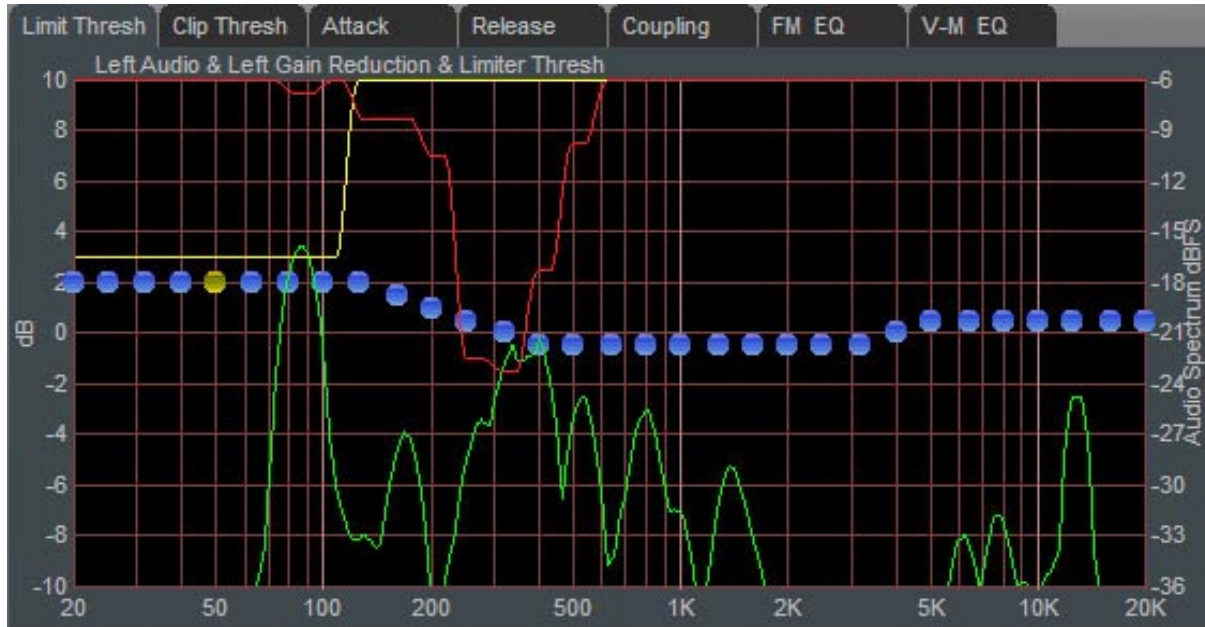


We can think of no really good reason to operate any of the bands in the uncoupled mode (see Coupling Controls Screen section). Nor can we think of any reason why someone would want to operate the 31-band section in the Wide *and* Timbral modes simultaneously. But these options are readily available to us in the DSP domain and we felt that it should be up to the end users to decide how they wish to utilize the coupling capabilities. See the 31-band APP note (AN2008-05) on the Vorsis website.

31-Band Limiter Dynamics Screens

Thresholds Screen

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



31-Band Limiter Threshold Screen

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

Each of the limiter thresholds can be adjusted separately or in groups according to the manner described in the earlier FM-2000 Graphical Interface Operation section. Lower numbers reduce the threshold, causing limiting to occur at lower audio levels. Conversely, higher thresholds cause limiting to occur at higher audio levels, resulting in higher output levels in that band.

Higher limiting thresholds cause more activity in the adjoining 31-band clipper. This is because the clipper section follows the limiter section and, as the limiter thresholds are raised, more audio escapes the limiter section “uncontrolled” to be routed next to the 31-band clippers.

Because of the frequency response contouring in FM pre-emphasis it is generally customary to adjust the limiter thresholds in a sloping fashion according to the limiting curve required. In this manner a frequency conscious limiter section can be created with any sensitivity curve desired. While radically different settings between adjacent bands won't cause strange sounds, operationally it doesn't make much sense.

Multiband Mode

This drop down list allows the 31-Band FM limiter section to perform as a Limiter only, a Clipper only, or as both a Limiter and a Clipper.

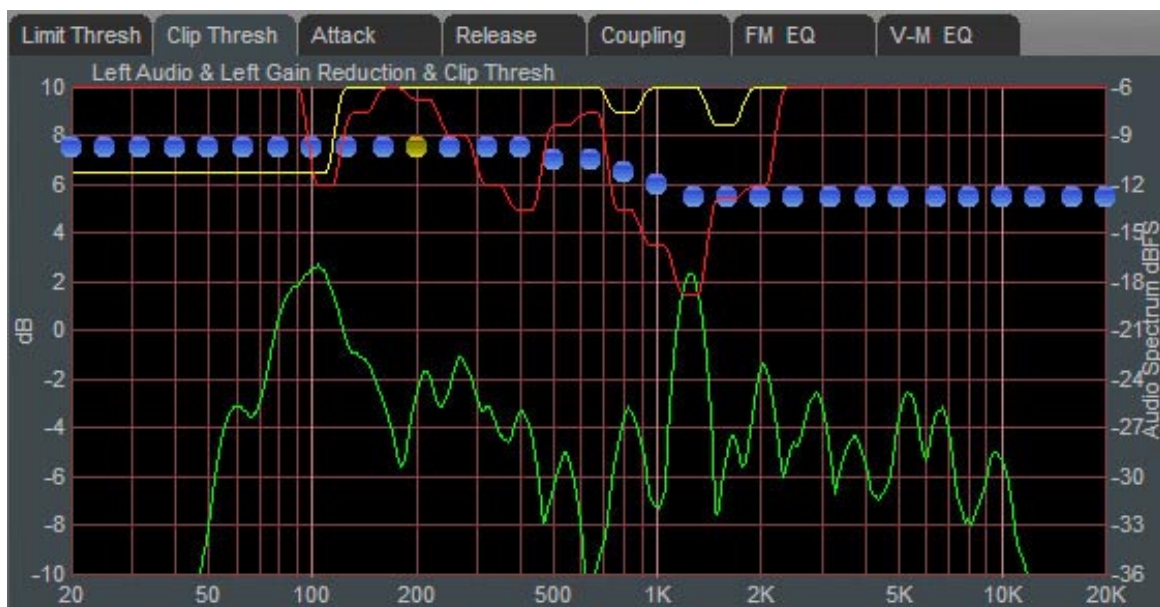
Limit & Clip – In this mode both the 31-band limiters and 31-band distortion masked clippers are active. The clippers are fed by the output of the limiters in this mode and therefore the clippers remove only limiter overshoots. Denser clipping will be seen in the clipper section if:

- the limiters are operating with excessively high thresholds;
- the limiters are operating with relatively long attack times;
- the clippers are operating with excessively low thresholds.

In the factory presets where both limiters and clippers are active the limiter parameters have been adjusted so that the clippers clip only on short duration peaks — ones that are intentionally missed by the 31-band limiters due to their threshold and/or attack time settings. This setting results in a more dynamic sound while still retaining excellent peak control.

Limit Only – In this mode the 31-band clippers are defeated and the 31-band limiters act alone. This mode may be useful for certain formats where the additional short-term density added by the multiband clippers is not desired. It is important to remember that the 31-band limiter section may be operated in the optional Wide and Timbral modes regardless of whether the 31-band clippers are enabled or not (see page 3-48).

Clip Only – In this mode the 31-band limiters and their associated Wide and Timbral modes are disabled, with all peak limiting in the 31-band section accomplished by the 31 distortion-masked multiband clippers.



Clipper Threshold Adjustment Screen

In addition to active distortion masking within each of the 31 clipper bands, each band has its own individually adjustable clip threshold, allowing clipping of program material to either control peaks or generate competitive loudness. The thresholds may be precisely tuned to taste, format, and competitive needs.

As we mentioned previously, although each band has its own distortion masking, continuous deep clipping within one or more bands (in excess of 6dB to 10dB) should be approached with caution. This is because, although the distortion is masked, extremely high levels of RMS loudness will be produced which could become fatiguing with certain types of music, depending on how other processor parameters prior to the 31-band section have been adjusted.

Attack Time Screen

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



Attack Time Adjustment Screen

The attack time of a limiter is a measure of how fast it responds to an audio peak that is above threshold, or too high. Shorter attack times allow more accurate peak control, but excessively short attack times can literally suck the life out of musical material. On the other hand, excessively long attack times will allow audio peaks to escape the limiters largely uncontrolled, leaving them to be controlled solely by the final look ahead output limiter (if enabled) or the final clipper.

The attack times for each band are adjustable from 0.1 milliseconds (100 microseconds) to 1000 milliseconds (1 second). Faster attack times exert more control over waveform excursions, while slower attack times allow peaks to escape the limiter section with less control. Attack times should be adjusted to taste, keeping in mind that the slower the attack time the more work the final clipper or limiter must do, and the faster the attack time the fewer short term dynamics the program material will have.

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

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

Release Time Screen

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



Release Time Adjustment Screen

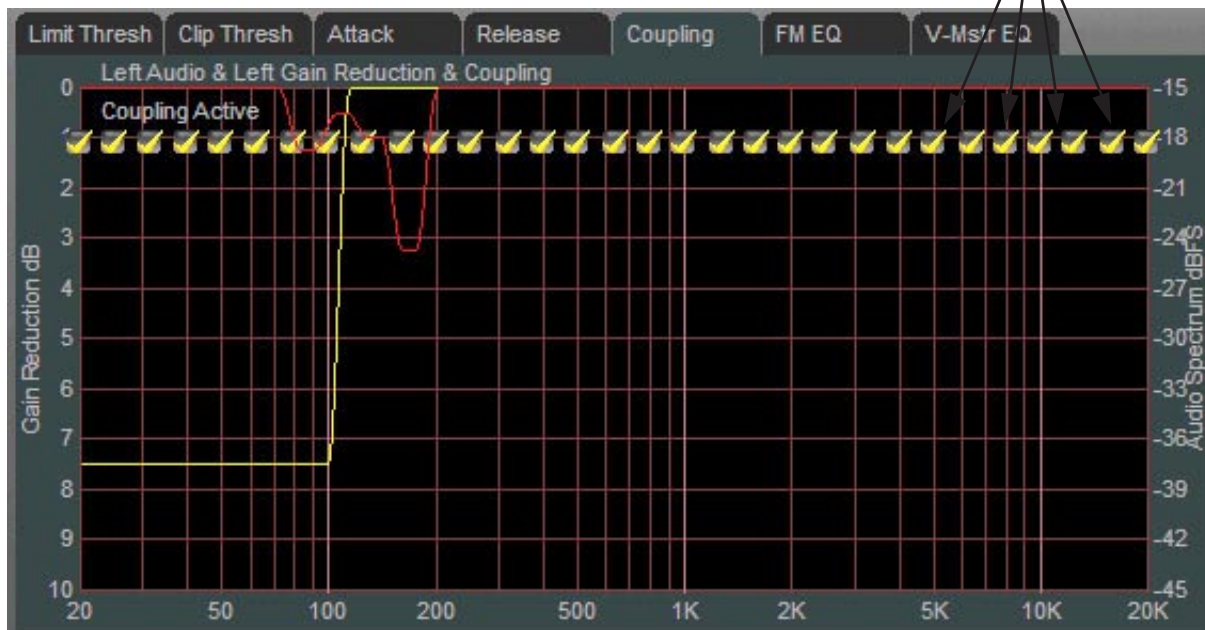
The release time of a limiter determines how fast it recovers from an audio peak that caused a momentary decrease in gain. Shorter release times create a more dense sound, and excessively short release times can also increase intermodulation distortion (the mixing of low and high frequency energy — or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

The release times for each band are adjustable from 30 milliseconds to 1000 milliseconds (1 second). Faster release allows a more rapid recovery after a limiting event while slower release times do the opposite. Release times should be adjusted to taste, keeping in mind that the slower the release time the more dynamic the sound but the more likely that audible “ducking” may be heard.

Coupling Controls Screen

The “Coupling” controls choose which bands should be “uncoupled” from the Wide or Timbral algorithm when it is running. This mode is adjusted primarily as a matter of personal taste and the usual mode of operation is to have all 31-Band Coupling selections “checked”.

Checkboxes for “Coupling”
(one for each band)



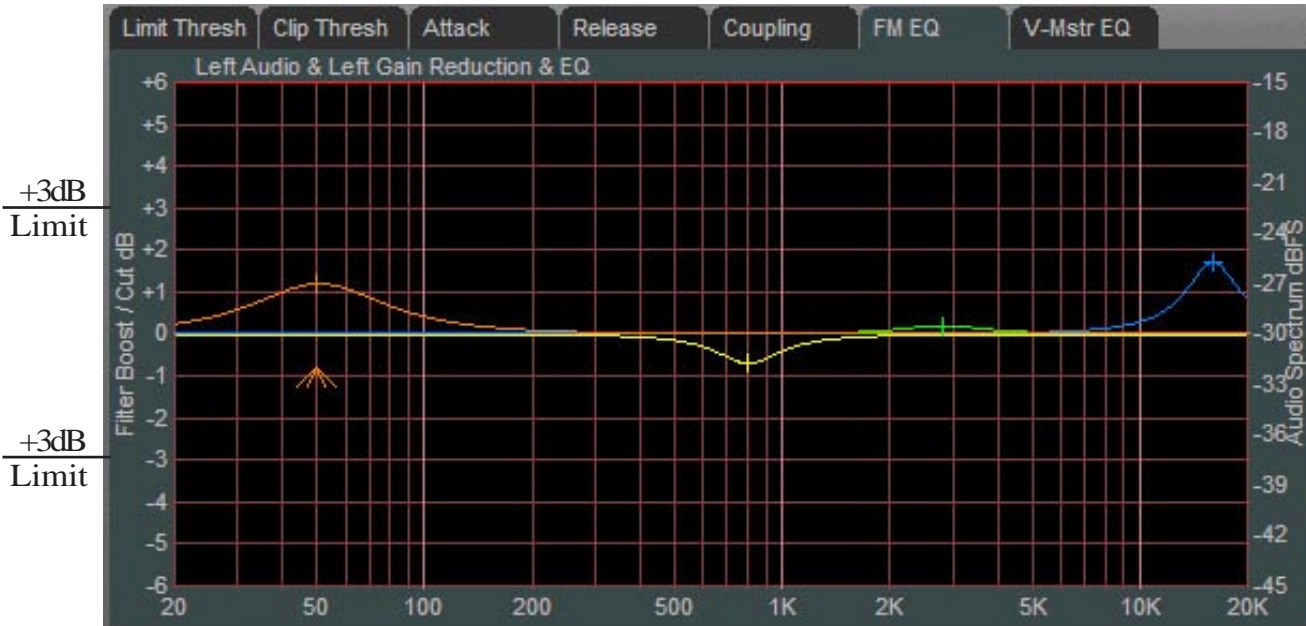
31-Band Limiter Coupling Screen - FM Only



As previously stated, we can think of no really good reason to operate any of the bands in the uncoupled mode. But this option is readily available to us in the DSP domain and we felt that it should be up to the end users to decide how they wish to utilize the coupling capabilities.

Final Equalization Screen

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



Final Equalization Screen

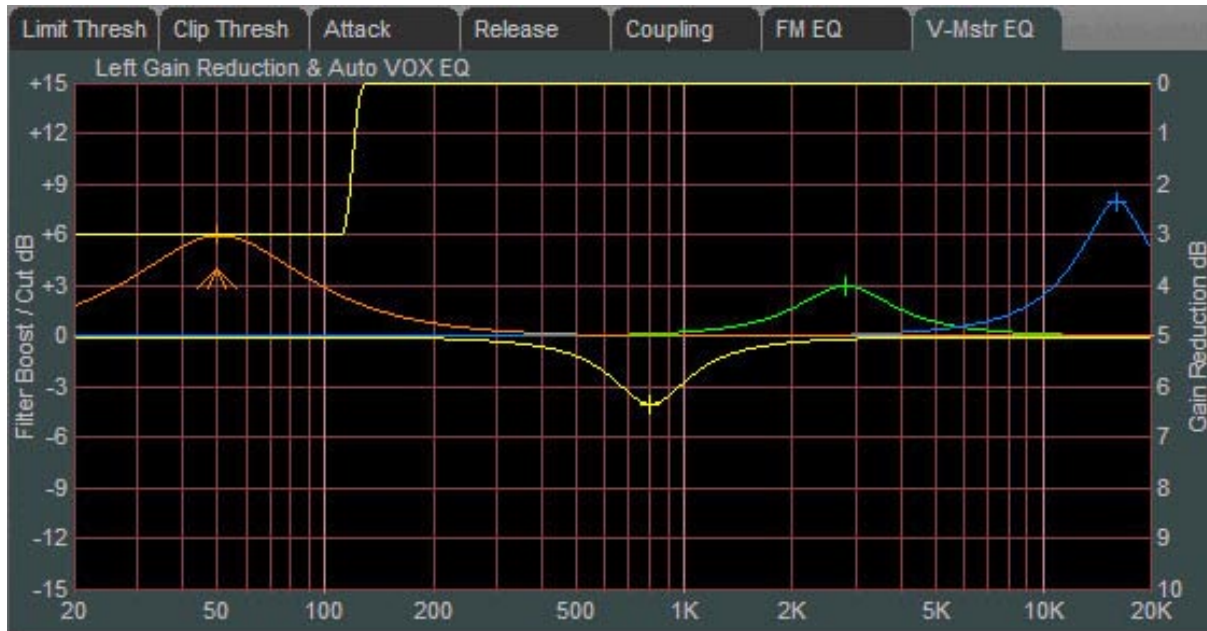
Under certain conditions it may be desired to *slightly* alter the sound exiting the final 31-band limiter. A *small* amount of equalization performed after limiting can “open” up the sound or provide a *slight* enhancement to its overall feel, and minor equalization tweaks can be accomplished after the limiters without radically disturbing the peak levels emerging from the 31-band algorithm.

Only very small amounts of boost equalization are recommended once final limiting has been accomplished in the 31-band section. In fact, the adjustment range has been limited to $\pm 3\text{dB}$ in order to curtail the temptation to perform aggressive equalization where it is the most inappropriate.

Small adjustments in EQ boost — on the order of 1dB or 2dB — are sufficient to radically alter the overall tonality of the 31-band section. Equalization boosts must be approached with caution because they can add significant energy that must be controlled later by either the look ahead limiter or final clippers. On the other hand, equalization cuts can be made at will because they do not *add* energy.

VoiceMaster Equalization

Below is a closer look at the equalization section of the graphical display.



VoiceMaster Equalization Screen

The VoiceMaster equalization controls may be adjusted just as the other parametric equalizers are adjusted — using the mouse to craft the desired equalization curve shape.



Double-clicking an EQ will select it, or you can right click on the graphical area and a four-choice menu (shown above) will appear, allowing you to choose which of the four parametric equalizer sections you'd like to adjust. All four sections of the VoiceMaster equalizer are identical and each may be adjusted/moved to any part of the 20Hz to 20kHz frequency range (including overlapping them!).

A Note Regarding VoiceMaster Equalization

The equalization for the VoiceMaster signal path is located just prior to the FM limiter/clipper. What this means is that some *care* is required when equalization boost is used.

Equalization boost of more than 3dB can be troublesome, although equalization cuts are *always* safe to make because they do not add to the pre-clipper headroom requirements.

Because the VoiceMaster's "Peak Ceiling" control is located after equalization, it can be used to tame the following clipper's behavior with live voice if more than a dB or so of VoiceMaster equalization has been dialed in.

STEREO ENCODER MENU



Stereo Encoder Screen - FM Only

The stereo encoder page contains the controls required to configure the FM-2000's internal DSP-based precision stereo generator.

Stereo Width Limiter

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

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

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

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

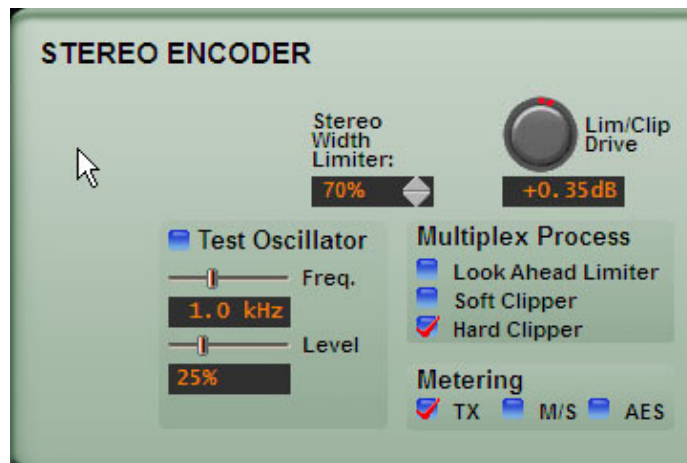
In order to make stereo enhancement “play nice” with the majority of stereo receivers in real-world listening environments with all types of program material, it is preferable to have some sort of controlling mechanism in place to “manage” the amount of L-R energy present in the transmitted signal as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

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

When in the Off position the Stereo Width Limiter is completely out of the circuit. When the control is at the 100% position the L-R may be permitted to achieve 100% of the instantaneous L+R audio level. At lesser percentages the amount of L-R energy is constrained to that percentage of the L+R level at that instant. A good starting point for this control is 70% which constrains the L-R to about 3dB below the L+R.

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

This is in contrast to a control in another audio processor that has “Stereo,” “-3dB,” “-6dB,” and “Mono” settings within its stereo generator control menu. In that product, setting the control to -3dB causes a static 3dB reduction in stereo separation regardless of the incoming program material! The Vorsis approach is more intelligent in how it works; it does not *decrease* stereo separation on program material that does not *need* it.



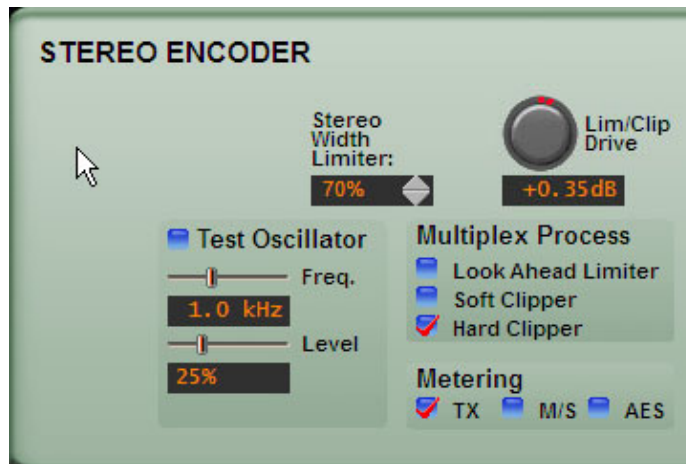
Lim/Clip (Limiter/Clipper) Drive

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

Test Oscillator

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

We’ve also included a special frequency, “Bessel,” which is a 31.250kHz test signal that can be used when the first Bessel null is being utilized for modulation monitor calibration.



Multiplex Process

The FM-2000 provides three separate ways to process the composite stereo waveform:

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

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

Because the Soft clipper has “gain” due to its 3dB transfer function, setting the Drive control to -3dB puts the Soft clipper right at the threshold of clipping with most program material. Settings higher than -3dB then drive the clipper harder. Note that we do not mathematically remove the higher order odd harmonics in this clipper algorithm like we do in the VBMS section because those harmonics are required in order to make the waveform have a more tightly defined peak amplitude.

Hard Clipper – uses a high ratio transfer function of 0.25dB to precisely control peaks without generating very high order distortion. This clipper has a brighter sound than the look ahead limiter and soft clipper and also creates more high order harmonic artifacts. Note that we do not mathematically remove the higher order odd harmonics in this clipper algorithm like we do in the VBMS section because those harmonics are required in order to make the waveform have a more tightly defined peak amplitude.

Output Metering

The FM meter pair can be switched to indicate operating levels of the TX 1 and TX 2 outputs, the Sum and Difference (M/S) levels, or the peak output levels of the FM AES3 output.

Stereo Encoder

The FM-2000 contains a mathematically perfect stereo encoder whose performance is limited only by the high speed output Digital to Analog converter (DAC) and its reconstruction filter. Stereo separation is typically 60dB between 20Hz and 15kHz.



Pilot

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

SCA 1

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

SCA 2

This control has the same functionality and range as SCA 1 and serves the second SCA input connector on the FM-2000 rear panel.

Analog L/TX 1

Controls the output amplitude of the first (left channel) analog audio output and the output appearing on the BNC connector for TX 1. This control can be set to OFF or to levels from -79.95dB to +6.00dB in 0.05dB steps.

Analog R/TX 2

Controls the output amplitude of the second (right channel) analog audio output and the output appearing on the BNC connector for TX 2. The range is the same as that of the TX 1 control.

Pilot Only

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

Pilot Phase

Corrections to the 19kHz stereo pilot phase may be made in 0.5 degree steps up to ± 22.5 degrees. The ability to vary the stereo pilot phase compared to that of the 38kHz subcarrier allows the FM-2000 to compensate for nonlinear time delays in the transmission system that would otherwise disturb the phase relationships between the 19kHz stereo pilot and the 38kHz stereo subcarrier and cause degradation of stereo separation.

Multiplexer

The stereo encoder can be operated in Stereo or Mono by selecting the desired operating mode with this switch.

Mask Filter

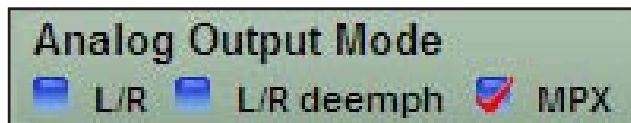
Regardless of which Multiplex Process algorithm is chosen, out of band harmonics above 53kHz are tightly controlled by the Mask Filter which, if desired, can be switched out. The Mask Filter provides an 80dB protection notch for the stereo pilot as well as over 90dB of attenuation above 53kHz to protect the SCA region.



Analog—Stereo Coder Output Options

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

L/R – The male XLR outputs *and* the female BNC connectors assigned to TX 1 and TX 2 carry line level audio. In the case of the XLR outputs the audio is electronically balanced. If pre-emphasis has been selected in the FM Limiter screen then these outputs are pre-emphasized also.

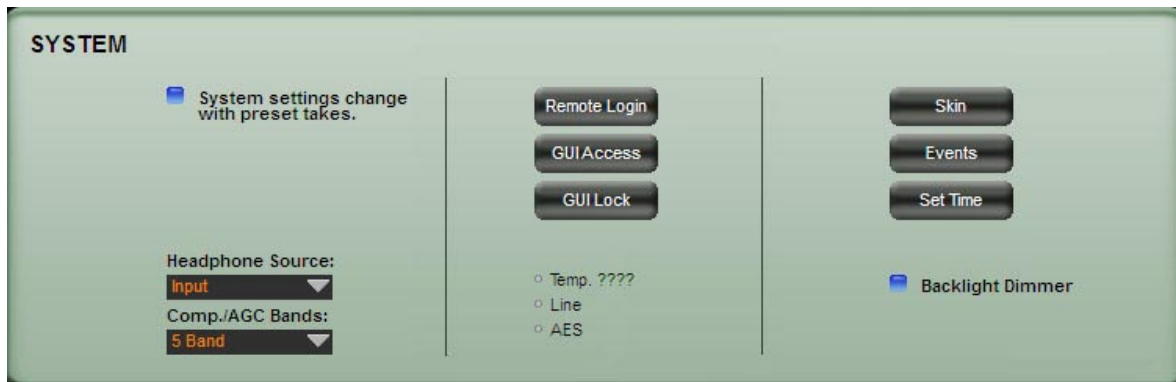


L/R deemph – The male XLR outputs *and* the female BNC connectors assigned to TX 1 and TX 2 carry line level audio but are de-emphasized complementary to any pre-emphasis that has been selected in the FM Limiter screen.

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

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

SYSTEM MENU



System Screen

The system screen allows system-wide settings to be configured. In the first third of the screen is a checkbox and a list box.

System Settings Change With Preset Takes

The following are considered to be FM-2000 “SYSTEM,” or global parameters which are saved with presets but not restored when presets are “taken” unless the “System settings change with preset takes” checkbox in the System menu is checked:

INPUT SCREEN:

Source - Analog or AES
 Analog Gain
 AES Gain
 L/R Balance
 Phase Rotator
 Input Failsafe

FM LIMITER SCREEN:

Emphasis - Off, 50, 75uS choices
 DeEmphasis
 PreDelay
 Digital Output
 Delay Coarse
 Delay Fine

STEREO ENCODER SCREEN:

Metering - TX, M/S, and AES choices

Pilot

Pilot Phase

Mask Filter

SCA 1

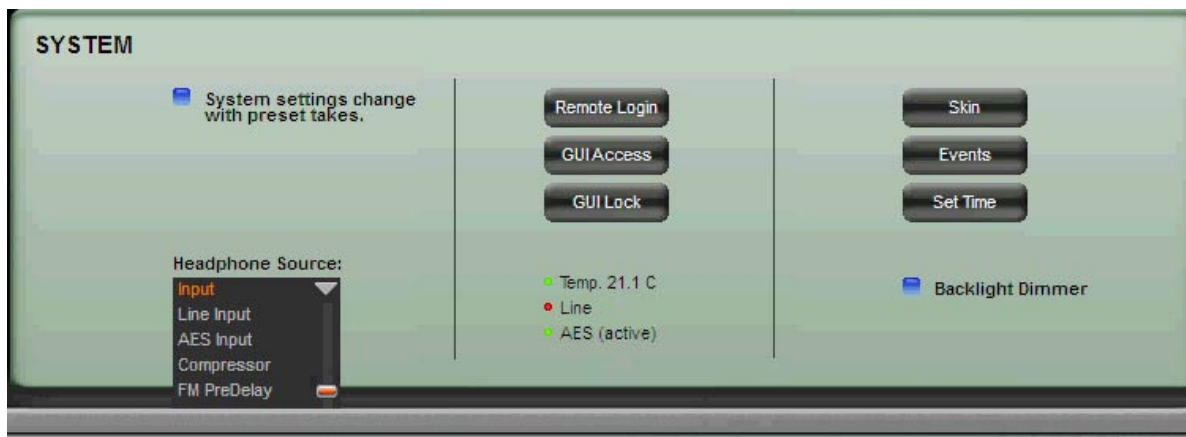
SCA 2

TX 1

TX 2

Pilot Only

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

Headphone Source

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

Input:	Output of the audio input selector, analog or AES.
Line Input:	Audio appearing on the analog line input.
AES Input:	Audio appearing on the AES3 digital input.
Compressor:	Output of the multiband compressor.
FM PreDelay:	Output of the FM limiter before the diversity delay.
FM Limiter:	Output of the FM limiter after the diversity delay.

NOTE: The FM-2000 headphone routing selector allows you to monitor an unselected audio input *without* having to put it on the air. This can be very handy for confirming a source is present without actually switching to it!

Headphone Control

On the right side of the front panel of the FM-2000 is a 1/4 inch female TRS jack for headphone use.

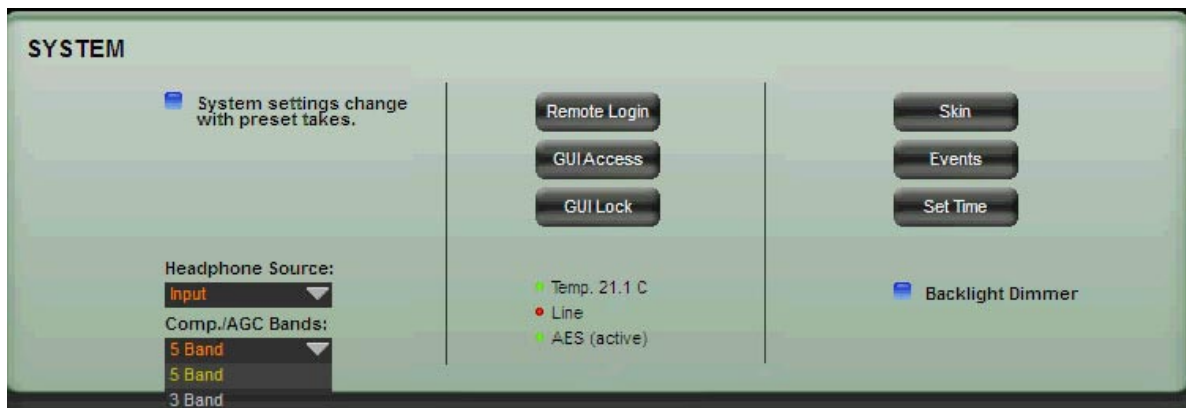
The volume control for the headphone output is located on the far bottom right of both the front panel display and the remote GUI.



Note: The headphone outputs of the FM-2000 are driven by National Semiconductor LM675 power operational amplifiers which are capable of driving very high currents ($>3A$) into any reasonable load. Actual headphone impedance is not a concern; however headphone circuit current limiting is provided at 160mA in order to protect external devices.

Comp./AGC Bands

This list box allows you to choose between using five bands (default) or three bands for the Compressor and AGC.

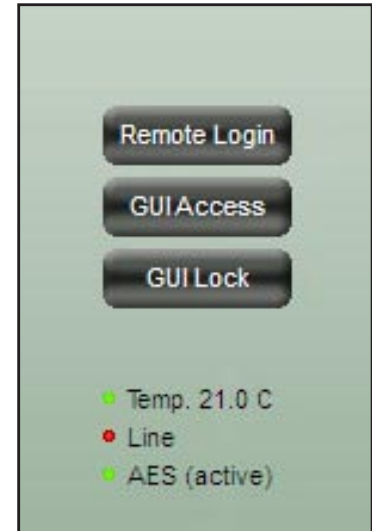
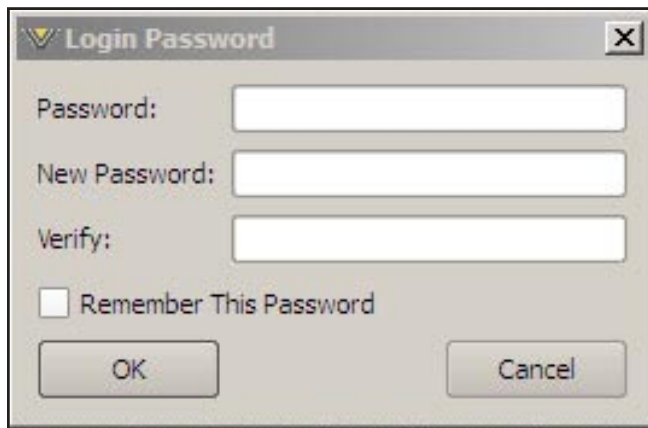


Remote Login

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

Remote login is for password restricting GUI access to an FM-2000 over Ethernet. When a GUI TCP connection session is initiated, the FM-2000 accepts the connection and then asks the remote GUI for a password.

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

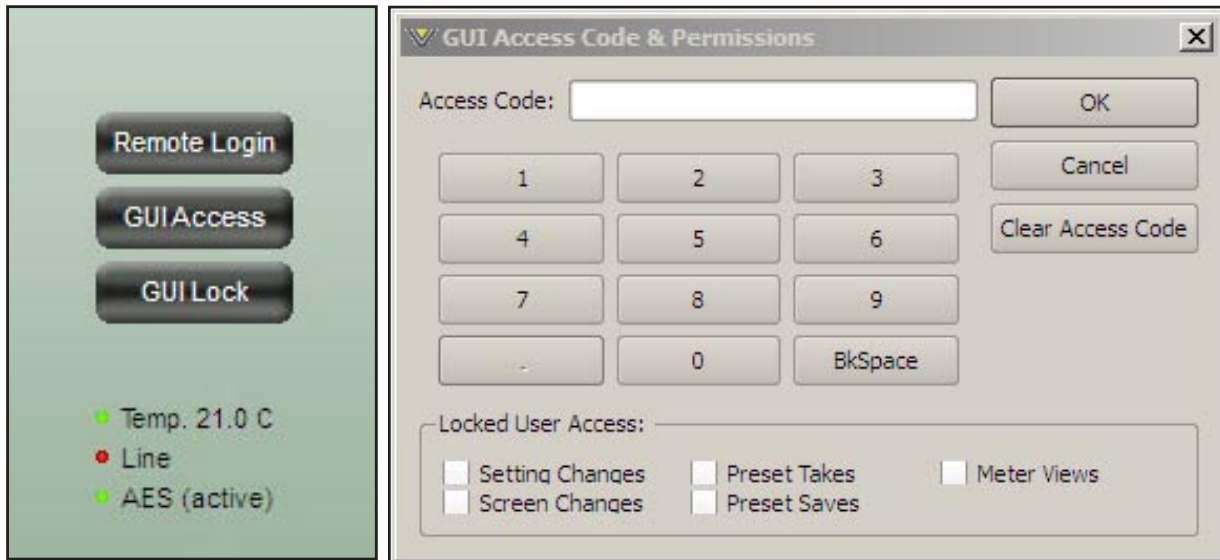


Until the login password can be verified the FM-2000 will not send any of its current settings *to* the remote GUI nor accept any changes *from* the connecting GUI.

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

Occasionally you may see the word “pending” temporarily appear in the connection status box of the remote GUI. That means the FM-2000 and Remote GUI are handshaking on the password string that is being sent.

GUI Access



The system allows you to grant access at certain levels of “permission” in order to control the privileges of those accessing the various features of the FM-2000. Privileges are granted by “access code,” not by user type. In fact, user types are created *through* the generation of access codes for those user types.

Examples:

- Suppose Access Code 1234 is created with “Preset Takes” being the only box checked. This means that a person assigned access code “1234” can *only* take presets.
- Suppose Access Code 5678 is created with “Preset Takes” and “Setting Changes” both checked. This means that a person with access code “5678” cannot only *take* presets, but can also *make* changes to the settings within them. Note however that because “Preset Saves” was *not* checked when this access code was created, that person may *make* changes to a preset but may not *save* them!

GUI Lock

In the event that security has been enabled by configuring the GUI Access Code & Permissions Dialog (above), then selecting GUI Lock will immediately lock the front panel prohibiting access until it has been unlocked with the proper access code. These controls also appear on the FM-2000 local front panel and behave exactly as they do in the remote GUI.

Unlocking the FM-2000 remote or local GUI if the passcode has been lost or forgotten:

The following instructions must be followed exactly as presented!

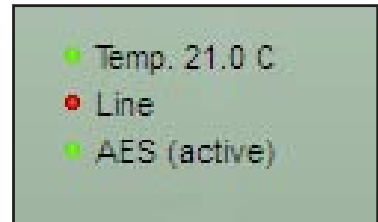
Note that if it is the GUI on your computer that is locked out and NOT the FM-2000 front panel itself please use only steps 1-14 below.

1. Re-install the PC Remote GUI, but this time into a NEW folder on your computer. DO NOT install the GUI back into the same folder and after the installation has completed DO NOT start up the GUI yet!!

2. Make sure that the processor is connected to your network or your computer and that you know its **HARDWARE** IP address (you can use the command prompt **PING** command to make sure that your computer can see the processor). **DO NOT PROCEED** if you do not have network connectivity to the Vorsis processor!
3. Start the new remote GUI on your computer but **DO NOT** click on the Connect button at the top of the GUI next to the Status display. Also, make **VERY** sure that you are not starting the old GUI instead of the new one!
4. Once the GUI is open click on the Devices button on the right hand side of the GUI.
5. In the Devices dialog that opens click the Add button.
6. The Edit Device window will open.
7. Enter a **NEW** name for your processor but use the **same** hardware IP address that your unit currently has.
8. Click OK to close the edit device dialog.
9. In the Devices dialog highlight your new device and click Select. The Devices box will close.
10. Now click on the Online/Offline button left of the Status window at the top of the GUI.
11. Once the GUI is online click on the SYSTEM tab. The System screen will open.
12. Click on **GUI Access** button (**DO NOT CLICK ON GUI LOCK!!!**). The GUI Access Code & Permissions box will open.
13. **Carefully** click **only** the **Clear Access Code** button. Then click OK.
14. The GUI Access Code & Permissions box will close. The PC GUI is now unlocked!
15. On the unit's front panel select a control that was causing the access code prompt to appear.
16. When it appears this time, **DO NOT ENTER A CODE**, **just click OK**. The password box should close.
17. Now, try to open another screen or move a control. The password prompt should not reappear again. **The FM-2000 is now unlocked.**

Indicators/Temperature Readout

Temp – The displayed value is the current operating temperature in degrees Celsius of the FM-2000 motherboard. Normal temperatures should be in the range of 25°C to 45°C, and will vary somewhat depending on ambient temperatures. When the “Temp” indicator is green, the operating temperature inside the FM-2000 is satisfactory. When it is red, the temperature has reached 50°C or higher and is too high.



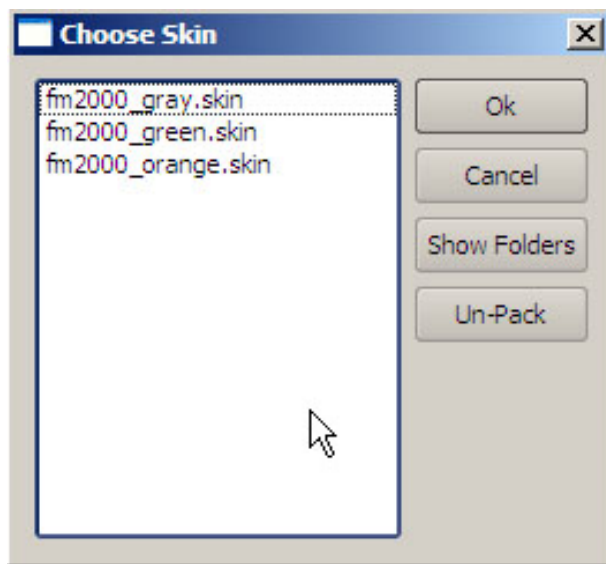
Note: The FM-2000 contains one small fan to stir the air inside the FM-2000 chassis in order to maintain a more uniform temperature distribution. Normal ventilation of the FM-2000 chassis is by natural convection — cooler air from outside the chassis air is drawn in through the lower side slots in the chassis and rising warmed air exits by way of the higher slots along the top of the rear panel. This scheme is quite adequate for controlling temperatures inside the FM-2000 without resorting to forced air cooling.

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

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

Skin

At this time we do not have alternate skins other than the skins shown in the Choose Skin dialog below, nor a method for users to create their own “skins.” When new skins become available they will be downloadable from our Vorsis website (www.vorsis.com).



Events

The FM-2000 contains a comprehensive scheduling utility that allows presets to be automatically changed on desired dates and times. There is also a “Long Term Rotation” utility to allow one time preset changes to be scheduled for any time in the future. To create, edit, or view entries in the Event Scheduler click the “**Events**” button on the System Screen.

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

The screenshot shows the 'Event Scheduler' window with a title bar and a close button. It contains two main sections: 'Weekly Rotation' and 'Long Term Rotation'. The 'Weekly Rotation' section has a 'View' dropdown set to 'All' and radio buttons for days of the week (S, M, T, W, T, F, S). Below this is a table with columns 'Day(s)', 'Time', and 'Preset'. The table contains two entries: '_MTWTF_' at '02:00:00 pm' with preset '019 - Classic Rock III', and '_MTWTF_' at '10:00:00 am' with preset '052 - Oldies Medium'. To the right of the table are 'Add...', 'Edit...', and 'Delete' buttons. The 'Long Term Rotation' section has a table with columns 'Date', 'Time', and 'Preset'. It contains one entry: 'Dec 24' at '06:00:00 pm' with preset '021 - Classical Big'. To the right of this table are 'Add...', 'Edit...', and 'Delete' buttons. At the bottom, it shows 'FM2000 Time: Sun, Jan 15, 2000 - 11:54:13 pm', a '24 Hour Time' checkbox, a 'Set Time...' button, and 'Okay' and 'Cancel' buttons.

Day(s)	Time	Preset
MTWTF	02:00:00 pm	019 - Classic Rock III
MTWTF	10:00:00 am	052 - Oldies Medium

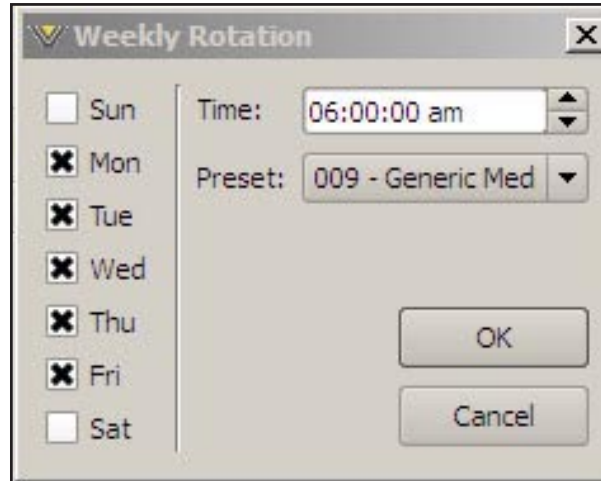
Date	Time	Preset
Dec 24	06:00:00 pm	021 - Classical Big

FM2000 Time: Sun, Jan 15, 2000 - 11:54:13 pm

☐ 24 Hour Time Set Time... Okay Cancel

To create a Weekly Rotation schedule:

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

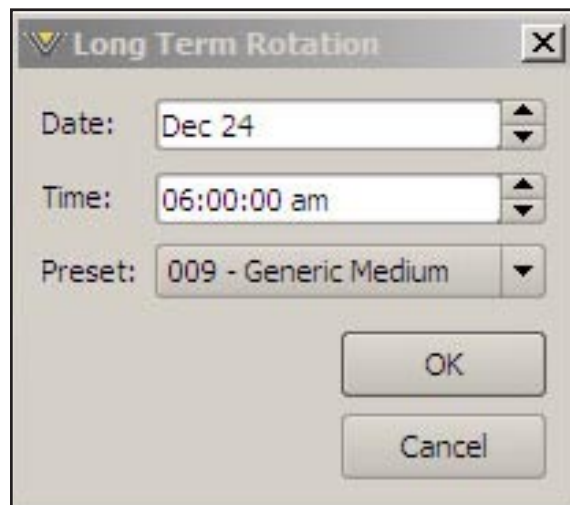


2. On the left column, select the days that the preset should be selected. Any combination of checkboxes may be chosen.
3. Next select the time that the preset should change.
4. Next choose the preset that should be selected at the time programmed in the previous step. When finished editing, click OK.

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

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

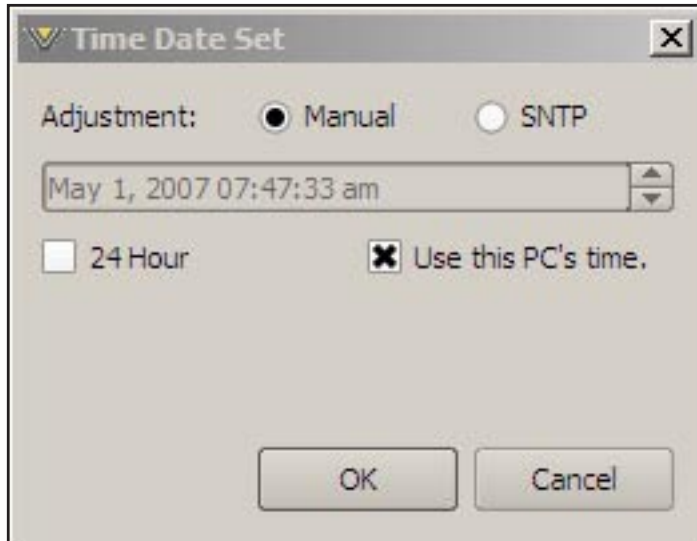
Adding a new event in the Long Term Rotation scheduler works in a similar manner. The Long Term Rotation entry window appears as follows:



Set Time

Allows the system time of the FM-2000 onboard computer to be set or changed. The FM-2000 system time can be synchronized to external time servers (SNTP, or Simple Network Time Protocol) if desired.

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



Backlight Dimmer

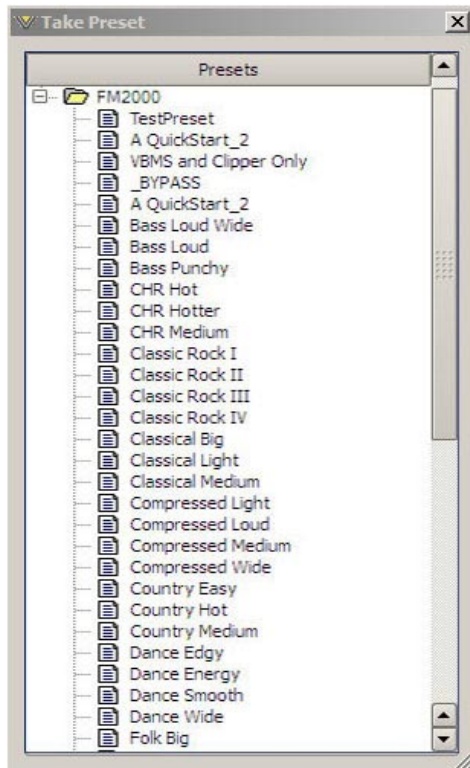
Configures the front panel backlight to be dimmed after about 1 minute of non-use.

Side Bar Region

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

Preset

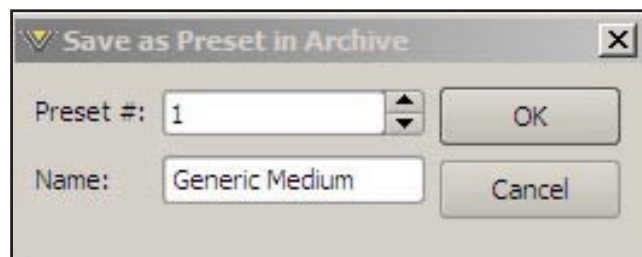
When left-clicked, a “Windows” box appears, showing folders that contain presets, typically “favorites,” and factory presets, this last containing factory presets already in place in the unit. Opening a folder (if not already so) lists the presets in order. Double-clicking on a preset brings it immediately into use in the FM-2000. The box stays open until deliberately closed, allowing differing presets to be double-clicked upon readily in succession. This is a very direct means of comparing presets. *It is important to remember to save the present settings as a preset before invoking others, or they will be lost.*



Note also that you can only Take presets that are already loaded in the FM-2000 from this dialog box.

Save

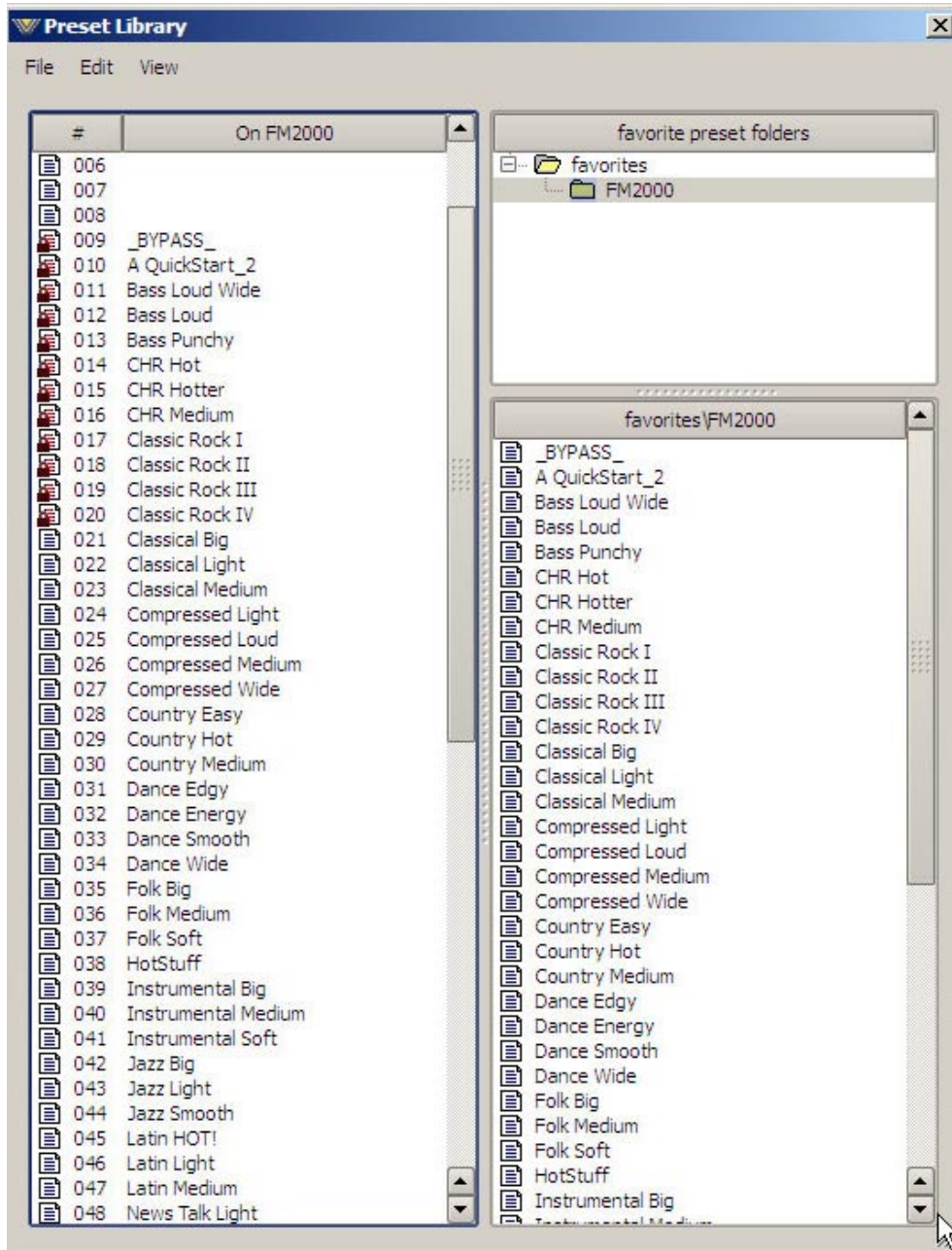
A small “Windows” dialog box appears, which prompts for a name under which to save the present settings of the FM-2000. The preset will be saved into the next available empty slot. Alternatively, by nudging the preset number (“PRE #”) up or down, it is possible to save the present settings in place of an existing preset’s settings. Needless to say, this erases, as in loses, as in destroys, the overwritten preset’s contents. Factory presets are protected from being inadvertently overwritten.



Library

A three-panel “Windows” box appears. The large panel on the left has two columns, being the preset number and the FM-2000 preset contents.

The upper and lower right hand panels give access to the GUI’s presets library; the contents of folders in the upper panel are shown in the lower panel; drag-and-drop allows movement between the library folders and the FM-2000 and its mirror archive in the GUI.



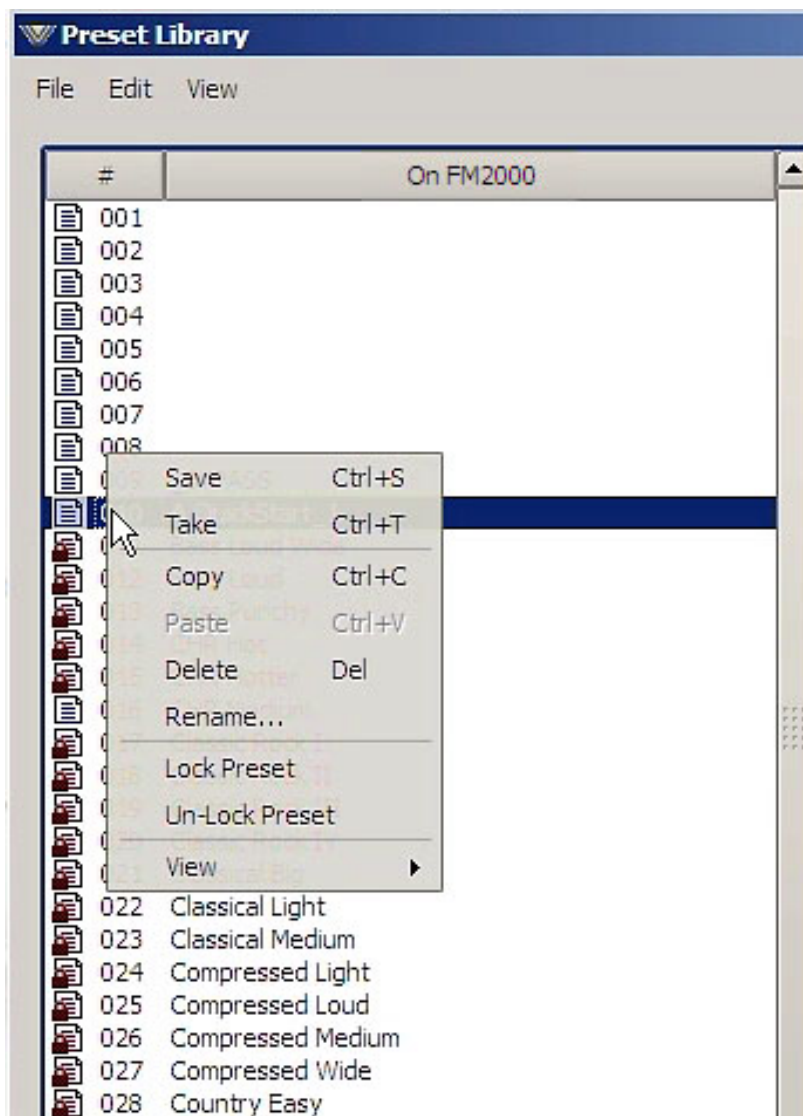
Locking Presets

Presets that have been stored within the processor hardware may be locked by the user to prevent inadvertent overwriting, renaming, or deletion. This is accomplished by opening the Preset Library by clicking on the Library button in the GUI. Once the list of presets is open, the ones actually stored within the processor hardware itself will be visible in the left pane.

There are two ways to manage the lock status of user presets:

- The first method is by highlighting a preset (single left click) and then right clicking it to open a dialog box. Among the options are Lock Preset and Unlock Preset.
- The second method is by highlighting a preset as above, and then clicking the Edit option at the top of the Preset Library dialog box to reveal the Lock Preset/Unlock Preset options.

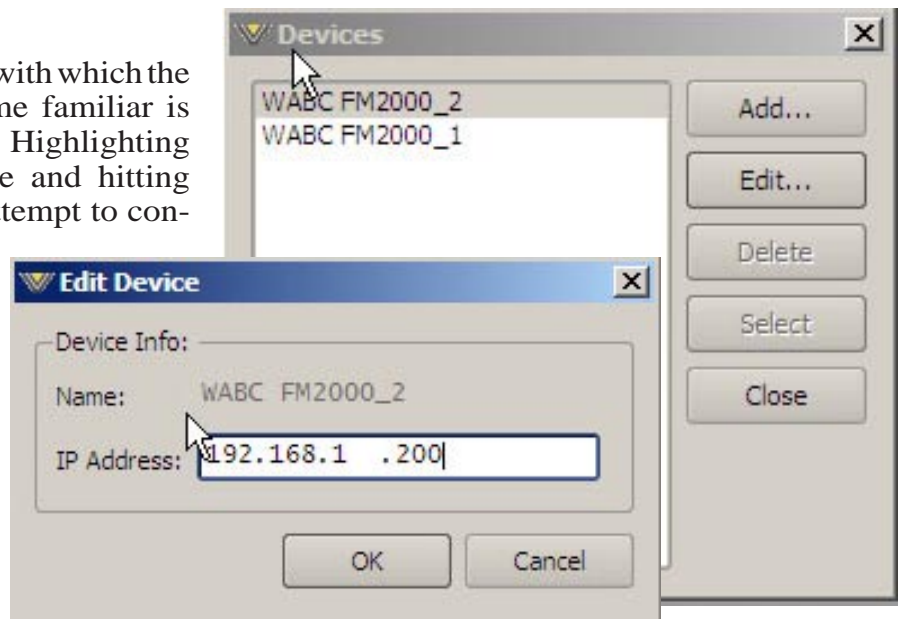
Note that these are user-level lock options and therefore cannot be used to override the lock status of a factory-locked preset!



Devices

A list of FM-2000 devices with which the GUI has had cause to become familiar is shown in a “Windows” box. Highlighting the desired FM-2000’s name and hitting “Select” causes the GUI to attempt to connect to it if you are Online.

In the event no FM-2000’s are listed, or one is adding a further device, or a new one for the first time, “Add” brings up a small dialog box which asks for the new prospect’s name and IP address.



Quick Save (QSave)

The FM-2000 has a unique feature that allows instant comparison of work in progress to a known reference, such as another preset, or some midpoint while adjustments are being made. We call this feature QSave (for Quick Save).

In the upper right side of the FM-2000 GUI you’ll find three buttons as shown below:

QSave A and QSave B represent two temporary buffers that can hold all current FM-2000 processing settings as long as power is applied to the unit.

While QSave A is highlighted green any adjustments that you make to controls on the FM-2000 are being saved to its temporary buffer “A.”

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

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

- Recall the factory or user preset that you wish to adjust.
- Ensure that QSave A is highlighted. If it is not, press its button to highlight it.
- Press the B=A button. This will copy the contents of QSave buffer A to QSave buffer B. Now the contents of both buffers are identical.
- Change some settings on the FM-2000. These settings will automatically be stored in the A buffer.
- Compare your changed settings to the recalled factory preset by pressing the QSave B button.



- Compare those settings back to the factory preset by pressing QSave A.
- When you are happy with your changes you can commit them to a user preset using the Save dialog explained previously.

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

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

Title Bar Region



Along the top edge of the FM-2000 GUI screen (in line with the “Vorsis FM-2000” 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 FM-2000 GUI is capable of controlling multiple FM-2000’s (“Devices”) and managing the “Presets” within them. Whether a connection is made (“Status”), and which device and which preset are presently under command are indicated. Double-clicking these indications will gain access to respective management screens.

Status

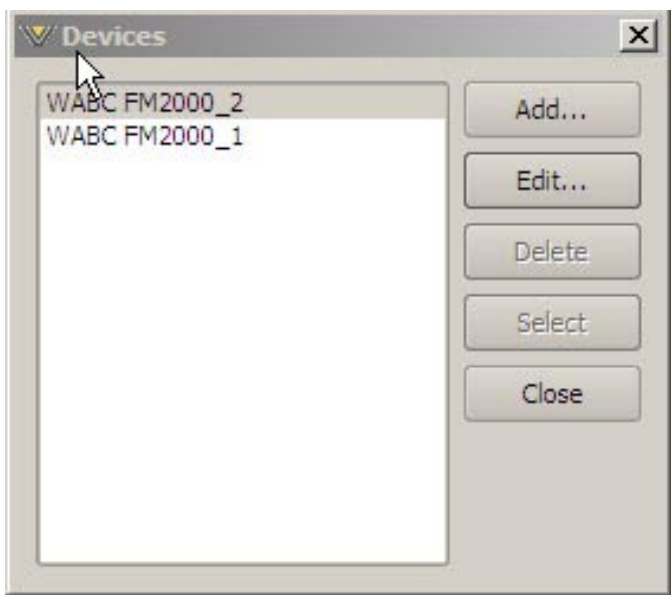
This indicates: “Online,” meaning the GUI is in communication with and directly reflects an FM-2000; “Offline,” meaning the GUI and FM-2000 are disconnected; “Trying,” meaning the GUI is attempting to find the FM-2000 on the LAN and connect to it. An adjacent button toggles between “Online” and “Offline.”

Devices

The name of the FM-2000 to which the GUI is connected (or with which it is attempting to connect) shows here in amber.

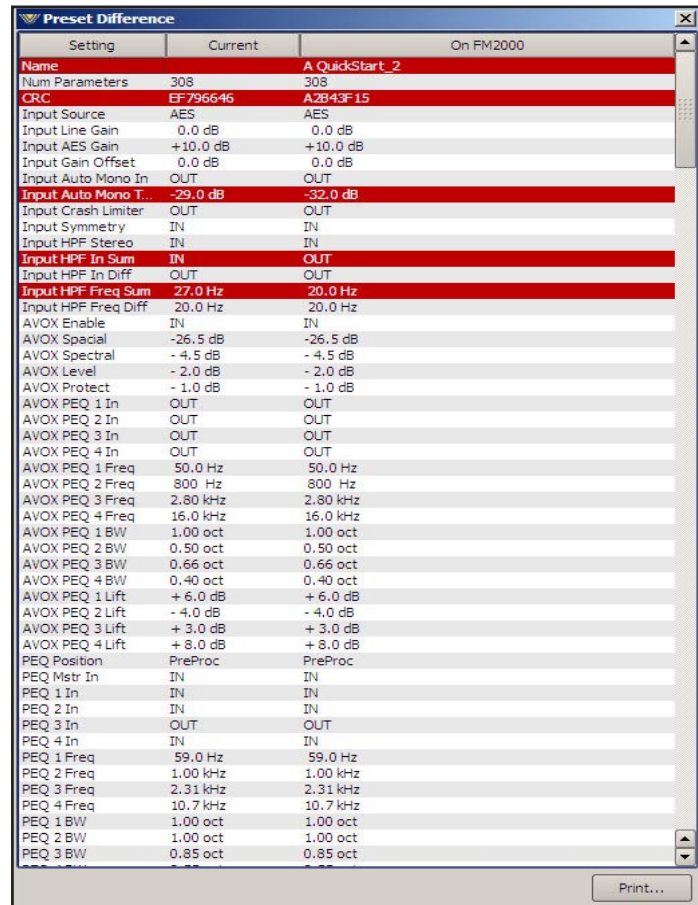
Double-clicking on the name area brings up the “Devices” Windows box. A list of FM-2000 devices with which the GUI has had cause to become familiar is shown in a Windows box. Highlighting the desired FM-2000’s name and hitting “Select” causes the GUI to attempt to connect to it if you are Online.

In the event no FM-2000’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 FM-2000's settings for this preset agree with those in the GUI's mirror archive; red indicates a discrepancy. Double-clicking on the name brings up a "Differences" — parameter-by-parameter, setting-by-setting — comparison list between the current FM-2000 settings and the recalled preset.



Setting	Current	On FM2000
Name		A Quick-Start_2
Num Parameters	308	308
CRC	EF 796646	A2B43F 15
Input Source	AES	AES
Input Line Gain	0.0 dB	0.0 dB
Input AES Gain	+10.0 dB	+10.0 dB
Input Gain Offset	0.0 dB	0.0 dB
Input Auto Mono In	OUT	OUT
Input Auto Mono T...	-29.0 dB	-32.0 dB
Input Crash Limiter	OUT	OUT
Input Symmetry	IN	IN
Input HPF Stereo	IN	IN
Input HPF In Sum	IN	OUT
Input HPF In Diff	OUT	OUT
Input HPF Freq Sum	27.0 Hz	20.0 Hz
Input HPF Freq Diff	20.0 Hz	20.0 Hz
AVOX Enable	IN	IN
AVOX Spatial	-26.5 dB	-26.5 dB
AVOX Spectral	- 4.5 dB	- 4.5 dB
AVOX Level	- 2.0 dB	- 2.0 dB
AVOX Protect	- 1.0 dB	- 1.0 dB
AVOX PEQ 1 In	OUT	OUT
AVOX PEQ 2 In	OUT	OUT
AVOX PEQ 3 In	OUT	OUT
AVOX PEQ 4 In	OUT	OUT
AVOX PEQ 1 Freq	50.0 Hz	50.0 Hz
AVOX PEQ 2 Freq	800 Hz	800 Hz
AVOX PEQ 3 Freq	2.80 kHz	2.80 kHz
AVOX PEQ 4 Freq	16.0 kHz	16.0 kHz
AVOX PEQ 1 BW	1.00 oct	1.00 oct
AVOX PEQ 2 BW	0.50 oct	0.50 oct
AVOX PEQ 3 BW	0.66 oct	0.66 oct
AVOX PEQ 4 BW	0.40 oct	0.40 oct
AVOX PEQ 1 Lift	+ 6.0 dB	+ 6.0 dB
AVOX PEQ 2 Lift	- 4.0 dB	- 4.0 dB
AVOX PEQ 3 Lift	+ 3.0 dB	+ 3.0 dB
AVOX PEQ 4 Lift	+ 8.0 dB	+ 8.0 dB
PEQ Position	PreProc	PreProc
PEQ Mstr In	IN	IN
PEQ 1 In	IN	IN
PEQ 2 In	IN	IN
PEQ 3 In	OUT	OUT
PEQ 4 In	IN	IN
PEQ 1 Freq	59.0 Hz	59.0 Hz
PEQ 2 Freq	1.00 kHz	1.00 kHz
PEQ 3 Freq	2.31 kHz	2.31 kHz
PEQ 4 Freq	10.7 kHz	10.7 kHz
PEQ 1 BW	1.00 oct	1.00 oct
PEQ 2 BW	1.00 oct	1.00 oct
PEQ 3 BW	0.85 oct	0.85 oct

Notes on “Online” and “Offline” Working

Most often operationally it will be required that the GUI act directly and instantaneously upon the FM-2000 selected under “Devices:” changes on the main control screen take immediate effect on the signal processing and any selected presets are immediately rendered active. This is “Online.”

“Offline,” however, is more of a supervisory and system management mode and is for those who feel the need to set up complex audio processors without listening to them. Changes can be made to preset settings and configurations, and manipulation of the presets selection list can be made without affecting signal processing within the FM-2000 itself, which can continue to be processing away obliviously. (Preset list manipulation can be undertaken “Online,” too).

Importantly, any adjustments made on the GUI “Offline” will not take effect until the FM-2000 is rendered “Online” again.

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

- “Online” - the double-clicked preset is immediately sent to the FM-2000 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 FM-2000's audio.

Accessing Menu Options

Right clicking anywhere on the Vorsis FM-2000 Control Panel will open a pop up menu tree with access to *File*, *Hardware*, and *Presets* choices. These choices lead to sub-menus and dialog boxes that may also be accessed by clicking on other dedicated buttons on the main Vorsis FM-2000 control panel. As with many Windows programs, there are multiple ways to access menu trees — go ahead and explore!



File Menu Items

The *File* menu tree may be accessed by right clicking anywhere on the main FM-2000 Control Panel.

About... - brings up the About box to indicate the GUI version.

Center Window - centers the main GUI window on the screen.

Choose Skin... - brings up the Choose Skin dialog box.

Exit - exits the GUI program.

Hardware Menu Items

The *Hardware* menu tree may be accessed by right clicking anywhere on the main FM-2000 Control Panel. Please note that many of these functions require you to be connected (Online) to an FM-2000. Sub menu choices include:

Devices... - opens the Devices dialog box. Allows the creation, editing, selection, and deleting of Vorsis processors connected to your system.

On-Line Mode... - toggles between ONLINE and OFFLINE modes.

Event Scheduler... - opens the Event Scheduler dialog box for editing Events.

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

Version... - displays the current software versions running in your FM-2000 hardware.

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

Update Front Panel GUI... - opens the “FM2000 Front Panel GUI Loader” dialog box. Only files of type *.lbn are visible. See below.

Future versions of the FM-2000 software may be released to implement new features or correct known problems. The Software Update menu choice opens a dialog box and prompts the user to select a file to be uploaded to the FM-2000. Upon completion, you will be asked to restart the FM-2000 (cycle power).

Only verified updates provided by Vorsis will work!

FM-2000 Hardware Update

The FM-2000 hardware is extremely easy to update whenever new firmware becomes available from Vorsis. There are two different files involved in updating the FM-2000. These are:

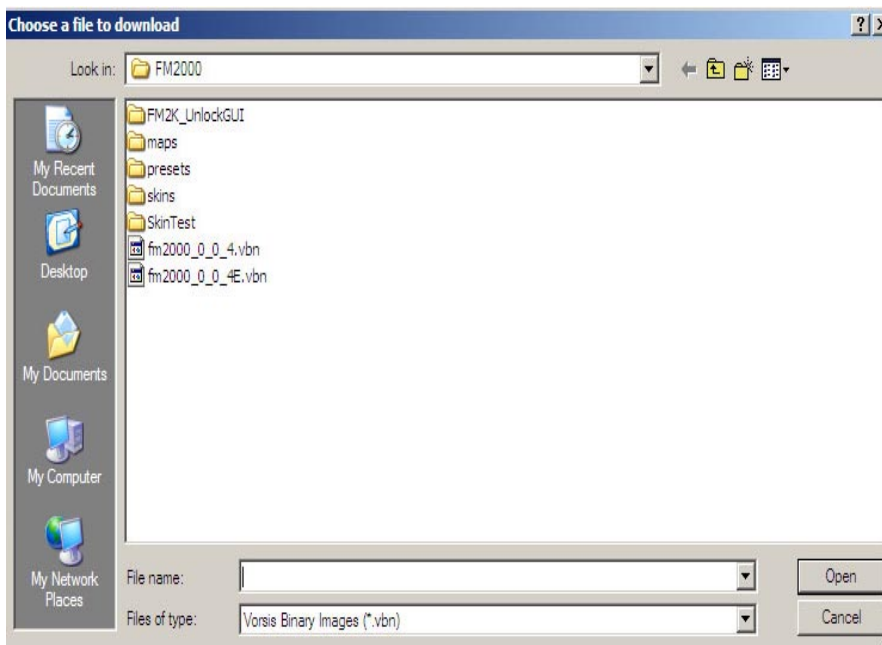
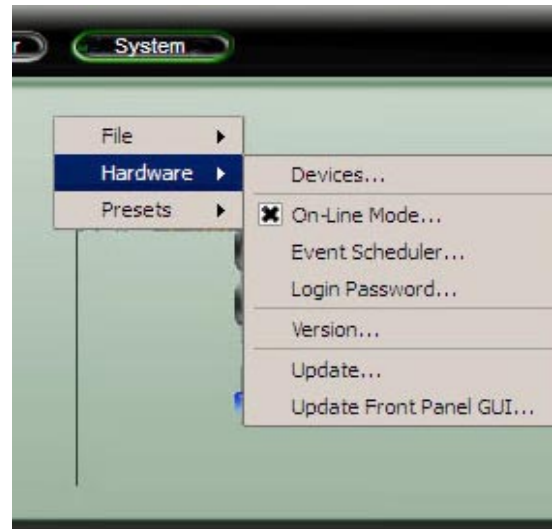
The “VBN” — VBN stands for “Vorsis Binary Nugget.” This file contains motherboard application code, FPGA images, and DSP code.

The “LBN” — LBN stands for “Linux Binary Nugget.” This file contains the front panel GUI application code and skins for the front panel graphics display.

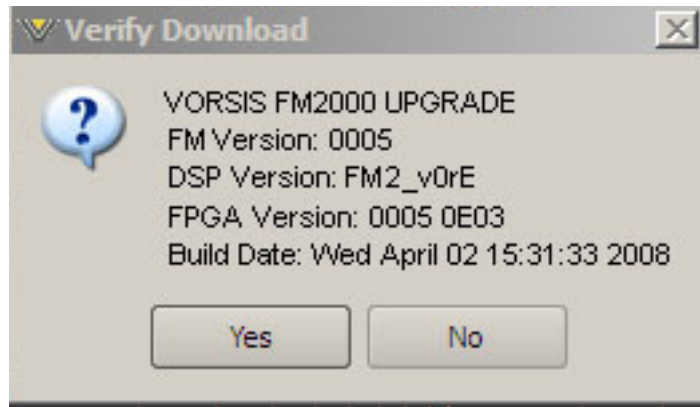
VBN and LBN file updates may be performed separately, or they may be performed together if updates for both sections of the FM-2000 have been made available.

To update the FM-2000, right click on any portion of the upper control area in any screen of the remote GUI to open popup menu tree.

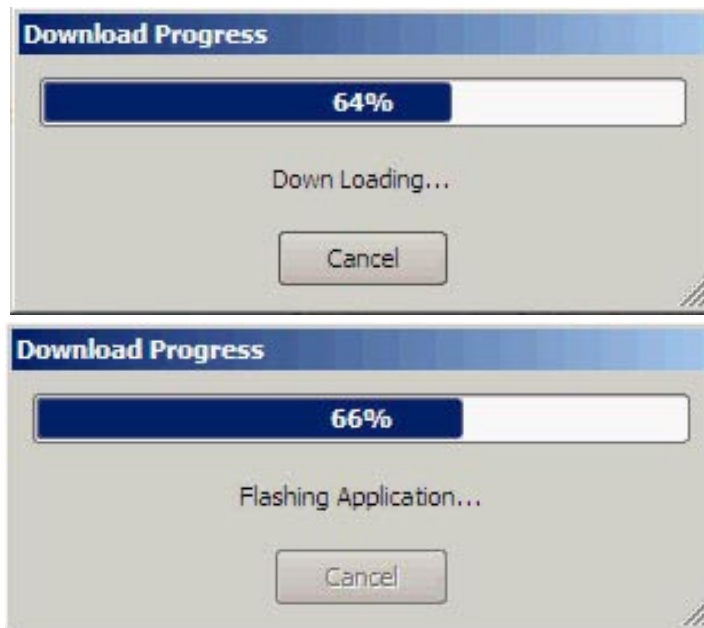
For VBN updates of the motherboard, select the “Update...” option listed above the “Update Front Panel GUI...” option. The following screen will appear.



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



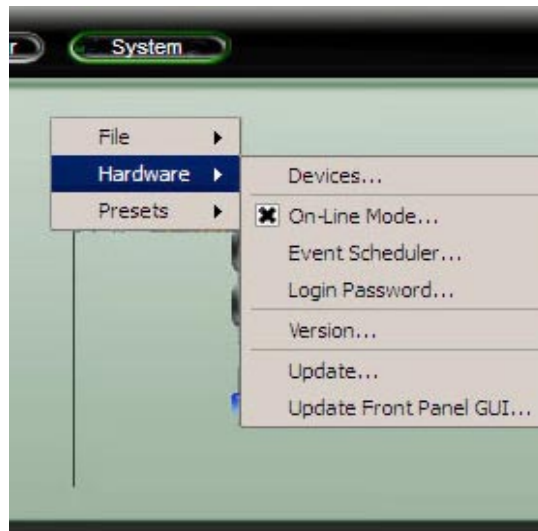
Check again that the versions match the documentation! Then click “Yes.” The following boxes will pop up as the download progress occurs:



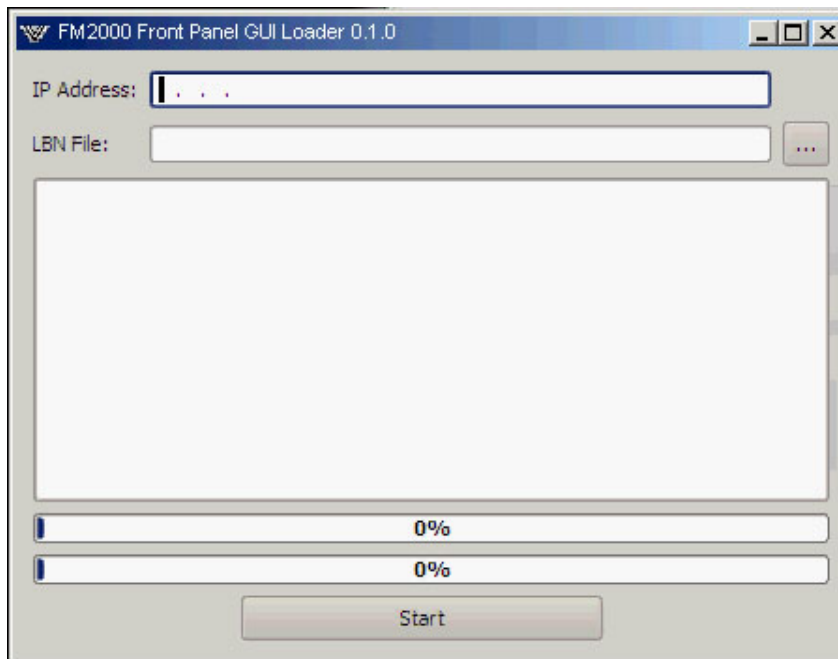
When the process has finished, you will be prompted to reboot the FM-2000 by the prompt below:



When the FM-2000 reboots the new VBN code will be running on the motherboard.

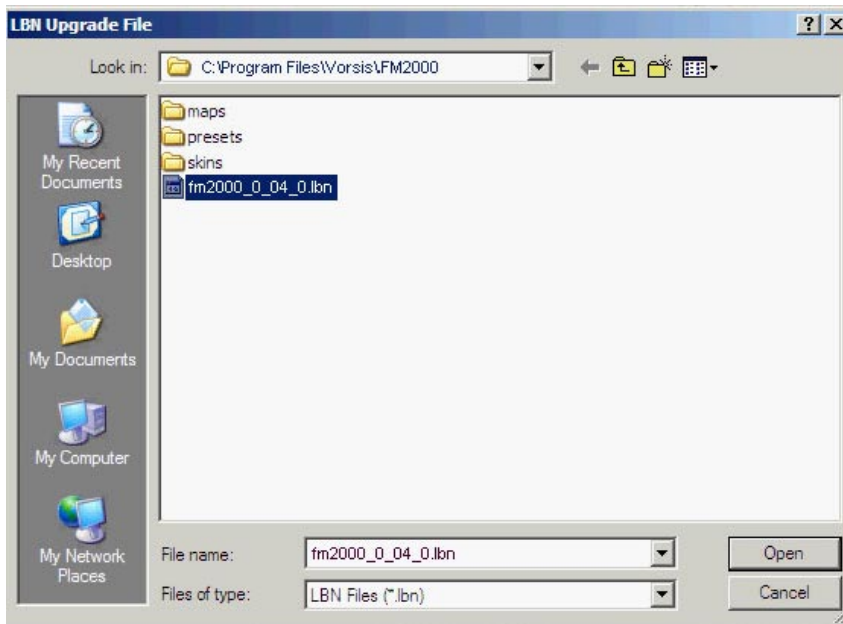


For LBN updates of the Front Panel, select “Update Front Panel GUI...” and the following screen will appear.

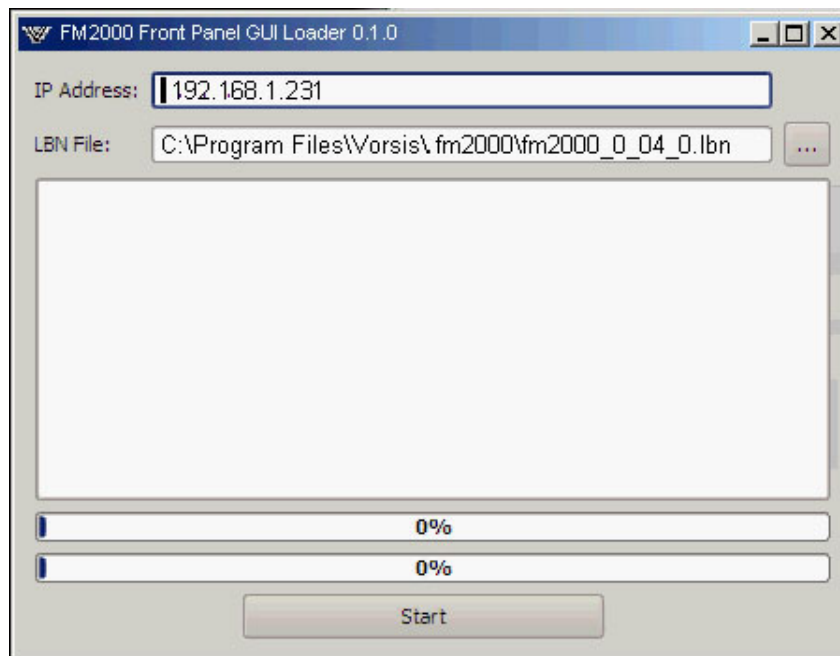


Notice that the IP Address field is blank. This must be filled in to agree with the IP address of your FM-2000 as it is currently configured. This information is available on the front panel of the unit itself. Use Right Click/File/Settings on the front panel itself to locate this data

Next, click on the “...” button at the right end of the “LBN File:” field to navigate to the location of the LBN file. Verify that the version number is correct in that it matches the version number listed in the documentation received about the firmware upgrade.

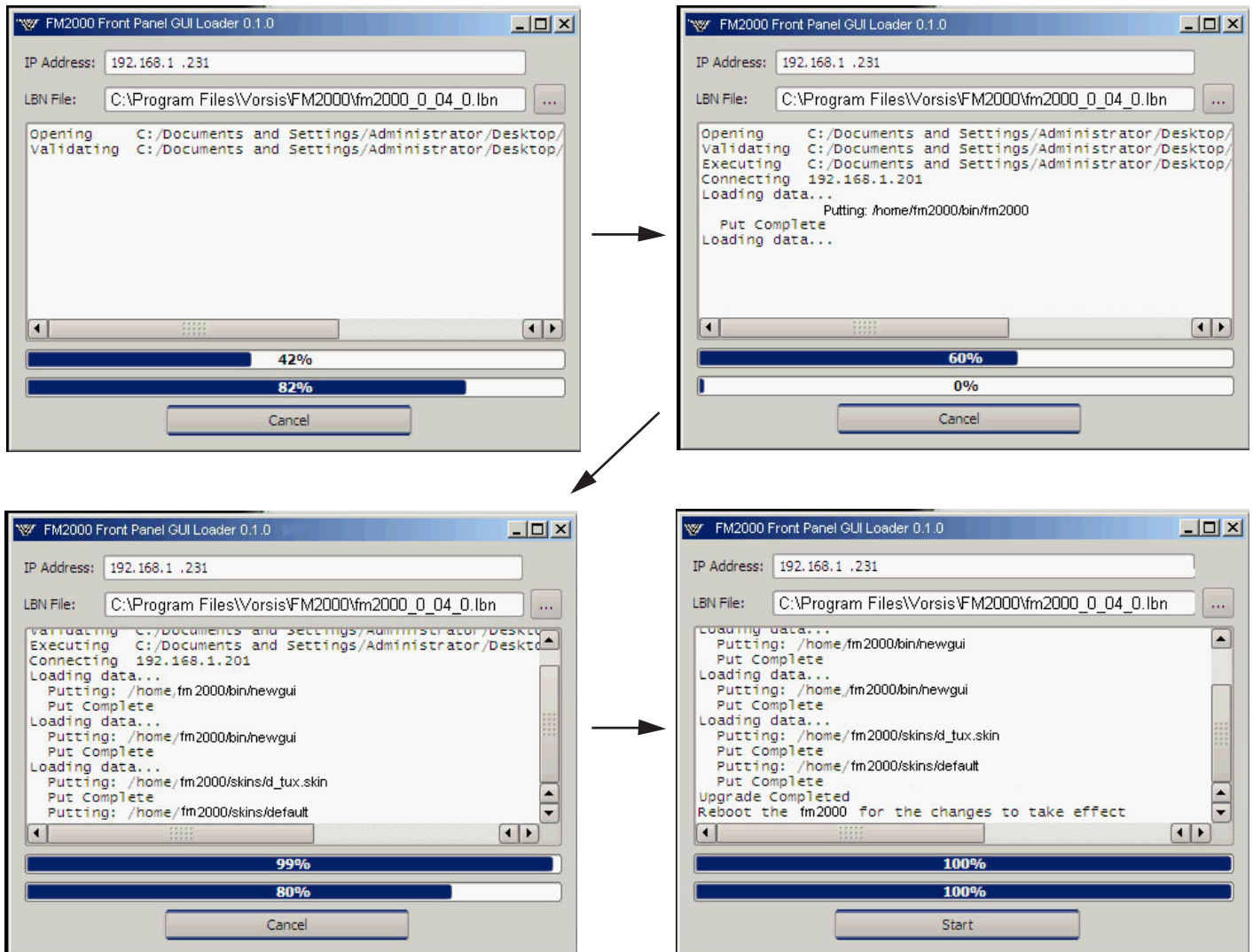


If the information is correct, select the “Open” option in the “Choose a file to download” window. The following dialog box will appear



When ready to begin the update process, click the Start button at the bottom. It is important to **NOT** interrupt the upgrade process by any means while it is in progress!!

While the download/update process is running the progress of the update will be shown. Refer to the screen shots on the next page for what this process should typically look like.



When the updating process has completed, you will be prompted to reboot the FM-2000 to bring the updates into effect.

VBN/LBN Updating Order Sequence

The VBN and LBN updates, while separate update processes, may be completed in any order and/or at the same time before rebooting the FM-2000 to bring the updates into effect.

For instance, you could perform the VBN update and then, when prompted to reboot, *not* reboot, and perform the LBN update instead, and THEN reboot after that process has completed.

Presets Menu Items

The *Presets* menu tree may be accessed by right clicking anywhere on the main FM-2000 Control Panel.

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

Save - brings up the Save as Presets dialog box.

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

I/O Schematic Drawings & Load Sheets

Chapter Contents

FM-2000 Signal Flow Diagram 4-2

Audio Processor (FM-2000)

Schematic 4-3

Load Sheet 4-8

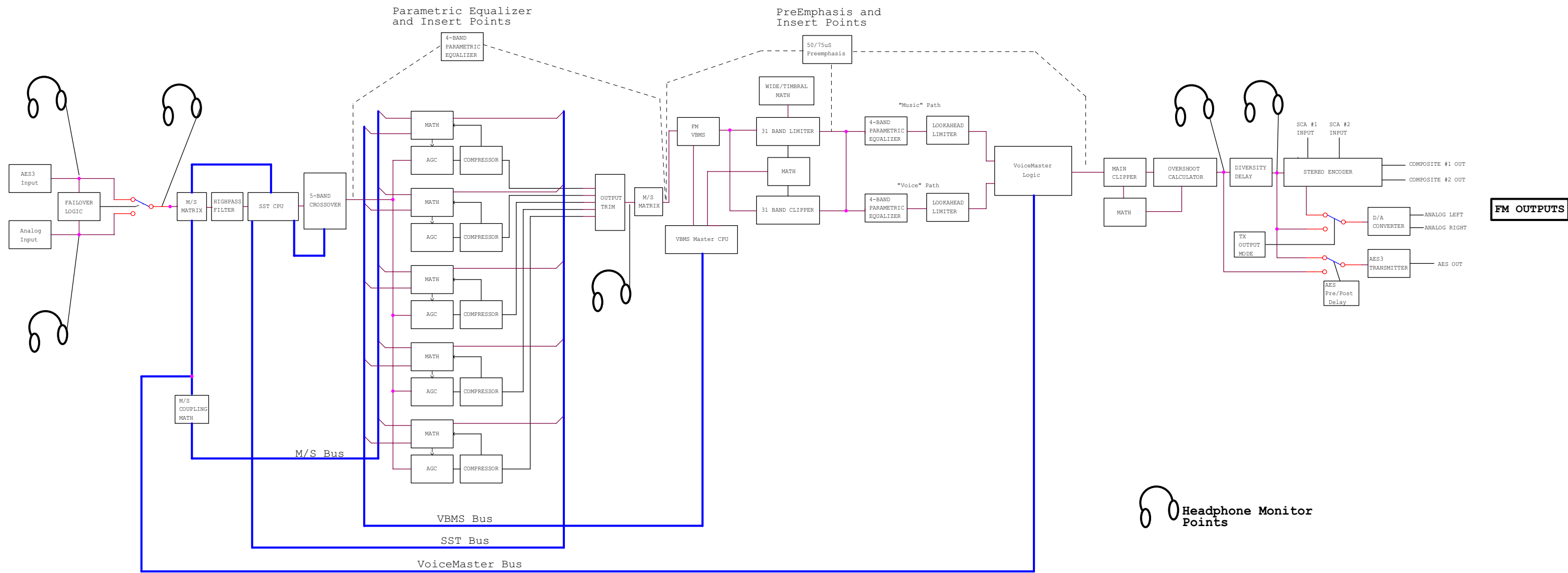
Audio Processor Switch Card (FMSW-2000)

Schematic 4-9

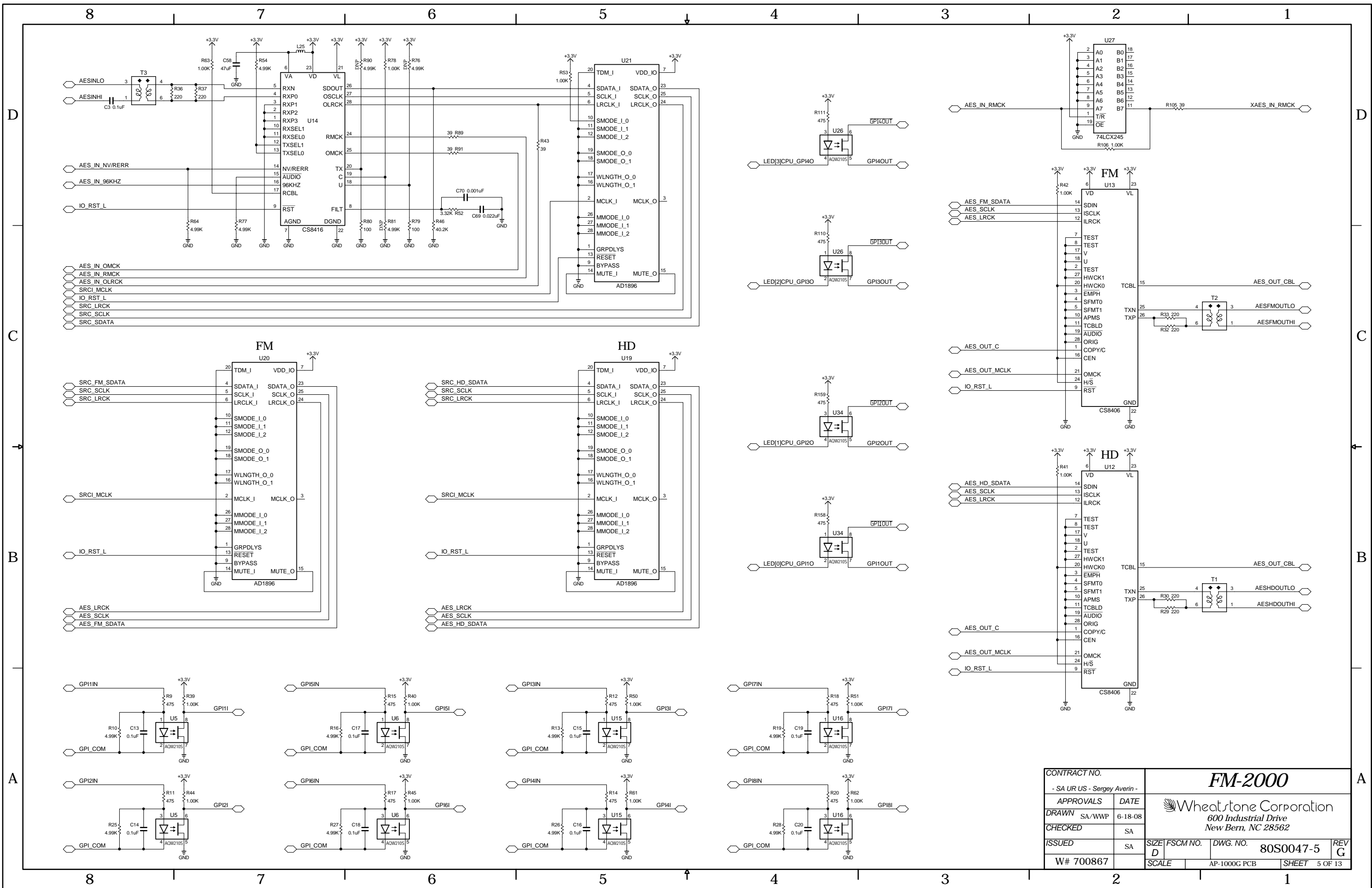
Load Sheet 4-10



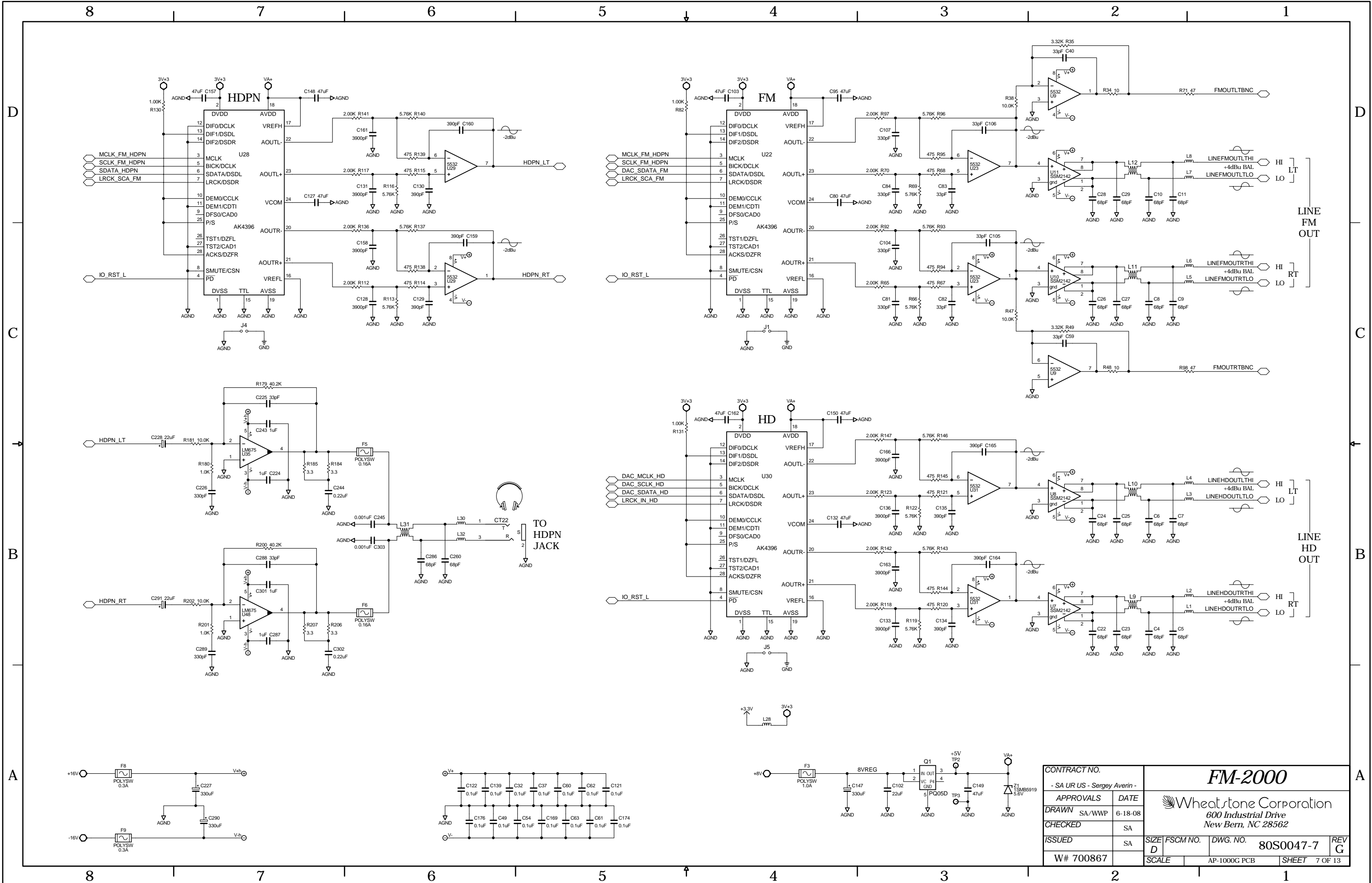
FM-2000 General Signal Flow



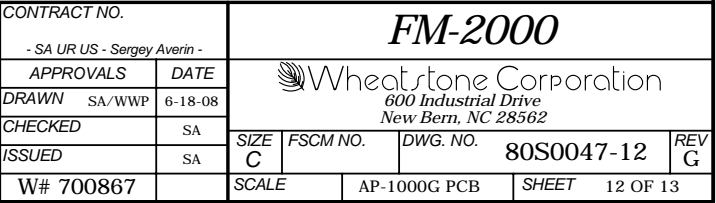
FM-2000 Signal Flow Diagram



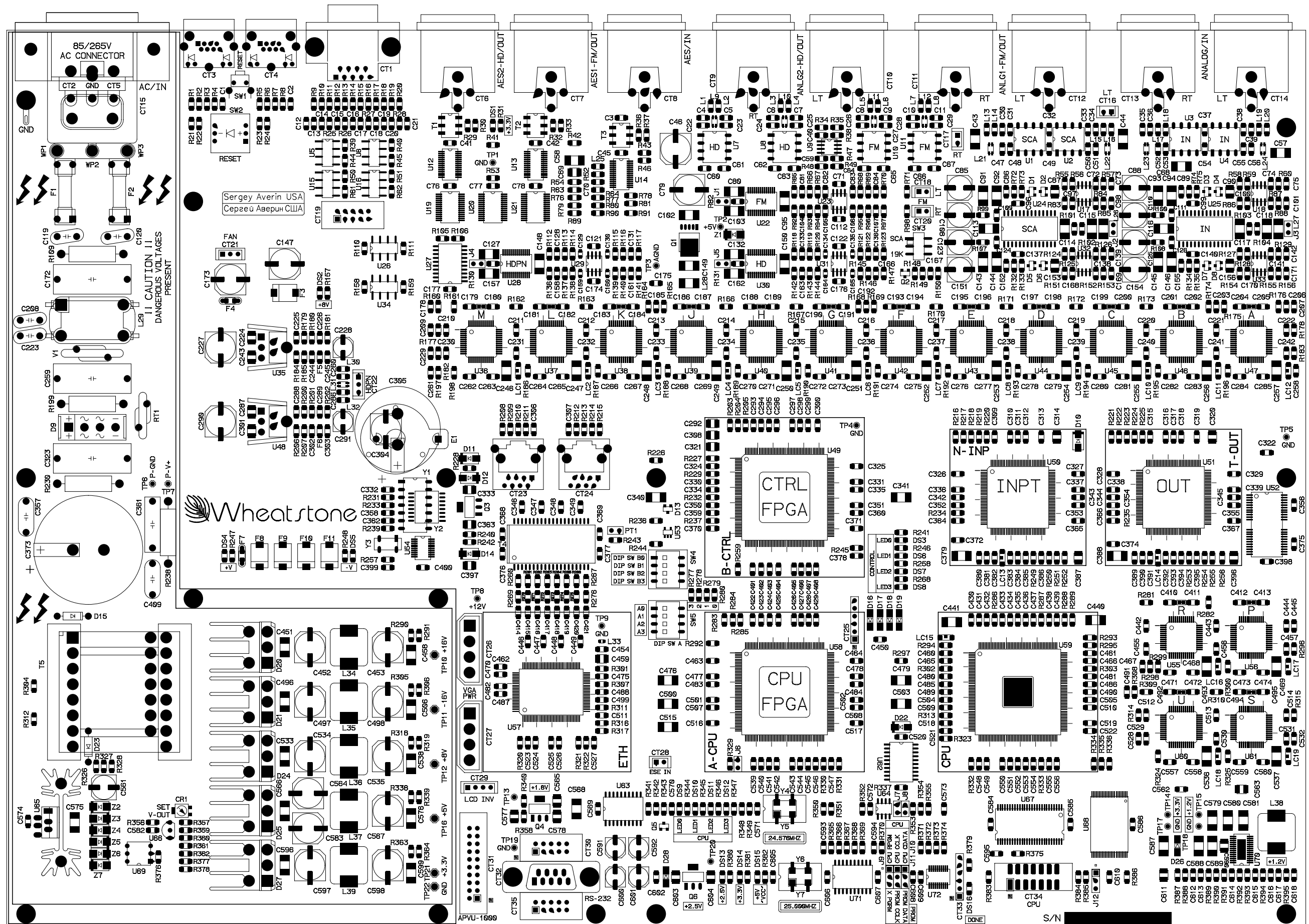
FM-2000 Audio Processor Schematic



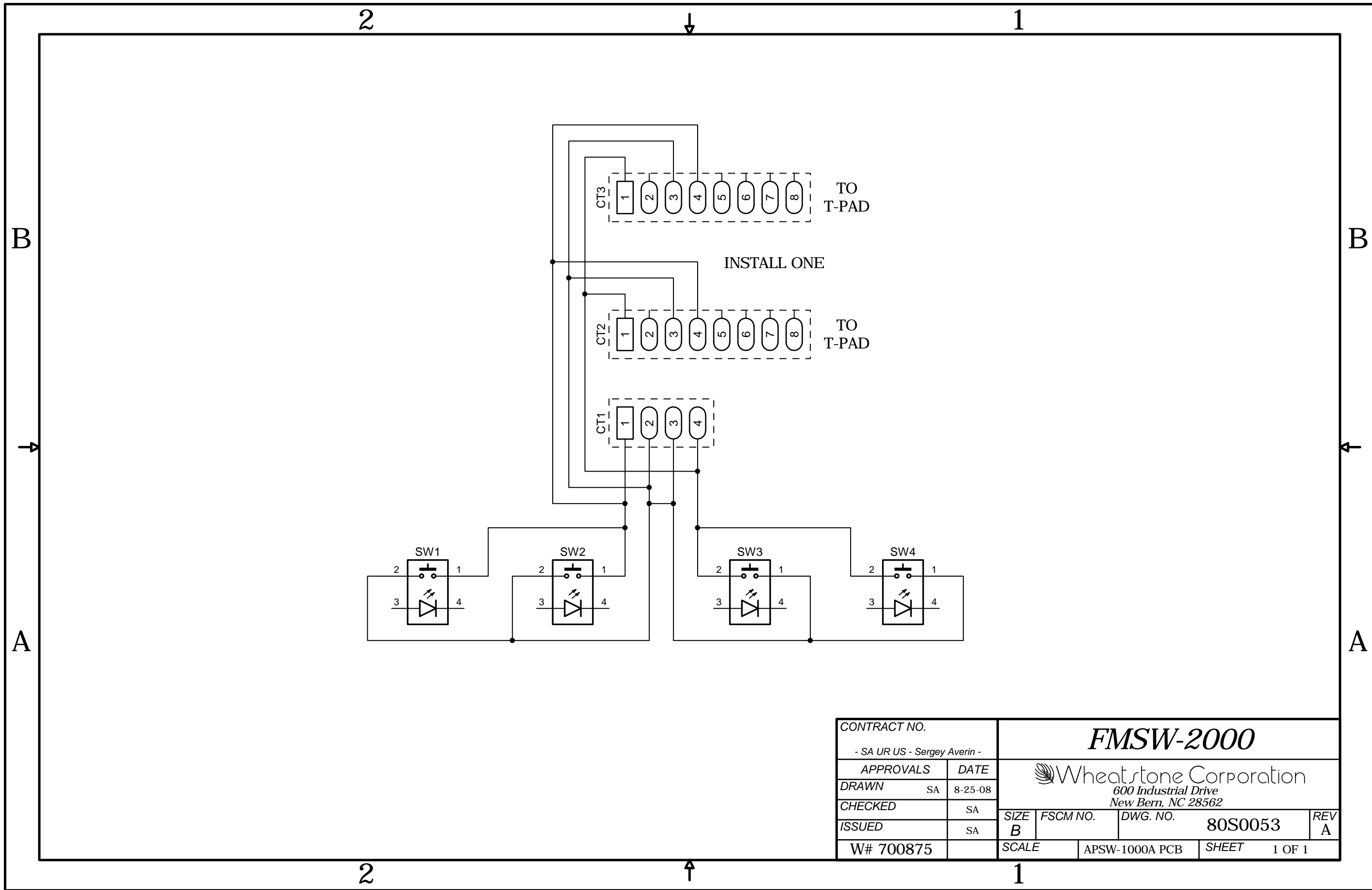
FM-2000 Audio Processor Schematic




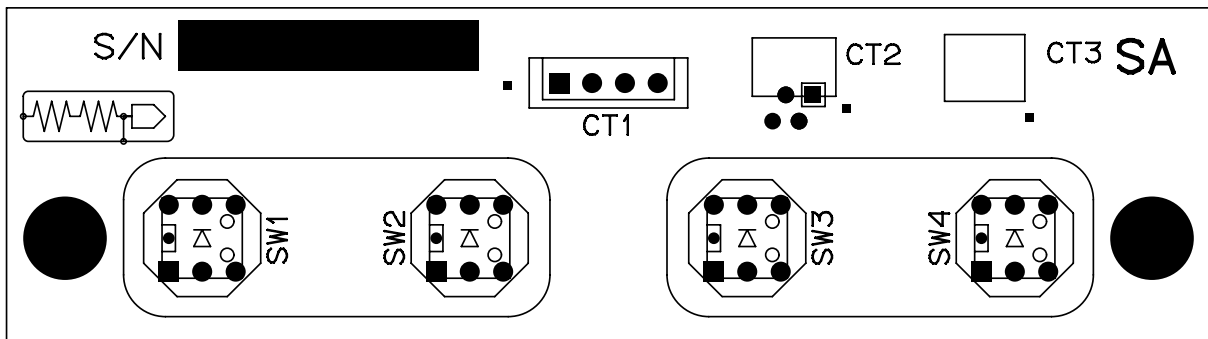
page 4 - 7



FM-2000 Audio Processor Load Sheet



CONTRACT NO.		<i>FMSW-2000</i>			
- SA UR US - Sergey Averin -		 Wheatstone Corporation 600 Industrial Drive New Bern, NC 28562			
APPROVALS	DATE				
DRAWN SA	8-25-08	SIZE	FSCM NO.	DWG. NO.	REV
CHECKED	SA	B		80S0053	A
ISSUED	SA				
W# 700875		SCALE	APSW-1000A PCB	SHEET	1 OF 1



FMSW-2000 Audio Processor Switch Card Load Sheet

Appendix

Contents

Parameters, Units and Ranges	A-2
------------------------------------	-----

Parameters, Units and Ranges

Approximately following the FM-2000's signal path, these are the values and ranges appropriate to each type of processing.

System Level

Headroom:	20dB
Nominal Operating Level:	-20dBFS digital

Analog Line Input

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

Digital Line Input

Data Standard:	AES3 (AES/EBU)
Data Amplitude:	Per AES3-2003 assuming minimum allowable output signal amplitude of 2V and minimum allowable input signal amplitude of 200mV

Input Gain Adjustment Ranges

Gain Adjustment:	+/- 24dB
Gain Adjustment Increments:	0.5dB
Gain Calibration:	A gain control setting of 0.0 aligns an external 0dBFS signal with the FM-2000's 0dBFS internal reference

Input Failsafe

Type:	Automatic
Analog Fail Cause:	Audio level below -48dBFS.
Response Time:	30 seconds
Digital Fail Cause 1:	Audio level below -48dBFS
Response Time:	30 seconds
Digital Fail Cause 2:	Corrupted or invalid AES data
Response Time:	Immediate

Audio Level Balance

Type:	Common to Analog and Digital inputs
Analog/Digital L/R Balance Range:	+/-12dB
Analog/Digital L/R Balance Increments:	0.5dB

Voice Symmetry Phase Rotation

Operating Modes:	In/Out
Filter Type:	4 th Order Allpass

Auto Monophonic

Auto Mono Modes:	Off/On
Auto Mono Threshold:	-40.0dB to 0.0 dB
Auto Mono Timing:	Program Controlled

High-Pass Filter

HPF Filter Class:	24dB/octave Butterworth
Frequency range:	20Hz – 300Hz
HPF Insert Modes:	Off / Stereo L/R / Sum/Difference

Equalization - Three sections of four-band Parametric Equalization

EQ Set 1 – routable to pre or post-five-band AGC.	
EQ Set 2 – post-FM multiband limiter.	
EQ Set 3 – VoiceMaster.	
Frequency:	20Hz – 20 kHz
Bandwidth:	0.2 – 3.0 octaves
Lift / Cut:	+/- 14.0dB pre/post five band AGC
	+/- 3.0dB post FM multiband limiter

Five-Band AGC/Compressor

Operates in sum/difference, or “matrix” mode with the gain linked at AGC time constants.

Drive Gain:	-80.0dB to +6.0dB in 0.5dB steps
Makeup Gain Range:	-20.0dB to +48.0dB, 0.5dB steps
AGC/Compressor Thresholds:	- 50dBFS to -80dBFS
AGC Threshold Backoff:	0dB to -12dB
AGC Attack:	50mS – 1500ms (1.5 sec)
AGC Release:	100mS – 7.0 seconds
AGC Band Coupling:	0dB to -30.0dB, 0.5dB steps
Compressor Attack:	3.0mS – 1000mS (1Sec)
Compressor Release:	20mS – 1000mS (1Sec)
Ratio:	1:1 – 20:1
AGC Threshold:	-20.0dB to -79dBFS, plus off
Gate Thresh Trim:	+/- 6dB per band
Gated Modes:	Ooze/Freeze
Gate Delay:	50mS to 500mS

SST Drive:	+30.0dB to +40.0dB in 0.5dB steps
SST Ramp Rate:	500mS – 7.0 seconds
SST Gate Thresh:	-6.0dB to -70.0dB in 0.5dB steps
SST Window Size:	0dB to 12.0dB in 0.5dB steps

Crossover Frequencies

Super Low to Low Band:	40.5 Hz to 120 Hz
Low to Low Mid Band:	149 Hz to 334 Hz
Low Mid Band to High Mid Band:	817 Hz to 1.59 kHz
High Mid Band to High Band:	2.52 kHz to 7.55 kHz
Band Output Trims, Sum:	+/- 6dB
Band Output Trims, Difference:	+/- 6dB

31-Band FM Output Peak Limiter

A four band parametric EQ follows the 31-band limiter and feeds the VBMS with multiple operating styles, then an oversampled, zero overshoot lookahead limiter or main clipper with multiple operating styles, and then an FM diversity delay.

Overall Controls	
Multiband/Drive:	0 – 100%
VBMS Drive:	+/- 10.0dB
VBMS Style:	Off/ Soft/ Hard
VBMS Output:	+/- 10.0dB
VBMS Frequency:	60 - 300Hz
Multiband Knee:	Soft/ Hard
Multiband Threshold:	+/-10dB relative to Drive
Multiband Attack:	0.1ms – 1000ms (1Sec)
Multiband Release:	30ms – 1000ms (1Sec)
Limiter Mathematical Modes:	Wide In/Out – Timbral In/Out
Pre-emphasis Modes:	Off/50uS/75uS
Pre-emphasis insert points:	Pre-MB Limiter Pre-lookahead limiter Post-lookahead limiter (pre-clipper)
Lookahead Limiter (Threshold) Trim:	+/- 6.0dB
Lookahead Limiter Attack:	0.2ms – 100ms
Lookahead Limiter Release:	33ms – 330ms
Lookahead Limiter Delayed Release:	100ms – 1000ms (1Sec)
FM Output Level:	-79.95dB to +12.00dB, or OFF
Limiter/Clipper Drive:	+/-42dB in 0.05dB steps

Main FM Clipper Styles:	Off/Hard/Firm/Round
De-emphasis:	Complementary as applied in pre-emphasis.
Output Signal available as:	Pre or post diversity delay AES 3-pin XLR (digital), and Stereo Analog (pair 3-pin XLR connectors)

Stereo Encoder

Reference grade stereo encoder with embedded composite processing, test oscillator, SCA digitizer, and balanced and unbalanced composite outputs.

Overall Controls	
Stereo Width Limiter:	10% to 100% in 10% steps, plus off
Stereo Pilot Injection:	0 – 20%, 0.1% steps
Stereo Pilot Phase:	+/- 22.5 degrees reference to 38kHz.
SCA 1 Input:	Analog, 10kohm input impedance, +24dBu max. input level
SCA Input 1 Gain:	-79.95dB to +10.00dB, or OFF
SCA 2 Input:	Analog, 10kohm input impedance, +24dBu max. input level
SCA Input 2 Gain:	-79.95dB to +10.00dB, or OFF
TX 1 Output Level:	-79.95dB to +6.0dB, 0.05dB steps, or OFF
TX 2 Output Level:	-79.95dB to +6.0dB, 0.05dB steps, or OFF
Composite Processor Modes:	May be selected to provide pilot tone only Lookahead Limiter Soft Clipper Hard Clipper
Stereo Encoder Output Operating Modes:	Analog L/R Analog L/R De-emphasized Unbalanced and Balanced Multiplex Outputs
TX 1/TX 2 Output Levels:	Nominal output level range 1V - 4V RMS or balanced from an XLR.

Presets

An advanced preset management system allows the creation, storing, and recall of 80 presets within the FM-2000 itself and an unlimited number on a host PC. All signal-processing parameters are contained in these presets, allowing the preservation of the complete processing “environment.” Optionally, Input and Output settings may be stored and recalled with presets.

Preset Encryption

Presets are encrypted using the FM-2000 internal 32 bit serial number as the encryption key.

Onboard Real-time clock

A highly accurate internal real-time clock allows the automatic recall of presets at pre-determined times. This clock may be set to “free-run” or be synchronized to an external on- or off-site SNTP server.

Day-Parting and Long-term Scheduling

A “weekly” preset scheduling establishes a weekly cycle of preset changes, allowing for automatic programmed day-by-day exclusions or additions.

Additionally, a separate long-term scheduling table permits the establishment of “one-off” preset changes for *any* future time and date.

General Purpose Input (GPI)

Eight optically isolated inputs are slaved to the first eight preset slots.

Presets may be recalled by providing the appropriate voltage to the associated GPI port to create a “Logic High” and may be either momentary or latching as desired. Voltages are applied between the appropriate GPI pin and Common with the GPI pin being the most positive voltage. Please see manual text for the required current limiting resistor for voltages above 3.3VDC.

Ethernet Interface

The FM-2000 contains a five-port, auto-sensing 10/100BaseT Ethernet switch with three of the ports available on the outside of the unit for field use. Because the switch is auto-sensing it is not necessary to utilize a crossover cable when connecting a PC directly to the FM-2000.

USB Interface

The front panel contains a standard USB port that is intended to be used with a local mouse when the operator prefers a mouse over the front panel’s integral touchpad. This USB port is not being serviced with software drivers for any other type of USB device that could be attached including, but not limited to, USB memory sticks, wireless network interfaces and the like. Attaching those devices to the FM-2000 may not damage them or the FM-2000; however they are not likely to operate.

Software Remote Control

Supplied Windows® Vorsis GUI software affords control of all system and processing parameters via a 10/100BaseT Ethernet interface. A front-panel color LCD screen and touchpad allows complete local control of all system and processing parameters if desired.