OPERATING MANUAL

OPTIMOD 5750

FM/HD/DAB+ Digital Audio Processor

orban

IMPORTANT NOTE: Refer to the unit for your Model and Serial number.

Model Number:
OPTIMOD 5750

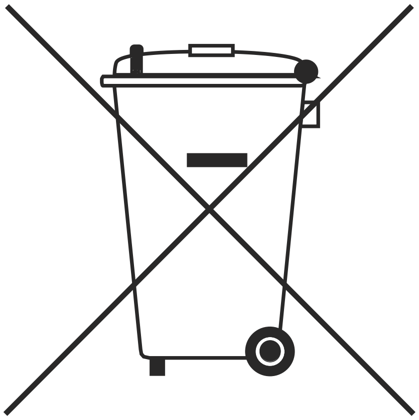
Description:
M/HD/DAB+ Digital Audio Processor OPTIMOD 5750, Stereo Encoder, Digital I/O, Digital MPX I/O, Protection Structure, Two-Band Structure, Multi-Band Structure, HD Radio™ / Digital Radio/Netcast Processing, Digital Composite Output, Dante AES67-Compliant Audio-Over-IP, µMPX streaming, Dual-Redundant Power Supply 90 V to 240V (automatically selected), switchable to 50µs or 75µs.

CAUTION: TO REDUCE THE RISK OF ELECTRICAL SHOCK, DO NOT REMOVE COVER (OR BACK). NO USER SERVICEABLE PARTS INSIDE. REFER SERVICING TO QUALIFIED SERVICE PERSONNEL.

WARNING: TO REDUCE THE RISK OF FIRE OR ELECTRICAL SHOCK,
DO NOT EXPOSE THIS APPLIANCE TO RAIN OR MOISTURE.

This symbol, wherever it appears, alerts you to the presence of uninsulated dangerous voltage inside the enclosure — voltage that may be sufficient to constitute a risk of shock.

This symbol, wherever it appears, alerts you to important operating and maintenance instructions in the accompanying literature. Read the manual.



In accordance to the WEEE (waste electrical and electronic equipment) directive of the European Parliament, this product must not be discarded into the municipal waste stream in any of the Member States. This product may be sent back to your Orban dealer at end of life where it will be reused or recycled at no cost to you.

If this product is discarded into an approved municipal WEEE collection site or turned over to an approved WEEE recycler at end of life, your Orban dealer must be notified and supplied with model, serial number and the name and location of site/facility.

Please contact your Orban dealer for further assistance.

www.orban.com

TO PREVENT ELECTRICAL SHOCK, DO NOT REMOVE COVER

NO USER SERVICABLE PARTS INSIDE.

REFER SERVICING TO QUALIFIED SERVICE PRESEONNEL.

PROTECT AGHAINST HUMIDITY

Published November 2023.

IMPORTANT SAFETY INSTRUCTIONS

All the safety and operating instructions should be read before the appliance is operated.

Retain Instructions: The safety and operation instructions should be retained for future reference.

Heed Warnings: All warnings on the appliance and in the operating instructions should be adhered to.

Follow Instructions: All operation and user instructions should be followed.

Water and Moisture: The appliance should not be used near water (e.g., near a bathtub, washbowl, kitchen sink, laundry tub, in a wet basement, or near a swimming pool, etc.).

Ventilation: The appliance should be situated so that its location or position does not interfere with its proper ventilation. For example, the appliance should not be situated on a bed, sofa, rug, or similar surface that may block the ventilation openings; or, placed in a built-in installation, such as a bookcase or cabinet that may impede the flow of air through the ventilation openings.

Heat: The appliance should be situated away from heat sources such as radiators, heat registers, stoves, or other appliances (including amplifiers) that produce heat.

Power Sources: The appliance should be connected to a power supply only of the type described in the operating instructions or as marked on the appliance.

Grounding or Polarization: Precautions should be taken so that the grounding or polarization means of an appliance is not defeated.

Power-Cord Protection: Power-supply cords should be routed so that they are not likely to be walked on or pinched by items placed upon or against them, paying particular attention to cords at plugs, convenience receptacles, and the point where they exit from the appliance.

Cleaning: The appliance should be cleaned only as recommended by the manufacturer.

Non-Use Periods: The power cord of the appliance should be unplugged from the outlet when left unused for a long period of time.

Object and Liquid Entry: Care should be taken so that objects do not fall and liquids are not spilled into the enclosure through openings.

Damage Requiring Service: The appliance should be serviced by qualified service personnel when: The power supply cord or the plug has been damaged; or Objects have fallen, or liquid has been spilled into the appliance; or The appliance has been exposed to rain; or The appliance does not appear to operate normally or exhibits a marked change in performance; or The appliance has been dropped, or the enclosure damaged.

Servicing: The user should not attempt to service the appliance beyond that described in the operating instructions. All other servicing should be referred to qualified service personnel.

The Appliance should be used only with a cart or stand that is recommended by the manufacturer.

Safety Instructions (European)

Notice For U.K. Customers If Your Unit Is Equipped With A Power Cord.

WARNING: THIS APPLIANCE MUST BE EARTHED.

The cores in the mains lead are coloured in accordance with the following code:

GREEN and YELLOW - Earth BLUE - Neutral BROWN - Live

As colours of the cores in the mains lead of this appliance may not correspond with the coloured markings identifying the terminals in your plug, proceed as follows:

The core which is coloured green and yellow must be connected to the terminal in the plug marked with the letter E, or with the earth symbol, or coloured green, or green and yellow.

The core which is coloured blue must be connected to the terminal marked N or coloured black.

The core which is coloured brown must be connected to the terminal marked L or coloured red.

The power cord is terminated in a CEE7/7 plug (Continental Europe). The green/yellow wire is connected directly to the unit's chassis. If you need to change the plug and if you are qualified to do so, refer to the table below.

WARNING: If the ground is defeated, certain fault conditions in the unit or in the system to which it is connected can result in full line voltage between chassis and earth ground. Severe injury or death can then result if the chassis and earth ground are touched simultaneously.

Conductor		WIRE COLOR	
		Normal	Alt
L	LIVE	BROWN	BLACK
N	NEUTRAL	BLUE	WHITE
E	EARTH GND	GREEN-YELLOW	GREEN

AC Power Cord Color Coding

PLEASE READ BEFORE PROCEEDING!

Manual

The Operating Manual contains instructions to verify the proper operation of this unit and initialization of certain options. You will find these operations are most conveniently performed on the bench before you install the unit in the rack. Please review the Manual, especially the installation section, before unpacking the unit. Trial Period Precautions If your unit has been provided on a trial basis: You should observe the following precautions to avoid reconditioning charges in case you later wish to return the unit to your dealer.

1) Note the packing technique and save all packing materials. It is not wise to ship in other than the factory carton. (Replacements cost $50.00).

2) Avoid scratching the paint or plating. Set the unit on soft, clean surfaces.

3) Do not cut the grounding pin from the line cord.

4) Use care and proper tools in removing and tightening screws to avoid burring the heads.

5) Use the nylon-washered rack screws supplied, if possible, to avoid damaging the panel. Support the unit when tightening the screws so that the threads do not scrape the paint inside the slotted holes.

Packing

When you pack the unit for shipping:

1) Tighten all screws on any barrier strip(s) so the screws do not fall out from vibration.

2) Wrap the unit in its original plastic bag to avoid abrading the paint.

3) Seal the inner and outer cartons with tape. If you are returning the unit permanently (for credit), be sure to enclose:
   - The Manual(s)
   - The Registration / Warranty Card
   - The Line Cord
   - All Miscellaneous Hardware (including the Rack Screws and Keys)
   - The Extender Card (if applicable)
   - The Monitor Rolloff Filter(s) (OPTIMOD-AM only)
   - The COAX Connecting Cable (THE OPTIMOD-5750 and OPTIMOD-TV only) Your dealer may charge you for any missing items. If you are returning a unit for repair, do not enclose any of the above items. Further advice on proper packing and shipping is included in the Manual (see Table of Contents).

Trouble

If you have problems with installation or operation:

(1) Check everything you have done so far against the instructions in the Manual. The information contained therein is based on our years of experience with OPTIMOD and broadcast stations.

(2) Check the other sections of the Manual (consult the Table of Contents and Index) to see if there might be some suggestions regarding your problem.

(3) After reading the section on Factory Assistance, you may call Orban Customer Service for advice during normal business hours. The number is +1 856.719.9900.

WARNING

This equipment generates, uses, and can radiate radio-frequency energy. If it is not installed and used as directed by this manual, it may cause interference to radio communication. This equipment complies with the limits for a Class A computing device, as specified by FCC Rules, Part 15, subject J, which are designed to provide reasonable protection against such interference when this type of equipment is operated in a commercial environment. Operation of this equipment in a residential area is likely to cause interference. If it does, the user will be required to eliminate the interference at the user's expense.

WARNING

This digital apparatus does not exceed the Class A limits for radio noise emissions from digital apparatus set out in the radio Interference Regulations of the Canadian Department of Communications. (Le present appareil numerique n'emet pas de bruits radioelectriques depassant les limites applicables aux appareils numeriques [de las class A] prescrites dans le Reglement sur le brouillage radioelectrique edicte par le ministere des Communications du Canada.)

IMPORTANT

Perform the installation under static control conditions. Simply walking across a rug can generate a static charge of 20,000 volts. This is the spark or shock you may have felt when touching a doorknob or some other conductive surface. A much smaller static discharge is likely to destroy one or more of the CMOS semiconductors employed in OPTIMOD-FM. Static damage will not be covered under warranty.

There are many common sources of static. Most involve some type of friction between two dissimilar materials. Some examples are combing your hair, sliding across a seat cover or rolling a cart across the floor. Since the threshold of human perception for a static discharge is 3000 volts, you will not even notice many damaging discharges.

Basic damage prevention consists of minimizing generation, discharging any accumulated static charge on your body or workstation, and preventing that discharge from being sent to or through an electronic component. You should use a static grounding strap (grounded through a protective resistor) and a static safe workbench with a conductive surface. This will prevent any buildup of damaging static.

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Table of Contents

Section 1: Introduction

The OPTIMOD 5750 is a 1 RU (rack unit) broadcast audio processor designed for use with analog FM broadcasts as well as digital broadcasts or streaming. It has a consistent 2 band AGC (Automatic Gain Control) followed by a 5 band processor and state of the art peak control. A separate peak control system in employed for those using the processing for streaming or codec based processing such as HD or DAB+.

The OPTIMOD 5750 is a result of multiple generations of broadcast audio processing experience combined with the latest technology to bring the end user a powerful tool for polishing their on air product. Take a little time now to familiarize yourself with the OPTIMOD 5750. A small investment of your time now will yield large dividends in audio quality.

Features of the OPTIMOD 5750 include:

- Six Processing structures to meet all format requirements from Classical to Contemporary and everything in between.
- Advanced, intelligent and consistent 2 band AGC with windowing, dual mode leveling and bass/master controls to eliminate pumping.
- A 5 band dynamics processor for unparalleled consistency with both music and voice whether sourced locally or from a network or syndicated program.
- OPTIMOD's exclusive MX peak control technology for transient punch and airy highs.
- An RDS/RBDS on-board encoder that supports dynamic PS scrolling and IP access.
- A full complement of factory tuned presets by professionals with the most experience in the industry. All factory presets can be adjusted using Orban's unique "Less/More" control which is an industry standard as a guide to dial in a unique signature sound. Of course, more intrigate controls are also available to experienced users for fine tuning the sound of any preset.
- AES67/SMPTE ST-2110 - Two redundant network interfaces are available for Audio-Over-IP connections supporting AES67, RAVENNA™ and SMPTE ST-2110. AES67 provides both DANTE and Livewire+ ™ compatibility.
- The 5750 can be controlled and configured via any HTML5 web browser. It also supports the SNMP v2 and the Ember+ protocols.
- The HTML5 web browser user interface offers a complete tool set to monitor and measure your signal, including an oscilloscope and FFT displays.
- Two internal and selectable Neilsen and Kantar watermarking encoder are available, allowing the FM and HD/DAB+ processing to be independently watermarked.

- The FM and HD/DAB+ processed signals can be monitored remotely via IP, allowing for processing adjustments where a clean over-the-air signal isn't available.
- Optional µMPX (micro MPX) allows transmission of DMPX over IP.
- High resolution front panel touch screen for ease of navigation.

Input/output Configuration

THE OPTIMOD-5750 will simultaneously accommodate:

- Digital AES3 left/right inputs.
- Two Digital AES3 outputs, both of which can be switched independently to carry the following signals: FM analog processed without diversity delay, FM analog processed with diversity delay, digital radio processed without delay, digital radio processed with delay, or low delay monitor.
- Digital AES11 sync reference input.
- Analog left/right inputs.
- Analog left/right outputs, which can be switched independently to carry the following signals: FM analog processed without diversity delay, FM analog processed with diversity delay, digital radio processed without delay, digital radio processed with delay, or low delay monitor.
- Analog and AES3 composite stereo outputs.
- Subcarrier (SCA and RDS/RBDS) input.
- Dante dual-redundant audio-over-IP left/right inputs and outputs, 44.1 or 48 kHz.

Digital AES3 Left/Right Input/Outputs

The digital input and outputs conform to the professional AES3 standard. They all have sample rate converters to allow operation at 32 kHz, 44.1 kHz, 48 kHz, 88.2 kHz, and 96 kHz sample frequency. For best peak control, operate at 44.1 kHz or higher.

The left/right digital input is on one XLR-type female connector on the rear panel; the left/right digital outputs are on two XLR-type male connectors on the rear panel. THE OPTIMOD-5750 simultaneously accommodates digital and analog inputs and outputs.

You can switch any of the 5750's outputs between the analog-channel processing, a low-delay monitor signal, and the HD-channel processing. You select whether THE OPTIMOD-5750 uses the digital or analog input on the Input/output screen, by PC remote control, or by GPI (General Purpose Interface) optically-isolated remote control.

Both analog and digital outputs are active continuously. Level control of the AES3 input is via software control through the INPUT/OUTPUT screens. In addition, an AES11 sync input can accommodate house sync. It will lock the 5750's two AES3 outputs to this sync even if the digital input is asynchronous to house sync. In Ratings Encoder Loop-Through mode, AES3 output #2 drives the ratings encoder and the OPTIMOD's Sync Input is repurposed to receive the output of the ratings encoder. When not being used to receive the output of a ratings encoder, the sync input can be used to receive audio as a silence-sense fallback source.

Analog Left/Right Input/Output

The left and right analog inputs are on XLR-type female connectors on the rear panel. Input impedance is greater than 10kΩ; balanced and floating. Inputs can accommodate up to +27 dBu (0 dBu = 0.775Vrms). The left and right analog outputs are on XLR-type male connectors on the rear panel. Output impedance is 50Ω; balanced and floating. They can drive 600Ω or higher impedances, balanced or unbalanced. The peak output level is adjustable from –6 dBu to +24 dBu.

Level control of the analog inputs and outputs is a accomplished via software control through System Setup.

Stereo Baseband Composite Output

The stereo encoder has two unbalanced analog baseband outputs on two BNC connectors on the rear panel. Each output can be strapped for 0 or 75Ω source impedance and can drive up to 8V peak-to-peak into 75Ω in parallel with up to 0.047μF (100ft/30m of RG-59/U cable) before any significant audible performance degradation occurs (see the footnote on page 1-16 and refer to Figure 2-3: Separation vs. load capacitance on page 2-10). Independent level control of each output is via software in the INPUT/OUTPUT > COMPOSITE screen.

Digital Composite Output

The 5750 provides an AES3 digital composite output at 192 kHz sample rate. Its output level is set in the Input/Output 5 screen titled Composite 2. At the PC remote, it is located in the I/O SETUP SCREEN with the tab DIGITAL ENCODER. This output appears on a male XLR-type connector on the rear panel. This output is fully compatible with and interoperable with the de-facto industry standard digital connection being implemented by transmitter manufacturers and others.

Subcarriers

The stereo encoder has two unbalanced 600Ω subcarrier (SCA) inputs with rear-panel BNC connectors to accept any subcarrier at or above 23 kHz. The subcarriers are mixed into each composite output and their level is not affected by the composite level control for that output.

The 5750 does not digitize subcarriers appearing at these two inputs; the mixing occurs after D/A conversion and is analog.

Subcarrier inputs sum into the composite baseband outputs before the digitally controlled composite attenuators. The sensitivity of the both SCA inputs are variable from 220 mV p-p to >10 V p-p to produce 10% injection. Internal PC-board-mounted trim pots determine the sensitivity.

The correct peak level of the stereo program applied to the stereo encoder sometimes depends on the number of subcarriers in use. Some regulatory authorities require the total baseband peak modulation to be maintained within specified limits. Thus, the level of the stereo main and subchannel must be reduced when a subcarrier is turned on. The 5750's remote control feature allows you to reduce the stereo main and subchannel level by connecting an on/off signal from your subcarrier generator. You define the amount of reduction (in units of percent modulation) on the Input/output screen. See page 2-62 for information on programming the remote control. A jumper on the circuit board can reconfigure the SCA 2 input to provide the stereo pilot tone only, which can provide a pilot reference for an RDS subcarrier generator.

Digitized Subcarrier Inputs

Two additional subcarrier inputs are provided on the rear panel. These are digitized and are summed only into the digital composite output. Their mix level is set in the Input/Output 5 screen titled Composite 2. At the PC remote, they are located in the I/O SETUP SCREEN with the tab DIGITAL ENCODER. If you need to use both the digital and analog composite outputs, you must split the outputs of your SCA generators with Y cables so that each generator output drives one digitized SCA input and one non-digitized SCA input.

Wordclock/10 MHz Sync Reference Input

The sync reference input (labeled REF IN) appears on a female BNC jack on the rear panel. It is available only on MPX hardware. It accepts a 1x 5V p-p squarewave wordclock signal at 32, 44.1, 48, 88.2, or 96kHz, or a 10MHz sinewave or squarewave signal, 0.5 to 5V peak. 10MHz is a common output frequency produced by GPS and rubidium frequency standards. You can configure the 5750 to lock its 19kHz pilot tone and output sample frequency to this input. When a valid reference signal is applied to this input, it locks the OPTIMOD's Internal DSP clock to this input, so if a given AES3 L/R output's sync source is set to INTERNAL, the sample frequency at that output will be locked to this reference signal.

Do not apply an AES3 signal to this input.

Remote Control Interface

The Remote Control Interface is a set of eight optically isolated inputs on a DB-25 connector that can be activated by 5-12V DC. They can control various functions of the 5750.

Computer Interface

On the rear panel of the 5750 are serial ports and Ethernet ports. These computer interfaces support remote control and metering, downloading software upgrades, and audio-over-IP connectivity. Each 5750 can be accessed via a web browser using the IP address configured on the 5750.

- **100 Mbps Ethernet Port for Remote Connectivity:** This port will connect to any Ethernet-based network that supports the TCP/IP protocol.

- **Dante Audio-Over-IP Ethernet Ports:** These ports support dedicated Dante (100% AES 67-compatible) audio-over-IP network connection, main and backup.

Warranty, User Feedback

User Feedback: We are very interested in your comments about this product. The 5750 was developed to make adjustments to audio processing easy to make. We will carefully review your suggestions for improvements to either the product or the manual. Please email us at support@orban.com

LIMITED WARRANTY: [Valid only for products purchased and used in the United States]

Orban warrants Orban products against defects in material or workmanship for a period of five years from the date of original purchase for use, and agrees to repair or, at our option, replace any defective item without charge for either parts or labor.

IMPORTANT: This warranty does not cover damage resulting from accident, misuse or abuse, lack of reasonable care, the affixing of any attachment not provided with the product, loss of parts, or connecting the product to any but the specified receptacles. This warranty is void unless service or repairs are performed by an authorized service center. No responsibility is assumed for any special, incidental, or consequential damages. However, the limitation of any right or remedy shall not be effective where such is prohibited or restricted by law.

Simply take or ship your Orban products prepaid to our service department. Be sure to include a copy of your sales slip as proof of purchase date. We will not repair transit damage under the no-charge terms of this warranty. Orban will pay return shipping.

No other warranty, written or oral, is authorized for Orban Products.

This warranty gives you specific legal rights and you may have other rights that vary from state to state. Some states do not allow the exclusion of limitations of incidental or consequential damages or limitations on how long an implied warranty lasts, so the above exclusions and limitations may not apply to you.

INTERNATIONAL WARRANTY: Orban warrants Orban products against evident defects in material and workmanship for a period of five years from the date of original purchase for use. This warranty does not cover damage resulting from misuse or abuse, or lack of reasonable care, or inadequate repairs performed by unauthorized service centers.

Performance of repairs or replacements under this warranty is subject to submission of this Warranty/Registration Card, completed and signed by the dealer on the day of purchase, and the sales slip. Shipment of the defective item is for repair under this warranty will be at the customer’s own risk and expense. This warranty is valid for the original purchaser only.

Section 2: Installation

Unboxing the 5750

Once you have taken delivery of the 5750, you should open the box and check the unit for any damage from shipping. Inside the box should be the following:

- Operating Manual
- 2 Line Cords (domestic, European)
- 4 Rack-mounting screws, 10-32 x ¾—with washers, #10
- 1 Ethernet crossover cable
- 1 OPTIMOD 5750 Audio Processor

If any of these items are damaged or missing, please contact your dealer or Orban as soon as possible.

Save all packing materials! If you should ever have to ship the 5750 (e.g., for servicing), it is best to ship it in the original carton with its packing materials because both the carton and packing material have been carefully designed to protect the unit.

Grounding

Establish a low impedance common ground in the facility and try to route all equipment grounds to that point, using conductors with the largest possible surface area while keeping those leads as short as possible. The 5750's ground reference (its chassis) should be connected to the station/transmitter site ground. Such a connection is especially important when the 5750 is operated in a high RF environment because it helps minimize differential voltages between the processor's chassis and other pieces of equipment.

Surge Protection

Always place surge protection circuits as close as possible to the 5750. AC power line surges should be handled in a way that keeps instantaneous potential differences between the power line hot, neutral, AC grounding conductor, the station ground and the processor chassis as low as possible. Likewise, measures should also be taken to keep the instantaneous potential difference between the audio cable shields and the processor chassis as low as possible (this applies to all audio equipment, not just the 5750), particularly when the equipment is located within the electrically hostile environment of a station's transmitter facility.

UPS/Power Conditioning

Choose the best power conditioning/UPS units that your budget will allow, focusing on the most important features and options that you actually need. Some questions to ask while reviewing features are:

- How does the unit handle AC power that is not exactly 60Hz, such as when the facility is on its backup generator?
- If the unit has onboard surge protection, what kind of surge capability does it have and where are those surges directed to?
- Is the unit equipped with remote monitoring capability?
- Does it have onboard monitoring and alarms to signal a problem such as batteries with low reserve?

Where to locate the 5750

The best location for THE OPTIMOD-5750 is as close as possible to the transmitter, so that its stereo encoder output can be connected to the transmitter through a circuit path that introduces the least possible change in the shape of THE OPTIMOD-5750's carefully peak-limited composite waveform—a short length of coaxial cable. If this is impossible, the next best arrangement is to feed the 5750's AES3 digital output through an all-digital, uncompressed path to the transmitter's exciter, although this will preclude using the 5750's composite limiter.

Use the 5750's left and right analog audio outputs in situations where the stereo encoder and exciter are under the jurisdiction of an independent transmission authority and where the programming agency's jurisdiction ends at the interface between the audio facility and the link connecting the audio facility to the transmitter. (The link might be telephone / post lines, analog microwave radio, or various types of digital paths.) This situation is not ideal because artifacts that cannot be controlled by the audio processor can be introduced by the link to the transmitter, by transmitter peak limiters, or by the external stereo encoder.

If the transmitter is not accessible: All audio processing must be done at the studio and you must tolerate any damage that occurs later.

If you can obtain a broadband (0-75 kHz) phase-linear link to the transmitter and the transmitter authority will accept the delivery of a baseband encoded signal, use the 5750's internal stereo encoder at the studio location to feed the STL. Then feed the output of the STL receiver directly into the transmitter's exciter with no intervening processing.

If an uncompressed left/right digital link is available to the transmitter, this is also an excellent means of transmission, although it will not pass the effects of the 5750's composite processor (if you are using it). However, if the digital link employs lossy compression, it will degrade peak control. To prevent overshoots caused by spectral truncation in the link, set the 5750's output sample rate to 44.1 kHz or higher.

If only an audio link is available, use the 5750's left and right audio outputs and feed the audio, without pre-emphasis, directly into the link. If possible, request that any transmitter protection limiters be adjusted for minimum possible action— THE OPTIMOD-5750 does most of that work. Transmitter protection limiters should respond only to signals caused by faults or by spurious peaks introduced by imperfections in the link. To ensure maximum quality, all equipment in the signal path after the studio should be carefully aligned and qualified to meet the appropriate standards for bandwidth, distortion, group delay and gain stability and such equipment should be re-qualified at reasonable intervals.

If the transmitter is accessible: You can achieve the most accurate control of modulation peaks by locating THE OPTIMOD-5750 at the transmitter site and then using its stereo encoder to drive the transmitter. You can usually also obtain good results by locating THE OPTIMOD-5750 at the studio and connecting the baseband output of its stereo encoder to the transmitter through a composite baseband STL.

However, many analog composite baseband STLs do not control peaks perfectly and locating THE OPTIMOD-5750 at the transmitter site (where it can control peaks just prior to the transmitter's RF exciter) is thus likely to maximize loudness. The ideal link is an uncompressed digital composite STL because these have virtually flawless waveform fidelity and allow full use of the 5750's composite limiter.

Because THE OPTIMOD-5750 controls peaks, it is irrelevant whether the audio link feeding THE OPTIMOD-5750's input terminals is phase-linear. However, the link should have low noise, the flattest possible frequency response from 30-15,000Hz, and low nonlinear distortion.

We strongly recommend that you use the 5750's internal stereo encoder to feed the output of the encoder directly. You will achieve a louder sound on the air, with better control of peak modulation, than if you use current external stereo encoders.

The shorter the baseband cable from THE OPTIMOD-5750 to exciter, the less likely that ground loops or other noise problems will occur in the installation. If you require a long cable run, you can use a Jensen JT-123-BMCF transformer1 to break any ground loops. This transformer will usually cure even the most stubborn hum or noise caused by the composite connection between THE OPTIMOD-5750 and the exciter.

If a separate stereo encoder must be used, feed the encoder directly from the 5750's left and right analog outputs. If possible, bypass the pre-emphasis network and the input low-pass filters in the encoder so that they cannot introduce spurious peaks.

Because of their special design, THE OPTIMOD-5750's pre-emphasis network and low-pass filters perform the same functions while retaining tight peak control. Connect the composite output of the 5750 to the baseband input of the exciter through less than 100 feet (30 meters) of coaxial cable.

100 feet of coaxial cable (assuming 30-pF / foot capacitance) will reduce measured separation at 15 kHz (worst case) to approximately 60dB. This separation is comfortably above the separation (approximately 20dB) that starts to cause perceptible changes in the stereo image.

Studio-Transmitter Link

Transmission from Studio to Transmitter: There are five types of studio-transmitter links (STLs) in common use in broadcast service: uncompressed digital, digital with lossy compression (like MPEG, Dolby®, or APT-x®), microwave, analog landline (telephone / post line), and audio subcarrier on a video microwave STL.

STLs are used in three fundamentally different ways. They can either:

1) Pass unprocessed audio for application to the 5750's input.

2) Pass the 5750's peak controlled analog or digital left and right audio outputs.

3) Pass the 5750's peak-controlled composite stereo baseband output.

The three applications have different performance requirements. In general, a link that passes unprocessed audio should have very low noise and low nonlinear distortion, but its transient response is not important. A link that passes processed audio doesn't need as low a noise floor as a link passing unprocessed audio. However, its transient response is critical. At the current state of the art, an uncompressed digital link using digital inputs and outputs to pass audio in left/right format achieves best results. We will elaborate below.

Digital Links: Digital links may pass audio as straightforward PCM encoding or they may apply lossy data reduction processing to the signal to reduce the number of bits per second required for transmission through the digital link. Lossy data rate reduction will almost invariably distort peak levels and such links must therefore be carefully qualified before you use them to carry the peak-controlled output of the 5750 to the transmitter. For example, the MPEG Layer 2 algorithm can increase peak levels up to 4 dB at 160kB / sec by adding large amounts of quantization noise to the signal. While the desired program material may psychoacoustically mask this noise, it is nevertheless large enough to affect peak levels severely. For any lossy compression system the higher the data rate, the less the peak levels will be corrupted by added noise, so use the highest data rate practical in your system.

It is practical (though not ideal) to use lossy data reduction to pass unprocessed audio to the 5750's input. The data rate should be at least of "contribution quality"— the higher, the better. If any part of the studio chain is analog, we recommend using at least 20-bit A/D conversion before encoding.

Because the 5750 uses multiband limiting, it can dynamically change the frequency response of the channel. This can violate the psychoacoustic masking assumptions made in designing the lossy data reduction algorithm. Therefore, you need to leave "headroom" in the algorithm so that the 5750's multiband processing will not unmask quantization noise. This is also true of any lossy data reduction applied in the studio (such as hard disk digital delivery systems).

For MPEG Layer 2 encoding, we recommend 384kB / second or higher.

Some links may use straightforward PCM (pulse-code modulation) without lossy data reduction. If you connect to these through an AES3 digital interface, these can be very transparent provided they do not truncate the digital words produced by the devices driving their inputs. Because the 5750's output is tightly band-limited to 16.5kHz, it can be passed without significant overshoot by equally well by any link with 44.1kHz or higher sample frequency.

Currently available sample rate converters use phase-linear filters (which have constant group delay at all frequencies). If they do not remove spectral energy from the original signal, the sample rate conversion, whether upward or downward, will not add overshoot to the signal. This is not true of systems that are not strictly band-limited to 15 kHz, where downward sample rate conversion will remove spectral energy and will therefore introduce overshoot.

If the link does not have an AES3 input, you must drive its analog input from the 5750's analog output. This is less desirable because the link's analog input circuitry may not meet all requirements for passing processed audio without overshoot.

If you use a digital link to pass the digital composite output, the link must be uncompressed. We recommend not using sample rate conversion in such a link, as sample rate converters may introduce filters that compromise stereo separation.

Composite Baseband Microwave STLs (Analog and Digital): The composite baseband microwave STL carries the standard pilot-tone stereo baseband and therefore receives the output of a stereo encoder located at the studio site. The receiver output of the composite STL is the stereo baseband signal, which is applied directly to the wideband input of the FM broadcast transmitter's exciter. Thus, no stereo encoder is needed at the transmitter.

In general, a composite microwave STL provides good audio quality, as long as there is a line-of-sight transmission path from studio to transmitter of less than 10 miles (16 km). If not, RF signal-to-noise ratio, multipath distortion, and diffraction effects can cause serious quality problems. Where a composite STL is used, use the 5750's stereo encoder to drive the composite STL transmitter.

Uncompressed digital composite baseband microwave STLs, if properly designed, have excellent performance and we recommend them highly. They are particularly desirable in an 5750 installation because they allow you to use the 5750's composite limiter to increase on-air loudness.

However, the fact that they are digital does not eliminate the requirement that they have low frequency response that is less than 3 dB down at 0.15 Hz. Any such STL should be qualified to ensure that it meets this specification.

Dual Microwave STLs: Dual microwave STLs use two separate transmitters and receivers to pass the left and right channels in discrete form. Dual microwave STLs offer greater noise immunity than composite microwave STLs. However, problems include gain- and phase matching of the left and right channels, overloads induced by pre-emphasis, and requirements that the audio applied to the microwave transmitters be processed to prevent over-modulation of the microwave system.

Lack of transparency in the path will cause overshoot. Unless carefully designed, dual microwave STLs can introduce non-constant group delay in the audio spectrum, distorting peak levels when used to pass processed audio. Nevertheless, in a system using a microwave STL, the 5750 is sometimes located at the studio and any overshoots induced by the link are tolerated or removed by the transmitter's protection limiter (if any). The 5750 can only be located at the transmitter if the signal-to-noise ratio of the STL is good enough to pass unprocessed audio. The signal-to-noise ratio of the STL can be used optimally if an Orban OPTIMOD-PC 1101 or OPTIMOD 6300 protect the link from overload. These are the preferred choices because their AGCs are identical to the AGC in the 5750.

Some microwave links can be modified so that the deviation from linear phase is less than +10° from 20Hz to 15kHz and frequency response is less than 3dB down at 0.15Hz and less than 0.1dB down at 20kHz. This specification results in less than 1% overshoot with processed audio. Many such links have been designed to be easily configured at the factory for composite operation, where an entire FM stereo baseband is passed. The requirements for maintaining stereo separation in composite operation are similar to the requirements for high waveform fidelity with low overshoot. Therefore, most links have the potential for excellent waveform fidelity if they are configured for composite operation (even if a composite FM stereo signal is not actually being applied to the link).

Nevertheless, in a dual-microwave system, the 5750 is usually located at the main FM transmitter and is driven by the microwave receivers. One of Orban's studio level control systems, such as the OPTIMOD 6300, protects the microwave transmitters at the studio from overload. These units also perform the gain riding function ordinarily executed by the AGC section of the 5750's processing and optimize the signal-to-noise ratio obtainable from the dual-microwave link.

Analog Landline (PTT / Post Office Line): Analog landline quality is extremely variable, ranging from excellent to poor. Whether landlines should be used or not depends upon the quality of the lines locally available and upon the availability of other alternatives. Due to line equalizer characteristics and phase shifts, even the best landlines tend to veil audio quality slightly. They will certainly be the weakest link in a FM broadcast chain. Slight frequency response irregularities and non-constant group delay characteristics will alter the peak-to-average ratio and will thus reduce the effectiveness of any peak limiting performed prior to their inputs.

STL and Exciter Overshoot

Earlier in this section, we discussed at length what is required to prevent STLs from overshooting. There are similar requirements for FM exciters. Nevertheless, in some installations some overshoot is inevitable. If this is a problem in your installation, the 5750's remote control feature offers the means to reduce the peak level of the 5750's audio output as necessary.

This way, you can still use the 5750's line-up tone to adjust the steady-state deviation to 75kHz. Yet, the reduced peak level of the audio emitted from the 5750 ensures that the carrier deviates no further than 75 kHz after overshoot. This overshoot reduction can be selected on the INPUT/OUTPUT screen and the remote operation can be selected in SYSTEM SETUP > NETWORK / REMOTE.

Monitoring on Loudspeakers and Headphones

In live operations, highly processed audio often causes a problem with the DJ or presenter's headphones. When an "MX" preset is active, the delay through the 5750 can be as much as 270 milliseconds. This delay will be audible as a distinct echo that almost no one can tolerate in their headphones while speaking. The "Multipath Mitigator" phase skew corrector adds about 250 ms of delay. The 5750 offers two main solutions to this problem:

- Using a non-MX preset, which provides the same algorithms and performance as THE OPTIMOD-8500.
- Using the 5750's low-delay "monitor" output to drive talent headphones.

Using the monitor output is appropriate for both HD Radio and non-HD Radio operations, while using a non-MX preset on-air is only useful in non-HD Radio operations where no diversity delay has been applied to the analog FM transmission.

Because of their higher performance, we recommend using MX presets in HD Radio operations. In this context, the preset delay becomes part of the HD Radio diversity delay and the 5750 automatically compensates for the preset delay to ensure that the diversity delay always agrees with the value you specified when setting the 5750 up.

With non-MX presets, the normal delay through the 5750 (from input to FM outputs) is about 18 ms when HARD or MEDIUM bass clipping is selected, as it is in all nonMX factory presets other than those with "LL" ("low latency") or "UL" ("ultra-low latency") in their names. An 18 ms delay is workable for most talent (although it may require some acclimatization) because 18 ms is below the psychoacoustic "echo fusion threshold," which means that talent will not hear discrete slap echoes in their headphones. This means that they can monitor comfortably off-air without being distracted or confused. Moreover, off-air cueing of remote talent is routine.

Some talent moving from an analog processing chain will require a learning period to become accustomed to the voice coloration caused by "bone-conduction" comb filtering. This is caused by the delayed headphone sound's mixing with the live voice sound, which introduces notches in the spectrum that the talent hears when he or she talks. All digital processors induce this coloration to a greater or lesser extent. Fortunately, it does not cause confusion or hesitation in the talent's performance unless the delay is above the psychoacoustic "echo fusion" (Haas) threshold of approximately 20-25 ms, where the talent starts to hear slap echo in addition to frequency response colorations.

Two lower-delay options are available. "Low latency" reduces input-to-FM-output delay to 13 ms and "ultra-low latency" reduces delay to about 3.7 ms. The trade-off for this reduction is approximately 1 dB decrease in loudness compared to the 5750's full look-ahead processing for low latency and about 2.5 dB loudness decrease for ultra-low latency.

- When using a non-MX preset, you can invoke the low latency mode by setting the BASSCLIPMODE control (in the CLIPPERS page of ADVANCED CONTROL) to LLHARD. You can also recall a preset with "LL" as part of its name.

  LLHARD differs in two ways from the normal HARD mode of the bass clipper:

  1) LLHARD automatically defeats the compressor look-ahead. (This action is functionally equivalent to setting the LOOK-AHEAD control to OUT, except that it reduces input/output delay by 5 ms).

  2) LLHARD prevents the bass clipper from switching to Medium mode whenever speech is detected. By constraining the system in these ways, it ensures that the delay is always 13 ms.

Switching the BASSCLIPMODE to LLHARD (from any other mode) removes five milliseconds of delay from the signal path. If it occurs during program material, switching can cause audible clicks, pops, or thumps (due to waveform discontinuity). If you have some presets with LLHARD bass clipper mode and some without, switching between these presets is likely to cause clicks unless you do it during silence. However, these clicks will never cause modulation to exceed 100%.

One of the essential differences between the HARD and LLHARD bass clipper modes is that switching between HARD and MED does not change delay and is therefore less likely to cause audible clicks.

- Ultra-low latency processing uses a separate, parallel processing structure and is invoked by recalling any "UL" preset. This structure operates simultaneously with other code, so, unlike the similar structure in Orban's OPTIMOD 8300, it does not require a code re-load and does not cause a gap in programming.

  The only way to create an ultra-low latency user preset is to start with a "UL" factory preset and then edit that preset. "UL" user presets cannot be directly converted to low latency or optimum latency presets because the preset customization controls are different—UL presets have fewer available controls because of the difference in processing structure.

UL presets are the closest emulations of OPTIMOD 8200 processing available in the 5750. These presets differ from OPTIMOD 8200 processing in two main ways: (1) the 5750 UL presets still use the 5750's stereo enhancement, equalization section, advanced-technology AGC, composite limiter, and multiplex power controller, and (2) the 5750 UL presets use anti-aliased clippers operating at 256 kHz sample rate.

Some talent moving from an analog processing chain will require a learning period to become accustomed to the voice coloration caused by "bone-conduction" comb filtering. This is caused by the delayed headphone sound's mixing with the live voice sound and introducing notches in the spectrum that the talent hears when he or she talks. All digital processors induce this coloration to a greater or lesser extent.

Fortunately, it does not cause confusion or hesitation in the talent's performance unless the delay is above the psychoacoustic "echo fusion" (Haas) threshold of approximately 20-25 ms and the talent starts to hear slap echo in addition to frequency response colorations.

Low-Delay Monitoring: The 5750's analog outputs can be switched to provide a low-delay monitoring feed with a delay of 5-10 ms. This uses a separate instance of the Ultra-Low-Latency structure to allow the monitor to provide a "FM-processed" sound. You can adjust the amount of "FM peak limiter sound" via the MONITOR DRIVE control.

If the talent relies principally on headphones to determine whether the station is on the air, simple loss-of-carrier and loss-of-audio alarms should be added to the system. The 5750 can be interfaced to such alarms through any of its eight its GPI remote control inputs, cutting off the low-delay audio to the talent's phones when an audio or carrier failure occurs.

PC Control

The OPTIMOD 5750 can be controlled via a web browser pointed at the address assigned to it. All system controls are accessible there.

Racking the 5750

The 5750 is designed to fit in a standard 19' (inch) rack. While the 5750 is robust and can be used in situations that are extreme, it is best that 1 rack space be left free above and below the processor to allow air to circulate.

The 5750 requires three standard rack units (5 inches / 12.7 cm). There should be a good ground connection between the rack and the 5750 chassis—check this with an ohm meter to verify that the resistance is less than 0.5Ω. Mounting the unit over large heat-producing devices (such as a vacuum-tube power amplifier) may shorten component life and is not recommended. Ambient temperature should not exceed 45°C (113°F) when equipment is powered. Equipment life will be extended if the unit is mounted away from sources of vibration, such as large blowers and is operated as cool as possible.

Using the power cable supplied (or equivalent), the 5750 should be energized. It cannot be stressed enough that Orban Labs recommends the 5750 be energized via an Uninterruptable Power Supply (UPS) to prevent sudden swings in voltage that may damage equipment. This isn't required because of a design flaw in the 5750 itself, it is good engineering practice to ensure all mission critical equipment is protected. The 5750 input voltage is 100-240 VAC 50-60Hz, making the unit universal electrically.

Once the 5750 has booted, it's now time to apply audio to the processor. This portion of the manual will discuss the various input features available to you and your 5750.

Cable

Orban recommends using two-conductor foil-shielded cable (such as Belden 8451 or equivalent) for the audio input and output connections because signal current flows through the two conductors only. The shield does not carry signal and is used only for shielding.

Analog Audio Input

Nominal input level between –14dBu and +8dBu will result in normal operation of the 5750. Analog inputs require an XLR type MALE connector.

- (0dBu = 0.775Vrms. For this application, the dBm @600Ω scale on voltmeters can be read as if it were calibrated in dBu.)
- The peak input level that causes overload is +27.0dBu.
- The electronically balanced input uses an ultra-low noise and distortion differential amplifier for best common mode rejection and is compatible with most professional and semi-professional audio equipment, balanced or unbalanced, having a source impedance of 600Ω or less. The input is EMI suppressed.
- Input connections are the same whether the driving source is balanced or unbalanced.
- Connect the red (or white) wire to the pin on the XLR-type connector (#2 or #3) that is considered HIGH by the standards of your organization. Connect the black wire to the pin on the XLR-type connector (#3 or #2) that is considered LOW by the standards of your organization.
- In low RF fields (like a studio site not co-located with an RF transmitter), connect the cable shield at 6300 input only—it should not be connected at the source end. In high RF fields (like a transmitter site), also connect the shield to pin 1 of the male XLR-type connector at the 5750 input.
- If the output of the driving unit is unbalanced and does not have separate CHASSIS GROUND and (–) (or LOW) output terminals, connect both the shield and the black wire to the common (–) or ground terminal of the driving unit.

Analog Audio Output

- Electronically balanced and floating outputs simulate a true transformer output. The source impedance is 50Ω. The output is capable of driving loads of 600Ω or higher; the 100% modulation level is adjustable with the AO 100% control over a –6dBu to +24dBu range.

- If an unbalanced output is required (to drive unbalanced inputs of other equipment), it should be taken between pin 2 and pin 3 of the XLR-type connector.

  Connect the LOW pin of the XLR-type connector (#3 or #2, depending on your organization’s standards) to ground; take the HIGH output from the remaining pin. No special precautions are required even though one side of the output is grounded.

- Use two-conductor foil-shielded cable (Belden 8451, or equivalent).

- At the 5750’s output (and at the output of other equipment in the system), do not connect the cable’s shield to the CHASSIS GROUND terminal (pin 1) on the XLR-type connector. Instead, connect the shield to the chassis ground at the input destination. Connect the red (or white) wire to the pin on the XLR-type connector (#2 or #3) that is considered HIGH by the standards of your organization. Connect the black wire to the pin on the XLR-type connector (#3 or #2) that is considered LOW by the standards of your organization.

AES3 Digital Input/Output

There are two AES3 and two AES3 outputs.

- Digital 1 input accepts a standard AES signal and can operate at 32, 44.1, 48, 88.2 and 96 kHz.

- Digital 2 input also accepts the same standard AES signals as listed above, or can be switched to an AES sync signal.

- Digital 1 Output carries either the FM Left/Right pre or de-emphasized output or the HD Left/Right output.

- Digital 2 Output carries the same outputs as Digital 1 Output but adds the option for Digital MPX (DMPX)

- Per the AES3 standard, each digital input or output line carries both the left and right stereo channels. The connection is 110Ω balanced. The AES3 standard specifies a maximum cable length of 100 meters. While almost any balanced, shielded cable will work for relatively short runs (5 meters or less), longer runs require used of 110Ω balanced cable like Belden 1800B, 1801B (plenum rated), multi-pair 180xF, 185xF, or 78xxA. Single-pair category 5, 5e, and 6 Ethernet cable will also work well if you do not require shielding. (In most cases, the tight balance of Category 5/5e/6 cable makes shielding unnecessary.)

- The AES3id standard is best for very long cable runs (up to 1000 meters). This specifies 75Ω unbalanced coaxial cable, terminated in male BNC connectors. A 110Ω/75Ω balun transformer is required to interface an AES3id program input or output. Conversely, the wordclock / AES11id sync input is designed for 75Ω operation.

- The digital input clip level is fixed at 0 dB relative to the maximum digital word. The maximum digital input will make the 5750 input meters display 0dB. The reference level is adjustable using the DI REF control.

- The 5750 is a “multirate” system whose internal sample rate is 64 kHz and multiples thereof (up to 512 kHz). The outputs processed for analog FM are band-limitedto 16.5 kHz, with a stopband that begins at 18 kHz.

  Therefore, the output can be passed through a 44.1 kHz (or higher) uncompressed link without adding significant overshoot. Because sample rate conversion is ordinarily a phase-linear process that does not add bandwidth, the 5750’s output signal will continue to be compatible with 44.1 kHz links even if it undergoes intermediate sample rate conversions (for example, 44.1 kHz to 96 kHz to 44.1 kHz) at various points in the program chain.

- The audio bandwidth of the HD-processed signal is adjustable from 15 kHz to 20 kHz in 1 kHz steps.

Dante AES67

The following instructions describe how to set up your OPTIMOD’s Dante audio-over-IP (AOIP) connection. It is assumed that you have previously set up your Dante network according to Dante’s instructions and that you know how to configure and control the Dante network using Broadway’s Dante Controller application, which includes a thorough manual. For Dante documentation, please visit https://www.Broadway.com/

1) Using a normal (not crossover) Ethernet cable, connect your audio-over-IP network switch to your OPTIMOD’s rear-panel AUDIO NETWORK 1 connector.

   - For a redundant Dante network, Audio Network 1 and Audio Network 2 must work with the same speed. For example, one port with 100 Mbps and the other with 1 Gbps will not work.

   - Note that audio routing between two Dante-enabled units will only be possible if both are set to the same sample rate and sample rate pull-up/- down. Bit depth can be different.

The Dante network will automatically discover the OPTIMOD and configure its IP address. Then the OPTIMOD will appear in applications like Dante Controller.

2) A Dante-enabled device will advertise information about itself to other Dante devices and Dante Controller, including:

   - Device name

   - Audio Channel Labels

   - Number of Audio Channels

   - Sample Rates and Bit Depths

This information can be seen when viewing a device on Dante Controller and allows Dante devices to determine compatibility with other devices, such as compatible sample rates to allow audio to be routed.

Dante hardware devices, like your OPTIMOD, are set to obtain their IP address automatically from the network. They will either:

- Automatically assign themselves an address in the range 169.254.*.* (172.31.*.* for the secondary network if present), or

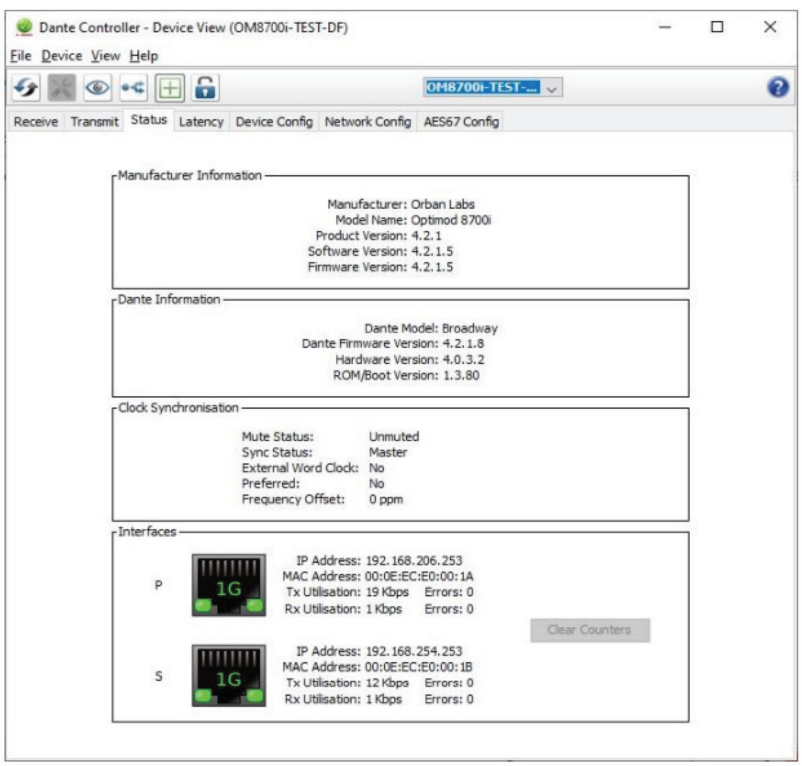
- Obtain an IP address from a DHCP server if it is present on the network

Your PC or Mac TCP/IP network configuration set should be set to "Obtain an IP address automatically." This way it will automatically acquire a Link Local automatic IP address in the same network as other Dante devices. If a DHCP server is present, the computer and Dante devices will all acquire their IP addresses via DHCP. Alternatively, you may assign static IP addresses for the primary and secondary networks via Dante Controller. If you do so, be sure to record them so you can connect to your network in the future.

The 5750 can be operated in REDUNDANT or SWITCHED modes:

- Redundant: When a device is set to REDUNDANT, the device will duplicate Dante audio traffic to both Ethernet ports, allowing the implementation of a redundant network via the secondary port.

- Switched: When a device is set to SWITCHED, the secondary Ethernet Port will behave a standard switch portal, allowing daisy-chaining through the 5750.



Figuer 2-1: A REDUNDANT setup with static IP addresses.

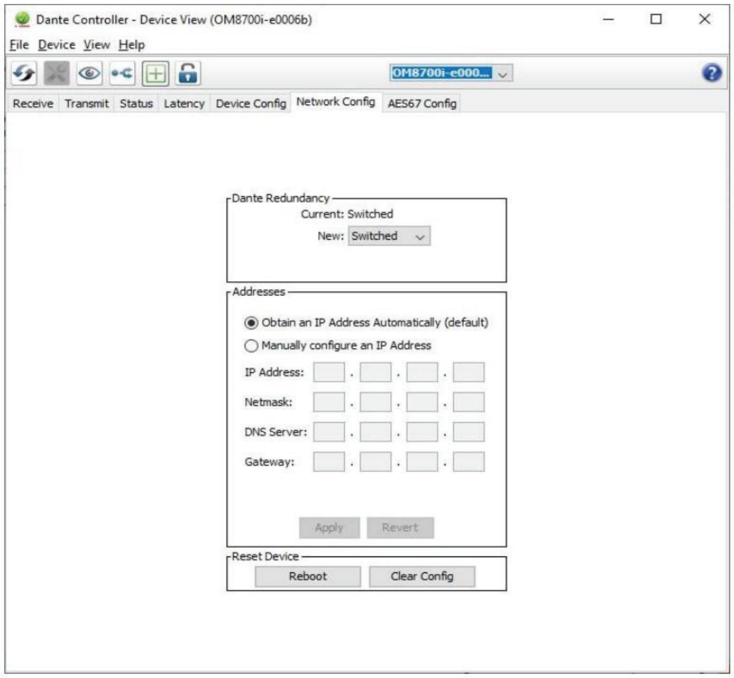


Figure 2-2: A SWITCHED setup with a DHCP-sourced IP address:

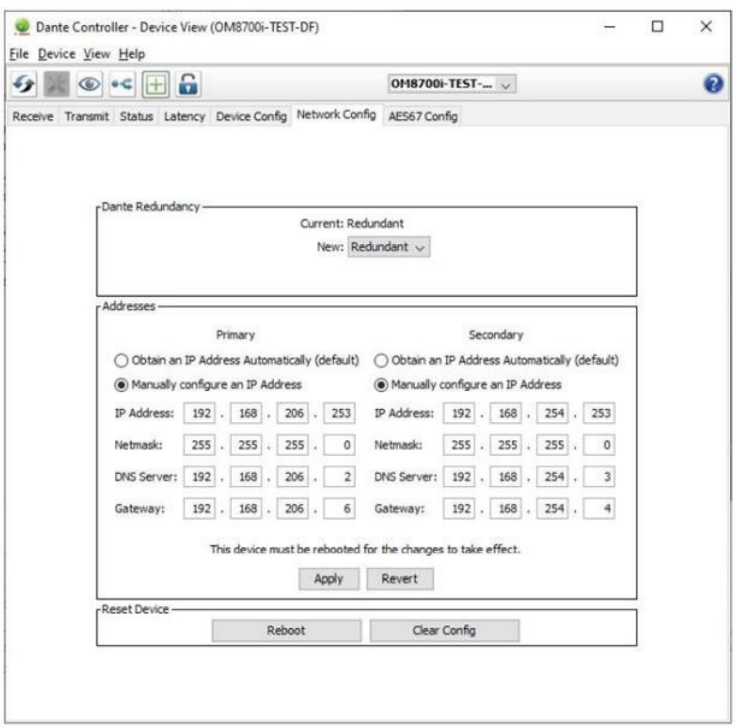
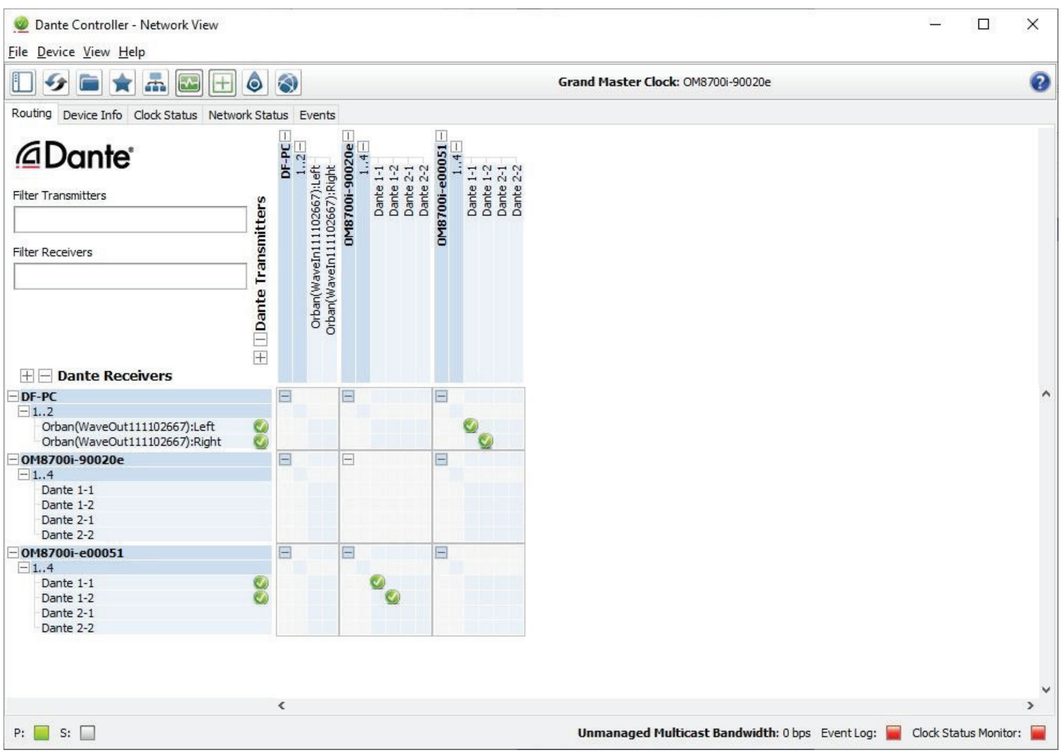


Figure 2-3: A Dante Controller reports the 5750 network status when a 5750 has been configured and connected for dual-redundant networks.

3) In the Dante network, route audio to and from the 5750.

A Dante device has a number of channels associated with it. These are either transmit (TX) or receive (RX) channels. Your OPTIMOD provides two Dante stereo transmit channels and two stereo receive channels. Receive channels and devices are listed down the left side of the grid in the Dante Controller software. Transmit channels and devices are listed along the top of the grid. Transmit channels are advertised on the network. A receiver uses this advertisement to establish a subscription to the channel. A transmit channel can be sent to multiple receivers using unicast or multicast. Receive channels are connected to transmit channels via a subscription. Each receive channel will receive audio over the network from at most one transmit channel. The 5750's Dante inputs are:

- Channel 1 = DANTE 1 LEFT INPUT
- Channel 2 = DANTE 1 RIGHT INPUT
- Channel 3 = DANTE 2 LEFT INPUT
- Channel 4 = DANTE 2 RIGHT INPUT



The 5750's Dante outputs are:

- Channel 1 = DANTE 1 LEFT OUTPUT
- Channel 2 = DANTE 1 RIGHT OUTPUT
- Channel 3 = DANTE 2 LEFT OUTPUT

- Channel 4 = DANTE 2 RIGHT OUTPUT

You can set the sample rate of the OPTIMOD's Dante output via the DEVICE CONFIG tab in the Dante Controller software. You must set the sample rate to match the rate of the Dante network device receiving the OPTIMOD's output.

4) Configure the 5750's Dante inputs and outputs.

Unlike the sample rate of the OPTIMOD's AES3 outputs, the OPTIMOD's GUI cannot control the sample rate of its Dante output. Instead. 44.1 kHz and 48 kHz are both suitable; choose the one that matches the setup of the rest of your transmission facility. Your OPTIMOD will automatically detect the network sample rate and configure its Dante inputs and outputs to match it, applying sample rate conversion.

a) Navigate to INPUT/OUTPUT > INPUT. Using the SET INPUT TO drop-down, choose DANTE 1 or DANTE 2, depending on how you set up the network in step 2 above.

b) Navigate to INPUT/OUTPUT > DANTE 1.

This screen is conceptually different from the other 5750 input/output screens because this screen contains both the Dante 1 input and output controls. Note that although they share a setup screen, the Dante 1 input and output streams are separate and distinct on the network. Note that PC Remote has separate tabs for DANTE INPUT and DANTE OUTPUT.

c) Set up the Dante 1 input controls:

- REFERENCE LEVEL VU / REFERENCE LEVEL PPM
  These two fields track each other with an offset of 7 dB. Adjustments in one field affect the other field.

- INPUT BALANCE
  This control trims the right channel gain. It is usually set to 0 dB.

d) Set up the DANTE 1 output controls:

- OUTPUT SOURCE
  Set to FM, FM+DELAY (includes HD Radio diversity delay), MONITOR, HD, OR HD+DELAY. Setting any OUT SOURCE to HD+DELAY automatically activates the DAB+ mode, delaying the HD output by the setting of the DAB+ DELAY control, which is located in the HD DIGITAL RADIO tab of PC Remote. This allows you to compensate for different delays in the links feeding the DAB+ transmitter and its associated FM transmitter (if used).

- OUTPUT LEVEL
  Sets the peak output level in units of dB below full scale.

- PRE-EMPHASIS
  This setting only applies if the output is set to FM or FM+DELAY.

It determines if the output is correctly pre-emphasized for FM transmission (PREEMPH) or if de-emphasis was applied after the OPTIMOD's peak limiter (FLAT). If it is FLAT, you must apply pre-emphasis in the transmitter.

- WORD LENGTH and DITHER
  The WORD LENGTH control sets the level of dither that the OPTIMOD applies to this output when you set DITHER to ON. Set the WORD LENGTH control to match the word length that digital input of your transmitter accepts.

e) Navigate to INPUT/OUTPUT > DANTE 2, and repeat steps (B) through (D) for this output.

f) Set up automatic fallback to the analog or digital input when DANTE 1 goes silent (optional):

- Navigate to INPUT/OUTPUT > SILENCE DETECT.
- Set the DANTE FALLBACK to DIGITAL or ANALOG if you wish the 5750 to switch automatically from Dante Input #1 to digital input or analog input respectively when silence is detected. Set the control to NO to defeat automatic switching.

Dante Firmware Updates

Dante chipset firmware updates require the Dante Firmware Updater, which can be downloaded here: https://www.Broadway.com/latest-firmware-update-manager

Dante Firmware Update Manager is a software application that allows you to:

- Select a firmware update file for a particular Dante module type
- Discover matching Dante-enabled audio devices on your network.
- Manage the firmware upgrade for these devices.
- Restore Dante-enabled modules in failsafe mode.

The firmware itself is not on Orban's FTP. Because your OPTIMOD's software was tested only with certain releases of Dante firmware, we suggest contacting Orban Customer Service before updating.

If you wish to update your 5750 with the Broadway Ultimo chip to support AES67, you may use Ultimo v4.1.2.5 firmware.

Composite Output and Subcarrier Input

There are two composite outputs. They carry the encoded stereo signal, the stereo pilot tone, and any subcarriers that may have been applied to the 5750's subcarrier inputs as well as the internal RDS generator.

These are unbalanced, with the shell connected directly to chassis/circuit ground.

Each output's level is independently adjustable from –12 dBu to +16.0 dBu.

The output impedance of composite 1 output and composite 2 output can be set to 0Ω or 75Ω via jumpers J7 and J8 respectively (located on the Composite/SCA daughterboard). As shipped, the link is on pins 3 and 4, yielding 0 Ω impedance. To reset a given output to 75Ω, place the link on pins 1 and 2 of its associated jumper. (See the schematic on page 6-38 and the parts locator diagram on page 6-34.

Each output can drive up to 75Ω in parallel with 0.047μF before performance deteriorates significantly.

Connect the 5750's composite output to the exciter input with up to 100 feet (30.5m) of RG-58/U or RG-59/U coaxial cable terminated in BNC connectors.

Longer runs of coax may increase problems with noise, hum, and RF pickup at the exciter. In general, the least troublesome installations place the 5750 close to the exciter and limit the length of the composite cable to less than 6 feet (1.8m).

- We do not recommend terminating the exciter input by 50Ω or 75Ω unless this is unavoidable. The frequencies in the stereo baseband are low by comparison to RF and video, and the characteristic impedance of coaxial cable is not constant at very low frequencies. Therefore, the transmission system will usually have more accurate amplitude and phase response (and thus, better stereo separation) if the coax is driven by a very low impedance source and is terminated by greater than 1kΩ at the exciter end. This also eases thermal stresses on the output amplifier in the stereo encoder, and can thus extend equipment life.

- Ground loops can occur if your exciter's composite input is unbalanced, although you can usually configure system grounding to break them (for example, by connecting the 5750's and exciter's power cords to adjacent sockets on an AC power strip). In difficult cases, you can always break a ground loop by using a Jensen JT-123-BMCF transformer.

- Even when its composite limiter is being used heavily, the 5750 will always protect the stereo pilot tone by at least 60 dB (±250Hz from 19 kHz) and will protect the region from 55 kHz to 100 kHz by at least 75 dB (re: 100% modulation.)

The subcarrier (SCA) inputs are provided for convenience in summing subcarriers into the baseband prior to their presentation to the FM exciter.

- The subcarrier inputs will accept any subcarrier (or combinations of subcarriers) above 23 kHz. Below 5 kHz, sensitivity rolls off at 6 dB/octave to suppress hum that might otherwise be introduced into the subcarrier inputs, which are unbalanced.

- The subcarrier inputs are mixed into the 5750's composite output in the analog domain, after D/A conversion of the 5750 stereo encoder's output. Rear-panel accessible PC-board-mounted trim pots allow the user to adjust the sensitivities of the two SCA inputs from <100 mV p-p to >10 V p-p to produce 10% injection with respect to 100% modulation = 4 V p-p at the 5750's composite outputs. (The factory setting is 4 V p-p to produce 10% injection.)

As shipped from the factory, the second SCA connector emits a stereo pilot tone reference for RDS or RBDS subcarrier generators. If you wish to reconfigure it to accept an SCA signal, move the link on jumper J6 (on the Composite/SCA daughter-board) from pins 3 and 4 to pins 1 and 2.

- To access J6, remove the 5750's top cover according to the instructions in step 1 on page 4-2. The schematic showing J6 is on page XXX.

Connect your subcarrier generator(s) to the 5750's subcarrier input(s) with coaxial cable terminated with BNC connectors.

- The subcarrier inputs have greater than 600Ω load impedance and are unbalanced. The sensitivity of both inputs is user-adjustable from <100 mV p-p to >10 V p-p to produce 10% injection with respect to 100% modulation = 4V p-p at the 5750's composite outputs. (The factory setting is 4 V p-p to produce 10% injection.)

- VR1 and VR2 on the Composite/SCA daughterboard set the sensitivity of SCA1 IN and SCA2 IN respectively and are accessible on the rear panel, You can use the 19 kHz reference control in the setup to determine whether the 19 kHz pilot reference output will be in-phase (0 DEG) with the pilot tone present in the composite output or will lead it by 90 degrees (90 DEG). 0 DEG is correct for most installations. Use 90 DEG only if your RDS/RBDS generator's 19 kHz reference input specifically requires this phase relationship.

DMPX

When Digital Output 2 is set for DMPX, the digital output XLR jack on the rear of the 5750 carries an AES over MPX digital composite signal that is compatible with exciters designed to ingest a digital composite signal. The benefits of using the DMPX signal is the elimination of the D-A and A-D converters found in the analog composite output.

WARNING: You should check with your exciter manufacturer to make sure your equipment has the capability to ingest the DMPX standard before using this option. Interfacing the 5750 with DMPX into a non-compatible exciter will cause unwanted results.

µMPX Composite over IP (optional)

The optional µMPX mode of composite transport is a specialized codec that can carry the composite payload (including RDS) over an internet connection of at least 320kbps. This is another option to placing the 5750 at the studio. While revolutionary in theory, there are some tradeoffs with µMPX such as an elevated noise floor. In most cases, this won't be an issue as the noise floor remains below acceptable standards.

Wordclock/10 MHz Sync Reference

The sync reference input accepts a 1x 5V p-p squarewave wordclock signal or a10 MHz sinewave or squarewave signal, 0.5 to 5 V peak. A menu item allows you to synchronize the output sample frequency to the frequency present at the sync. The connector is a female BNC with the shell grounded to chassis.

To permit daisy-chaining sync signals, the input impedance is greater than 1 KΩ. If the 5500 is the last device driven by the sync coaxial cable, you should terminate it by using a BNC Tee connector and a 75Ω BNC terminator. This will prevent performance-degrading reflections in the cable. This is required for both wordclock and AES11id operation.

WARNING: Do NOT apply an AES3 or AES3id signal to this input. Doing so will eventually cause your OPTIMOD to suffer a "communications board error."

Audio Grounding

Very often, grounding is approached in a "hit or miss" manner. However, with care it is possible to wire an audio studio so that it provides maximum protection from power faults and is free from ground loops (which induce hum and can cause oscillation).

In an ideal system:

- All units in the system should have balanced inputs. In a modern system with low output impedances and high input impedances, a balanced input will provide common-mode rejection and prevent ground loops, regardless of whether it is driven from a balanced or unbalanced source.
- The 5750 has balanced inputs. Its subcarrier inputs are unbalanced, but frequency response is rolled off at low frequencies to reject hum.
- All equipment circuit grounds must be connected to each other; all equipment chassis grounds must be connected together.
- In a low RF field, cable shields should be connected at one end only — preferably the source (output) end.
- In a high RF field, audio cable shields should be connected to a solid earth ground at both ends to achieve best shielding against RFI.
- Whenever coaxial cable is used, shields are automatically grounded at both ends through the terminating BNC connectors.

Audio Circuit Grounds

To maintain the same potential in all equipment, the circuit (audio) grounds must be connected together:

- When the 5750's stereo encoder is driving an unbalanced exciter input, you may encounter a ground loop. (Some older exciters have unbalanced inputs.) Unlike some older Orban FM processors, the 5750 does not have a ground lift switch. If you cannot reconfigure your grounding scheme to eliminate such a loop, you can balance and float the exciter input with a Jensen JT-123-BMCF transformer.

- In high RF fields, the system is usually grounded through the equipment rack in which the 5750 is mounted. The rack should be connected to a solid earth ground by a wide copper strap. Wire is completely ineffective at VHF because of the wire's self-inductance.

Optically Isolated Remote Control Connections

These are terminated in a type DB-25 male connector located on the rear panel. It is wired according to the diagram below on the following page.

To select the desired function, apply a 5-12V AC or DC pulse between the appropriate Remote Interface terminals. The (–) terminals can be connected together and then connected to power common at pin 1 to create a Remote Common. A current limited +12VDC source is available on pin 25. If you use 48V, connect a 2kΩ ±10%, 2- watt carbon composition resistor in series with the Remote Common or the (+) terminal to provide current limiting.

In a high-RF environment, these wires should be short and should be run through foil-shielded cable, with the shield connected to CHASSIS GROUND at both ends.

PIN ASSIGNMENT

1.	DIGITAL GOUND	
2.	REMOTE	1+
3.	REMOTE	2+
4.	REMOTE	3+
5.	REMOTE	4+
6.	REMOTE	5+
7.	REMOTE	6+
8.	REMOTE	7+
9.	REMOTE	8+
10.	TALLY	1
11.	TALLY	2
12.	N/C	
13.	ANALOG GROUND	
14.	REMOTE	1-
15.	REMOTE	2-
16.	REMOTE	3-
17.	REMOTE	4-
18.	REMOTE	5-
19.	REMOTE	6-
20.	REMOTE	7-
21.	REMOTE	8-
22-24.	N/C	
25.	+12 VOLTS DC	

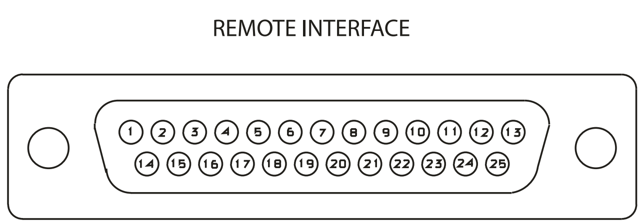
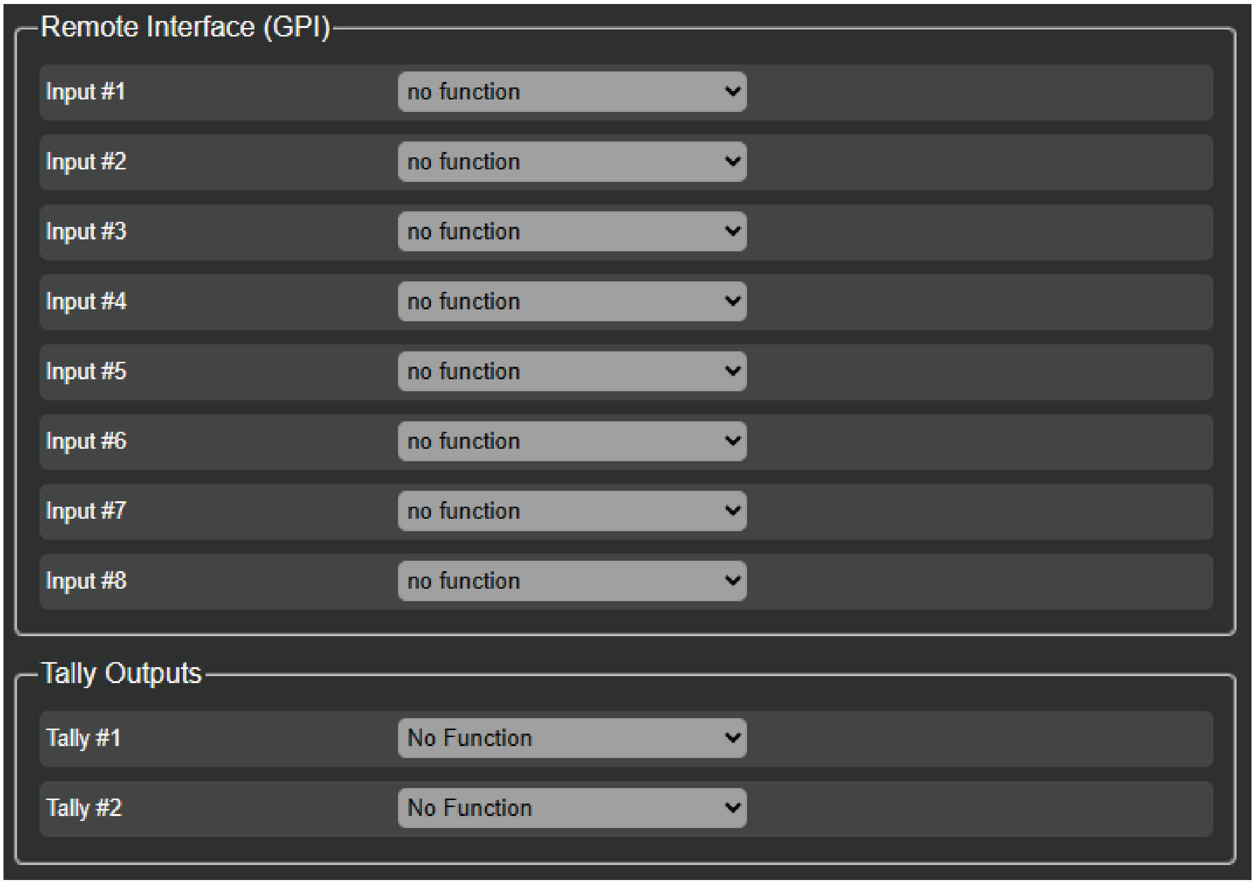


Figure 2-1: Wiring the 25-pin Remote Interface Connector



There are two tally outputs, which are NPN open-collector and operate with respect to pin 1 (common). Therefore, the voltage applied to the load (such as a relay or optoisolator) must be positive. You can use the 12 VDC source on pin 25 to drive the high side of the load, taking into account the fact that the voltage on pin 25 is current limited by a 310Ω resistor.

The tally outputs are protected against reverse polarity.

- To avoid damaging the 5750, limit the current into a tally output to 30 mA. DO NOT connect a tally output directly to a low impedance voltage source! The tally outputs are not protected against this abuse and Q3 or Q4 is likely to burn out.

Note that the tally outputs have no special RFI protection. Therefore, it is wise to use shielded cable to make connections to them.

You can program the two tally outputs to indicate a number of different operational and fault conditions.

1) Navigate to the I/O Settings and then REMOTE INTERFACE

   - Program the tallys for what you need to switch with the contact closure

- The default for the tally control is NO FUNCTION.

THE 5750 FRONT PANEL

The 5750 Front Panel was designed for easy navigation through the various menus via touchscreen. Status lights and a USB port are also incorporated

STATUS LIGHTS

There are four status lights that, at a glance, can tell you the current operating state of the 5750. They are:

- Power Supply 1 (PS1): When green, the power supply is operating as intended. If flashing red, there is either no power coming into the supply or there is an issue with the supply.
- Power Supply 2 (PS2): Same functionality as PS1.
- System (SYS): Under normal conditions, this will not be lit. If it is lit, there is some issue with the processor which should be addressed with Orban technical support. Please do not disassemble the unit.
- Temperature (TEMP): When illuminated, there is an overheating issue with the processor. You should attempt to remedy the temperature issue in the room, or in the rack where the 5750 is located. To prevent damage to the unit, if you cannot remedy the temperature situation, you should remove power to the unit so permanent damage does not occur.

USB PORT

Unused at this time.

TOUCHSCREEN DISPLAY

The heart of the front panel is the navigable touchscreen. As you walk thru the menus of the touchscreen, you will find that it is easy to make your way to the options you want to adjust.

The default display is the metering for the FM processing. In the upper right hand corner you will see the following information:

- Active preset
- Active input source
- Date and Time

Below this information are three buttons.

- Main Menu: Will step you into the processing menus
- Headphone Volume: Will open a menu that will show the adjustable volume level and a button to choose the source feeding the headphones. The options are:

  A) FM

B) FM + Delay

C) Monitor

D) HD

E) HD + Delay

F) Analog In

G) Digital In 1

H) Digital in 2

I) AoIP In 1

J) AoIP In 2

K) Stream In

Use the up and down buttons to scroll thru the options and touch the source you would like to monitor.

For convenience (and your ears), the headphone LEVEL button is in the top right of your screen to jump back to the level control to adjust. In the lower right corner is the BACK button which takes you back to the main screen.

The final button is the next button “>” which changes the meter display to show the HD/DAB metering. When in that mode, the “>” button becomes “<” to switch back to the FM metering.

MAIN MENU

The third and final button on the meter screen takes you to the MAIN MENU. This is where you can setup your 5750 and check on the status of the hardware.

INFO

Info is the diagnostic screens for the 5750. The following information can be found under the info tab

- Version
- Serial Number
- MAC Address
- IP, Subnet and Gateway addresses
- Hostname
- Temperature
- Fan Speed

- Supply Voltages

System status and any failures

SETUP

There are four options with submenus under setup.

- DISPLAY BACKLIGHT – Sets the brightness of the display.

- NETWORK SETTINGS – Allows you to set the IP, Subnet and Gateway addresses for the processor (or choose DHCP), and allows you to set port forwarding under REMOTE SETTINGS. STREAMING AUDIO is not yet available in this version of the 5750.

- DATE/TIME – Allows you to set the time to an NTP server or manually set the time.

   A) If you are using an NTP server, touch "NTP" and the button will change from blue to green. You can then select a server from the list of NTP servers by touching that button..

   B) When setting the TIME ZONE, first choose the continent, then your region. For example, if you choose AMERICAS you can choose Chicago and you're set. However, if you are in Argentina, the third column will display localities in Argentina for you to select.

- ACCESS CONTROL – Allows you to restrict parts of the processing to those you don't want to access or adjust.

The MAIN MENU button takes you back to the main menu.

RECALL

Allows you to pick a preset to start with. You can use the up/down dobbies to scroll thru the menu, then touch the LISTEN/SELECT button.

If you would like to compare presets, you may do so by scrolling to another preset and touching COMPARE. Every time you touch COMPARE, it will toggle between the original preset and the new one you have selected.

Presets may be grouped by FACTORY, USER and FAVORITES for easier scrolling.

MODIFY

Unused at this time. Please connect to the IP address of the processor with a browser to modify a preset.

CONNECTING TO THE 5750 via WEB INTERFACE

The default address for the 5750 is 192.168.254.254 with a subnet of 255.255.255.0 and the gateway address 0.0.0.0. It is preferred that you use a static IP address as opposed to HDCP. You may assign a static IP address of your own using standard network protocols. Please use a switch between the PC and 5750 when connecting to set up a proper network.

The 5750 has a default USER NAME and PASSWORD that you will be prompted to input upon connection to continue. They are:

USERNAME: ADMIN (all caps)

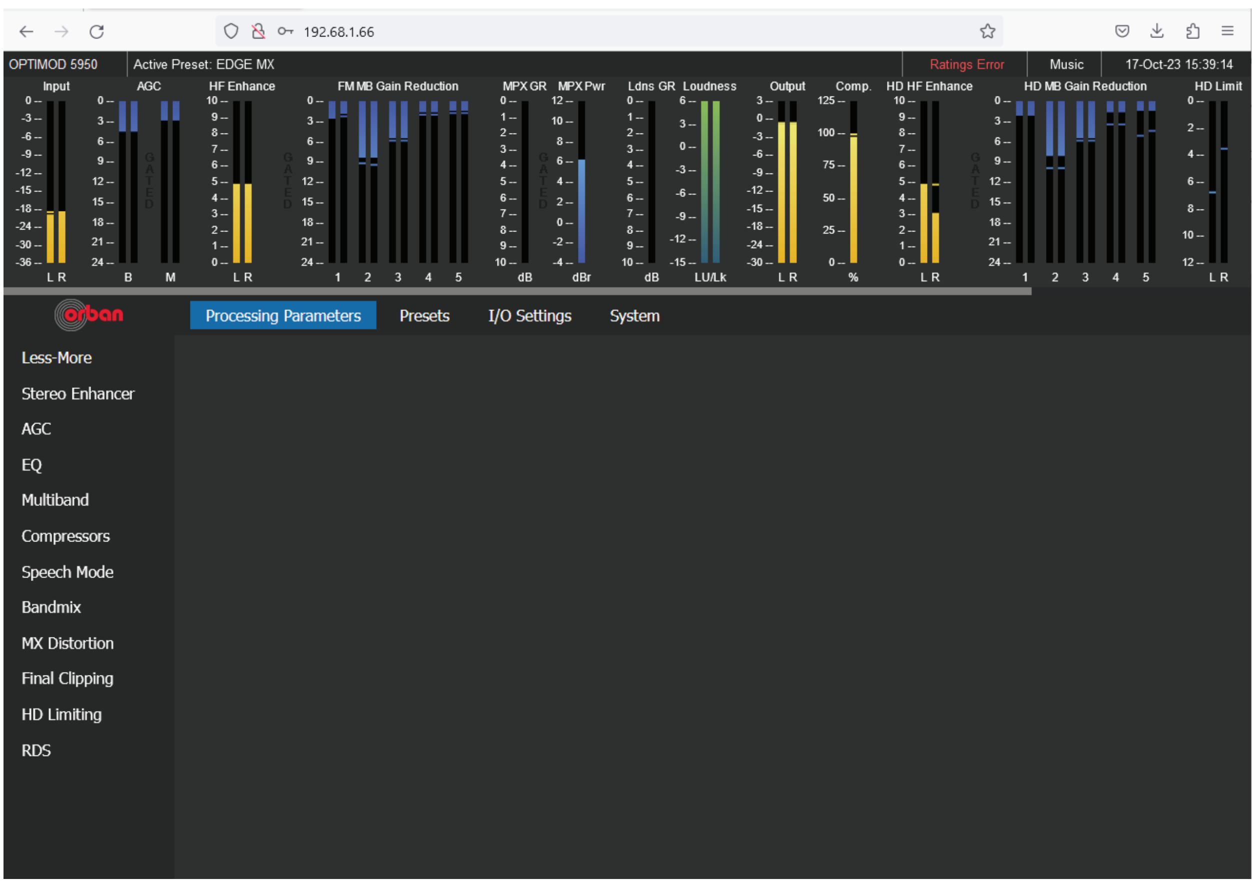
PASSWORD: 1234

To change the default username and password:

From the main panel, select SETUP then ACCESS CONTROL. You will see the default ADMIN account that you logged in to. From here you can ADD an account, or DELETE the existing ADMIN account so the processor can be accessed by any computer on the network.

CONNECTING WITH A BROWSER

- Open a browser and point it to the IP address you set for the 5750.
- If configured properly you will see the browser connect along with metering and control options.



Section 3: Operation

Introduction to Processing

Some Audio Processing Concepts

Reducing the peak-to-average ratio of the audio increases loudness. If peaks are reduced, the average level can be increased within the permitted modulation limits. The effectiveness with which this can be accomplished without introducing objectionable side effects (such as pumping or intermodulation distortion) is the single best measure of audio processing effectiveness.

Compression reduces the difference in level between the soft and loud sounds to make more efficient use of permitted peak level limits, resulting in a subjective increase in the loudness of soft sounds. It cannot make loud sounds seem louder. Compression reduces dynamic range relatively slowly in a manner similar to riding the gain: Limiting and clipping, on the other hand, reduce the short-term peak-toaverage ratio of the audio.

Limiting increases audio density. Increasing density can make loud sounds seem louder, but can also result in an unattractive busier, flatter, or denser sound. It is important to be aware of the many negative subjective side effects of excessive density when setting controls that affect the density of the processed sound.

Clipping sharp peaks does not produce any audible side effects when done moderately. Excessive clipping will be perceived as audible distortion.

Look-ahead limiting is limiting that prevents overshoots by examining a few milliseconds of the unprocessed sound before it is limited. This way the limiter can anticipate peaks that are coming up.

The 5750 uses look-ahead techniques in several parts of the processing to minimize overshoot for a given level of processing artifacts, among other things.

It is important to minimize audible peak-limiter-induced distortion when one is driving a low bitrate codec because one does not want to waste precious bits encoding the distortion. Look-ahead limiting can achieve this goal; hard clipping cannot.

One can model any peak limiter as a multiplier that multiplies its input signal by a gain control signal. This is a form of amplitude modulation. Amplitude modulation produces sidebands around the “carrier” signal. In a peak limiter, each Fourier component of the input signal is a separate “carrier” and the peak limiting process produces modulation sidebands around each Fourier component.

Considered from this perspective, a hard clipper has a wideband gain control signal and thus introduces sidebands that are far removed in frequency from their associated Fourier “carriers.” Hence, the “carriers” have little ability to mask the resulting sidebands psychoacoustically. Conversely, a look-ahead limiter’s gain control signal has a much lower bandwidth and produces modulation sidebands that are less likely to be audible.

Simple wideband look-ahead limiting can still produce audible intermodulation distortion between heavy bass and midrange material. The lookahead limiter in your OPTIMOD uses sophisticated techniques to reduce such IM distortion without compromising loudness capability.

Distortion in Processing

In a competently designed processor, distortion occurs only when the processor is controlling peaks to prevent the audio from exceeding the peak modulation limits of the transmission channel. The less peak control that occurs, the less likely that the listener will hear distortion. However, to reduce the amount of peak control, you must decrease the drive level to the peak limiter, which causes the average level (and thus, the loudness) to decrease proportionally.

Loudness and Distortion

In FM processing, there is a direct trade-off between loudness, brightness, and distortion. You can improve one only at the expense of one or both of the others. Thanks to Orban's psychoacoustically optimized designs, this is less true of Orban processors than of others.

In the 5750, the tradeoff between brightness and the other two parameters has been considerably improved (by 2.5 – 3 dB above 6 kHz) when an "MX" preset is active compared to when an "8500-style" preset is active. Nevertheless, all intelligent processor designers must acknowledge and work within the laws of physics as they apply to this trade-off.

Perhaps the most difficult part of adjusting a processor is determining the best trade-off for a given situation. We feel that it is usually wiser to give up ultimate loudness to achieve low distortion. A listener can compensate for loudness by simply adjusting the volume control. However, there is nothing the listener can do to make an excessively compressed or peak-limited signal sound clean again.

If processing for high quality is done carefully, the sound will also be excellent on small radios. Although such a signal might fall slightly short of ultimate loudness, it will tend to compensate with an openness, depth, and punch (even on small radios) that cannot be obtained when the signal is excessively squashed.

OPTIMOD 5750—from Bach to Rock

You can adjust the OPTIMOD-5750 so that the output sounds:

- As close as possible to the input at all times (using the Two-Band structure), or
- open but more uniform in frequency balance (and often more dramatic) than the input (using the Five-Band structure with slow release times), or
- dense, quite squashed, and very loud (using the Five-Band structure with fast or medium-fast release times).

The dense, loud setup will make the audio seem to jump out of car and table radios, but may be fatiguing and invite tune-outs on higher quality home receivers. The loudness/distortion trade-off explained above applies to any of these setups.

You will achieve best results if Engineering, Programming, and Management go out of their way to communicate and cooperate with each other. It is important that Engineering understand the sound that Programming desires, and that Management fully understands the trade-offs involved in optimizing one parameter (such as loudness) at the expense of others (such as distortion or excessive density).

Never lose sight of the fact that, while the listener can easily control loudness, he or she cannot make a distorted signal clean again. If such excessive processing is permitted to audibly degrade the sound of the original program material, the signal is irrevocably contaminated and the original quality can never be recovered.

Fundamental Requirements: High-Quality Source Material and Accurate Monitoring

A major potential cause of distortion is excess peak limiting. Another cause is poor quality source material, including the effects of the station's playback machines, electronics, and studio-to-transmitter link. If the source material is even slightly distorted, that distortion can be greatly exaggerated by the OPTIMOD-5750—particularly if a large amount of gain reduction is used. Very clean audio can be processed harder without producing objectionable distortion. A high-quality monitor system is essential. To modify your air sound effectively, you must be able to hear the results of your adjustments. In too many stations, the best monitor is significantly inferior to the receivers found in many listeners' homes!

Unfortunately, many contemporary CDs are mastered using levels of audio processing formerly used only by "aggressively-processed" radio stations. These CDs are audibly distorted (sometimes blatantly so) before any further OPTIMOD processing. The result of 5750 processing can be to exaggerate this distortion and make these recordings noticeably unpleasant to listen to over the air. There is a myth in the record industry that applying "radio-style" processing to CDs in mastering will cause them to be louder or will reduce the audible effects of on-air processing. In fact, the opposite is true: these CDs will not be louder on air, but they will be audibly distorted and unpleasant to listen to, lacking punch and clarity.

Another unfortunate trend is the tendency to put so much high frequency energy on the CDs that this energy cannot possibly survive the FM pre-emphasis / de-emphasis process. Although the 5750 loses less high frequency energy than many previous Orban processors (due to improvements in high frequency limiting and clipping technology), it is nevertheless no match for CDs that are mastered so bright that they will curl the vinyl off car dashboards. We hope that the record industry will come to its senses when it hears the consequences of these practices on the air.

If the waveforms on a given CD are noticeably clipped, it may be possible to improve the sound by using de-clipping software, which attempts to reconstruct the clipped-off sections of the waveform by extrapolating the clipped-off part of the waveform from audio that surrounds it. Beyond this, our best advice regarding 5750 processing is to use slow multiband release times and considerable band 4 to band 5 coupling, which will not further exaggerate distortion already on the CD. As of this writing, two audio restoration programs that offer de-clipping are Diamond Cut DC8 and iZotope Rx.

About the 5750's Signal Processing Features

Dual-Mono Architecture

The 5750 implements full dual-mono architecture in both the AGC and the multiband compressor sections. You can couple each band in both the AGC and multiband compressors to a variable extent—anywhere from perfect stereo coupling to completely uncoupled operation. The coupling control determines the maximum amount of gain imbalance permitted between the left and right channels in a given band, and therefore the amount of stereo image shift permitted in each frequency band.

Although the processing is dual-mono, you cannot adjust setup controls independently on the left and right channels—we assumed that the 5750 would always process stereo program material.

Signal Flow

The signal flows through the 5750 through the following blocks:

- Input Conditioning, including sample rate conversion, defeatable 30 Hz highpass filtering, and defeatable phase rotation
- "Multipath Mitigator" phase corrector
- Stereo Enhancement
- Two-Band Gated AGC, with target-zone window gating and silence gating
- Equalization, including high-frequency enhancement and Subharmonic synthesizer
- Multiband Compression with embedded HF clipping and additional HF limiter
- "Intelligent" Clipping with distortion control, distortion cancellation, and anti-aliasing
- Overshoot Compensation
- DSP-derived Stereo Encoder (generator)
- Composite Level Control Processor

Input Conditioning: The 5750 operates at a 64 kHz sample rate and power-of-two multiples thereof (up to 512 kHz in the stereo encoder). This allows user-selectable bandwidths from 15 to 20 kHz at the HD output.

The 15 kHz lowpass filtering in the analog processing's peak limiting section has a stopband that begins at 17 kHz. This provides the necessary ±2 kHz protection for RDS/RBDS subcarriers as well generous protection of the 19 kHz pilot tone.

The 5750's output spectral control is immaculate, ensuring maximum stereo and RDS coverage.

Because there is very little energy above 16 kHz, the 5750's digital output will pass through any uncompressed digital STL without adding noticeable overshoot and without the need for distortion-producing overshoot compensation schemes.

A defeatable 30 Hz 18 dB/octave highpass filter and a defeatable phase rotator complete the input-conditioning block. These have both been features in Orban FM processors for many years. Most users will defeat the 30 Hz filter and leave the phase rotator in-circuit, although the choice is always yours.

Stereo Enhancement: The 5750 provides two different stereo enhancement algorithms. The first is based on Orban's patented analog 222 Stereo Enhancer, which increases the energy in the stereo difference signal (L–R) whenever a transient is detected in the stereo sum signal (L+R). By operating only on transients, the 222 increases width, brightness, and punch without unnaturally increasing reverb (which is usually predominantly in the L–R channel).

Gating circuitry detects "mono" material with slight channel or phase imbalances and suppresses enhancement so this built-in imbalance is not exaggerated. It also allows you to set a "width limit" to prevent over-enhancement of material with significant stereo content, and will always limit the ratio of L–R / L+R to unity or less.

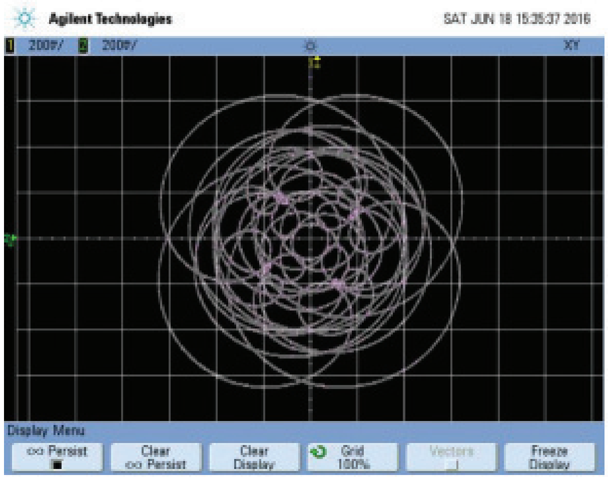
The second stereo enhancement algorithm is based on the well-known "Max" technique. This passes the L–R signal through a delay line and adds this decorrelated signal to the unenhanced L–R signal. Gating circuitry similar to that used in the "222- style" algorithm prevents over-enhancement and undesired enhancement on slightly unbalanced mono material.

"Multipath Mitigator" Left/Right Phase Skew Correction

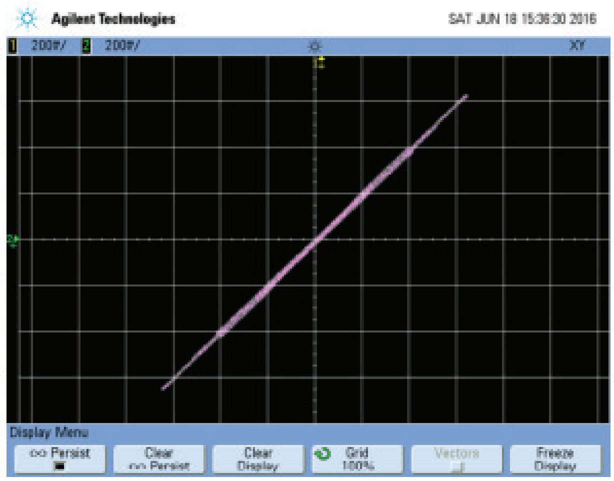
The phase skew corrector maximizes the quality of a mono mixdown or blend that might occur in a receiver. At higher frequencies (where audible comb filtering of the mono sum is most likely to occur), the corrector removes phase differences between the left and right channels, converting the HF signal into "intensity stereo" while preserving phase differences at lower frequencies where these differences are important for psychoacoustic "envelopment." The PHASE CORRECTOR CROSSOVER control in the EQ tab of the active Processing Preset sets the crossover frequency above which phase correction occurs.

By removing phase shifts between the left and right channels, the process minimizes the amount of energy in the stereo subchannel, which consequently minimizes multipath distortion without compromising stereo separation. It can allow more stereo enhancement to occur for a given amount of multipath distortion. The process also minimizes the amount of peak overshoot during SSB/VSB operation of the stereo encoder, thus minimizing the amount of composite limiting needed to constrain peak modulation to 100%.

This process can not only correct problems due to phase skew between the left and right channels of an analog recording due to head gap misalignment, it can also correct comb filtering caused by spaced microphones feeding the left and right channels, which can occur on drum kits and other sources that have been multi-miced.



10-Tone Lissajous Pattern,
250-9250 Hz, 90 degree phase difference



10-Tone Lissajous Pattern,
Processed by Phase Corrector

Because the process can subtly alter the stereo spatial effect, it may be inappropriate of "audiophile" formats, although its advantages in reducing multipath distortion are likely to be far more subjectively important. It can be smoothly activated and defeated via a delay-matched crossfade, so it is practical to do live switching between a preset with the process active and one where it is inactive.

Because it adds about 250 ms of delay, the phase skew corrector can be bypassed completely in Setup. If you are not using it and do not need to activate it smoothly "on-air," bypass it.

Figure 3-1 shows a 10-tone test stereo waveform with a 90 degree phase difference between each tone in the left and right channels. Power (RMS) in the two channels is the same. Frequencies range from 250 to 9250 Hz in 1 kHz increments. The 90- degree phase shift produces a different differential time delay for each tone: Each time the frequency is halved, the delay doubles. Figure 3-2 shows the effect of the phase correction: all 10 tones are now in-phase.

Two-Band Gated AGC: The AGC is a two-band device, using Orban's patented "master/ bass" band coupling. It has an additional important feature: target-zone gating. If the input program material's level falls within a user-settable window (typically 3 dB), then the release time slows to a user-determined level. It can be slow enough (0.5 dB/second) to effectively freeze the operation of the AGC. This prevents the AGC from applying additional, audible gain control to material that is already well controlled. It also lets you run the AGC with fast release times without adding excessive density to material that is already dense.

The AGC contains a compression ratio control that allows you to vary to ratio between 2:1 and essentially ∞:1. Lower ratios can make gain riding subtler on critical formats like classical and jazz.

The AGC has its own silence-gating detector whose threshold can be set independently of the silence gating applied to the multiband compressor.

Equalization: The 5750 has steep-slope bass shelving equalizer and three bands of fully parametric bell-shaped EQ.

You can set the slope of the bass shelving EQ to 6, 12, or 18 dB/octave and adjust the shelving frequency.

The 5750's bass, midrange, and high frequency parametric equalizers have curves that were modeled on the curves of Orban's classic analog parametrics (like the 622B), using a sophisticated, proprietary optimization program. The curves are matched to better than 0.15 dB. This means that their sound is very close to the sound of an Orban analog parametric. They also use very high quality filter algorithms to ensure low noise and distortion.

The 5750 HF Enhancer is a program-controlled HF shelving equalizer. It intelligently and continuously analyzes the ratio between broadband and HF energy in the input program material and can equalize excessively dull material without over-enhancing bright material. It interacts synergistically with the five-band compressor to produce sound that is bright and present without being excessively shrill.

Subharmonic Synthesizer: The subharmonic synthesizer generates subharmonics of fundamental frequencies in the 50-90 Hz range or 60-120 Hz range depending on the setting of the SUBHARMONIC CUTOFF FREQ control. The subharmonics are one octave below the frequencies from which they are generated (i.e., 25-45 Hz or 30-60 Hz) and track the levels of their generating frequencies.

If input program material below 45 or 60 Hz is present, the subharmonic synthesizer automatically reduces the level of the synthesized subharmonics to prevent excess build-up of energy below 45 or 60 Hz.

To prevent introducing unnatural coloration in male speech. the subharmonic synthesizer is defeated when the automatic speech/music detector detects speech. This is particularly important when the SUBHARMONIC CUTOFF FREQ control is set to 120 HZ.

Multiband Compression: The multiband compressor/limiter can be operated in fiveband or two-band mode. The 5750 controls high frequencies with distortioncanceled clipping and, in all but 5-band MX presets, with a high frequency limiter as well. The clipper operates at 256 kHz-sample rate and is fully anti-aliased.

Usually, the gain reduction in band 5 is slaved to the gain reduction in band 4 (as determined by the setting of the B4 > B5 COUPLE control); these bands are only independent from the viewpoint of the downward expander and multiband clippers. However, a high frequency limiter causes additional gain reduction in band 5 when band 5 multiband clipping alone would be insufficient to prevent HF distortion. The HF limiter uses a sophisticated analysis of the signal conditions in the 5750's clipping system to do this.

Except in MX presets, a clipper, embedded in the crossover, protects bands 1 and 2 from transient overshoot. This clipper has a shape control, allowing you to vary the "knee" of its input/output transfer curve from hard (0) to soft (10). Instead of a clipper, MX presets use a sophisticated bass pre-limiter located immediately before the system's main distortion-controlled clipper. In non-MX presets, the multiband compressor/limiter offers look-ahead compression to minimize overshoot and its associated clipping distortion. This look-ahead functionality can be turned on or off manually, or the 5750's speech/music detector can activate it automatically.

The Ultra-low Latency structure does not offer compressor look-ahead.

"Intelligent" Clipping: In non–MX presets, the 5750 prevents excess clipping distortion by dynamically reducing the drive level to the clippers as required, using an intelligent analysis of the clipping distortion produced in the final clipper and overshoot compensator.

To minimize latency, the Ultra-low Latency structure does not have this feature. This is the principal reason why it achieves less on-air loudness that the optimum-latency and low-latency processing for a given amount of distortion.

MX presets use an advanced peak limiting structure that uses additional "intelligence," novel processing structures, and other strategies to produce lower perceived distortion with "difficult" program material. This peak limiter also offers substantially improved transient impact and high frequency power handling capability compared to the "intelligent" clipping in the non–MX presets.

For more information on the MX clipper technology, see Appendix D.

Speech Mode: You can set many of the processing parameters separately for speech signals, as detected by the 5750's speech/music detector. This allows you to tune the processing separately for speech and music.

A SPEECH DETECT control allows you to force the 5750 into Music mode, overriding the Speech/Music detector. This control is contained in the processing preset. In fiveband presets, it is found in the Speech Mode screen (Advanced Modify 6) and in two-band presets, it is found in the Two Band screen (Advanced Modify 4).

Note that the speech detector will detect most speech mixed with music as "music" unless the music is at a very low level compared to the speech. Speech must also be centered in the stereo sound field to be detected as "speech."

DSP-derived Stereo Encoder: The 5750's stereo encoder is derived from algorithms first developed for the high-performance Orban 8218 stand-alone encoder. The 5750's stereo encoder operates at 512 kHz-sample rate to ease the performance requirements of the D/A converter's reconstruction filter, making it possible to achieve excellent stereo separation that is stable over time and temperature. DSP-based group delay and magnitude equalizers for the entire composite analog path further improve separation.

The 5750 has two independent composite outputs, whose levels are both software settable. For convenience, two SCA inputs sum into the 5750's analog composite output amplifier. The second SCA input can be configured to provide a 19kHz reference output for subcarrier generators that need it. See page 6-2.

The 5750 does not digitize SCAs.

SSB Stereo Encoder Operation: The 5750 allows its stereo encoder's stereo subchannel modulator to operate in an experimental compatible single sideband/vestigial sideband mode. SSB/VSB operation suppresses the upper sideband of the stereo subcarrier above 38,150 Hz, which reduces the occupied bandwidth of the FM-modulated RF signal. In SSB mode, the subchannel modulator acts as a pure SSB generator for L–R material in the frequency range of 150 Hz to 17 kHz and as a vestigial sideband generator below 150 Hz.

In normal operation, the stereo subchannel modulator produces a double sideband suppressed carrier signal with pairs of mirror image sidebands around 38 kHz. With respect to an L+R gain of 1, the gain of each sideband is 0.5. In SSB/VSB mode, the upper sideband is suppressed by at least 80 dB above a modulating frequency of 150 Hz and the gain of the lower sideband is 1.0. Below 150 Hz, the sum of the gains of the sideband pairs is 1.0. (The conventional DSB case is a limiting case of this, where the gains of the upper and lower sidebands are both 0.5 and sum to 1.) This "summation to 1" criterion is necessary to achieve compatibility with normal FM radios that use synchronous demodulation of the stereo subchannel. Almost every radio manufactured since 1973 works like this. We have verified that the 5750's SSB generator produces more than 60 dB of separation from 50 to 15,000 Hz when measured on a Belar FMSA-1 "Wizard" modulation monitor, which was originally designed for convention double sideband operation.

However, there are consumer-based radios and car audio systems which detect the missing upper sideband and misinterpret this as a signal issue and blend towards mono. Other systems will develop distortion because of the increased amplitude in the lower sideband which is needed to allow the system to work at all. This distortion manifests itself as a headroom issue. Before applying the system, engineering, programming and management should be aware of the tradeoffs using SSB/VSB mode vs DSB operation.

In SSB/VSB mode, the bandwidth of the 5750's composite output signal extends to 38,150 Hz when the 5750's composite limiter is not used. When the composite limiter is used, the limiting action will produce energy up to 55 kHz (as it does with normal DSB operation) but this energy will be much lower in level than the energy that would have been produced by normal DSB operation in the frequency range occupied by the upper sideband.

SSB operation causes irreducible, "laws of physics" composite peak modulation overshoots to occur with certain combinations of left and right channel signals that are independently peak limited to 100% modulation, which is the correct limiting technique for conventional double-sideband transmission. The worst-case irreducible SSB overshoot occurs when the left and right channels contain correlated signals whose phase difference is 90°. The 5750's Multipath Mitigator, which removes inter-channel phase shifts and converts input audio to "intensity stereo," is important to optimum SSB/VSB operation because its action minimizes the amount of modulation overshoot.

Suboptimal system design can cause additional overshoots. To prevent this type of overshoot, the 5750's SSB/VSB generator uses constant-delay filters and its frequency response extends to DC (because of the VSB operation below 150 Hz).

To control irreducible overshoots, the SSB generator includes a look-ahead overshoot limiter. To eliminate all overshoots, this limiter must be used together with the 5750's Half-Cosine Interpolation composite limiter, which is located after the look-ahead limiter in the system block diagram.

The group delay of the phase-linear filters needed to create the SSB/VSB waveform and the audio delay in the look-ahead limiter together add approximately 12 ms to the delay of the stereo encoder. When diversity delay is applied to the 5750's composite output, the 5750 adjusts the delay automatically so that it is constant regardless of mode.

SSB stereo encoder mode can be selected from the MODULATION TYPE drop-down in the INPUT/OUTPUT > COMPOSITE screen. Choose SSB to turn on SSB/VSB operation or STEREO to turn on normal DSB operation. It can also be controlled via the 5750's GPI inputs and by PC Remote.

The look-ahead overshoot controller is always active in SSB mode, while the Half Cosine Interpolation Composite Limiter is controlled by the COMPOSITE LIMIT DRIVE control as usual.

Composite Limiter/Clipper: Orban has traditionally opposed composite clipping because of its tendency to interfere with the stereo pilot tone and with subcarriers, and because it causes inharmonic aliasing distortion, particularly between the stereo main and subchannels. Protecting the pilot tone and subcarrier regions is particularly difficult with a conventional composite clipper because appropriate filters will not only add overshoot but also compromise stereo separation—filtering causes the single-channel composite waveform to "lift off the baseline."

Nevertheless, we are aware that many engineers are fond of composite clipping. We therefore undertook a research project to find a way to peak-control the composite waveform without significantly compromising separation, pilot protection, or subcarrier protection and without adding the pumping typical of simple gain-control "look-ahead" solutions.

We succeeded in our effort. The 5750 offers a patented "Half-Cosine Interpolation" composite limiter that provides excellent spectral protection of the pilot tone and SCAs (including RDS), while still providing approximately 60 dB of separation when a single-channel composite waveform is clipped to 3 dB depth. To ensure accurate peak control, the limiter operates at 512 kHz sample rate.

For those who prefer the sound of conventional composite clipping, we also offer a defeatable composite clipper. This also provides excellent spectral protection for the pilot tone and subcarriers. The composite clipper drives the "Half-Cosine Interpolation" composite limiter, which serves as an overshoot compensator for the composite clipper when it is active. (Overshoot compensation necessary to remove overshoots introduced by the pilot- and SCA-protection filters following the composite clipper.)

Like conventional composite clipping, the "Half-Cosine Interpolation" composite limiter can still cause aliasing distortion between the stereo main and subchannels. However, this is the inevitable cost of increasing the power-handling capability beyond 100% modulation above 5 kHz—the characteristic that makes some people like composite clipping. This exploits the fact that the fundamental frequency in a square wave has a higher peak level than the square wave itself. However, any process that makes squared-off waveforms above 5 kHz creates higher harmonics that end up in the stereo subchannel region (23-53 kHz). The receiver then decodes these harmonics as if they were L–R information and the decoded harmonics appear at new frequencies not harmonically related to the original frequency that generated them.

While the processing never clips the pilot tone, the extra spectrum generated by the processing can fall into the 19 kHz region, compromising the ability of receivers to recover the pilot tone cleanly. Therefore, the 5750's composite processor has a 19 kHz notch filter to protect the pilot tone. This filter does not compromise stereo separation in any way.

We still prefer to use the 5750's main clipping system to do the vast majority of the work because of its sophisticated distortion-controlling mechanisms. This means that the 5750 does not rely on composite processing to get loud. Consequently, broadcasters using its left/right-domain AES3 digital output can enjoy the loudness benefits of the 5750's processing—the 5750 gets competitively loud without composite clipping. However, it is also possible to reduce the drive level to the 5750's left/right domain overshoot compensators and to increase the composite limiter drive by a corresponding amount.

This arrangement uses the overall composite limiter (with or without the composite clipper's being active) to provide overshoot compensation. It has a different sound than using the left/right domain overshoot compensators—the sound is brighter but has more aliasing distortion (as discussed above). If the composite clipper is active, stereo separation will decrease.

ITU-R 412 Compliance for Analog FM Broadcasts

ITU-R 412 requires the "average multiplex power" to be limited to a standard value. The 5750 contains a defeatable feedback multiplex power limiter that constantly monitors the multiplex power according to ITU-R 412 standards. The power controller automatically reduces the average modulation to ensure compliance. It allows you to set the "texture" of the processing freely, using any preset.

If a given processing setting would otherwise exceed the multiplex power limit, the power controller automatically reduces the drive to the peak limiting system. This action retains the compression texture but reduces distortion while controlling multiplex power.

The 5750 gives you control over the Multiplex Power Threshold (in the Input/output Utilities screen). This allows you to compensate for overshoots in the signal path upstream from the 5750, preventing excessive reduction of the multiplex power.

Power control is applied to all outputs, not just the composite output.

Two-Band Purist Processing

In addition to five-band processing, suitable for pop music and talk formats, the 5750 offers a very high-quality two-band algorithm. This is phase-linear and features the same AGC as the five-band processor, followed by a two-band processor with look-ahead limiting. Sophisticated multiband high frequency limiting and distortion-cancelled clipping complete the chain.

We believe that this is the ideal processing for classical music because it does not dynamically re-equalize high frequencies; the subtle HF limiter only acts to reduce high frequency energy when it would otherwise cause overload because of the FM pre-emphasis curve. We have heard four-band, allegedly “purist” processing that caused dynamic HF lift. This created a strident, unnatural sound in strings and brass. In contrast, the 5750’s two-band phase-linear structure keeps the musical spectrum coherent and natural.

The look-ahead limiter prevents speech from being audibly clipped and prevents similar audible problems on instruments with rapidly declining overtone structures like grand piano, classical guitar, and harp.

Digital Radio Processing

Only the phase rotator, highpass filter, AGC, and Multipath Mitigator are common between the FM analog and digital radio processing chains. The processing chain splits into two paths after the AGC. Each path contains a structurally identical but independently adjustable equalizer and multiband compressor. Each preset has an FM->HD CONTROL COUPLING control that determines if audio controls affecting the HD equalizer and HD multiband compressor/limiter will follow their counterparts in the FM analog processing chain or if the HD and FM controls can be adjusted independently.

The peak limiter in the digital radio processing chain is a mastering-quality lookahead limiter. This limiter minimizes IM distortion in addition to minimizing harmonic distortion. The resulting peak limiting is almost always undetectable when used with reasonable amounts of gain reduction (i.e., frequently recurring gain reduction of 3-4 dB).

Certain unusual program material may cause infrequent instances of gain reduction as high as 12 dB with the above settings. This occurs on isolated transients and is no cause for concern unless it is frequent.

Except for the fact that its input has been de-emphasized, the HD look-ahead limiter receives the same processing as the FM peak limiting section if the FM->HD CONTROL COUPLING is set to FM->HD. Earlier processing has often been adjusted to help compensate for the inevitable high frequency loss caused by pre-emphasis limiting in the FM peak limiter. Therefore, the HD output can be excessively bright without further adjustment.

In FM->HD mode, you can use the 5750’s parametric high frequency shelving filter to supply a high frequency rolloff that tames excessive brightness in the HD output. Simultaneously, this HF rolloff may reduce high frequency artifacts in the relatively low bite-rate codec used in the Xperi HD Radio system.

With the FM->HD CONTROL COUPLING set to INDEPENDENT, there are several approaches to minimizing brightness and conditioning the signal to work well at low bitrates.

- Use little or no high frequency boost in the HD equalization and band mix sections.
- Set the HD BAND 4>5 COUPLING to 100%.
- Set the HD B5 THRESH to match the codec and its bitrate. Adjust the threshold until you find a good compromise between presence and high frequency codec artifacts. We find the range from -6.0 to +6.0 dB to be useful.
- Use a moderate Band 5 attack time. 25 ms works well.
- If necessary, lower the HD B4 THRESH.

BS.1770 Compliance for Digital Radio

The 5750 includes an ITU-R BS.1770 Loudness Meter and Safety Limiter in the digital radio processing path. The Safety Limiter should only be activated if the regulatory authority in your country requires constraining BS.1770 Integrated Loudness to a specified threshold.

Input/Output Delay

The sophisticated look-ahead algorithms in the 5750 have one significant cost—the input/output time delay is longer than that of an analog processor and can cause problems if an off-air pickup is used to feed talent headphones.

To make intelligent decisions about how to process, the 5750 needs to look ahead at the next part of the program waveform.

(Slowly changing bass waveforms require particularly long look-ahead delays.) As digital on-air processing advances further and further from its analog roots, this is the inevitable price of progress.

The amount of delay depends on several things. Because of their unprecedentedly sophisticated processing, “MX” presets introduce delays of approximately 265 ms, which makes real-time monitoring through headphones impossible for talent. The 5750 offers a low-delay headphone monitoring output to solve this problem, which also occurs in HD Radio installation where the diversity delay is typically 8-10 seconds.

If a non-MX preset is active, the delay is usually low enough to allow talent to monitor live through headphone although a learning period may be required.

For a thorough discussion of delay issues and solutions, see Monitoring on Loudspeakers and Headphones on page 1-12.

Customizing the 5750's Sound

Gain Reduction Metering

Unlike the metering on some processors, when any THE OPTIMOD-5750 gain reduction meter indicates full-scale (at its bottom), it means that its associated compressor has run out of gain reduction range, that the circuitry is being overloaded, and that various nastinesses are likely to commence.

Because the various compressors have 25 dB of gain reduction range, the meter should never come close to 25 dB gain reduction if THE OPTIMOD-5750 has been set up for a sane amount of gain reduction under ordinary program conditions.

To accommodate the FM pre-emphasis curve, Band 5 of the Five-Band Structure is capable of 30 dB of gain reduction.

Further, be aware of the different peak factors on voice and music—if voice and music are peaked identically on a VU meter, voice may cause up to 10 dB more peak gain reduction than does music! (A PPM will indicate relative peak levels much more accurately.)

To Create or Save a User Preset

Once you have edited a preset, you can save it as a user preset. The 5750 can store an indefinite number of user presets, limited only by available memory.

- You cannot give a user preset the same name as a factory preset. If the name that you have selected duplicates the name of a factory preset, a warning box will appear saying:

  Factory Presets Cannot Be Overwritten

- If the name you have selected duplicates the name of an existing user preset, the 5750 warns you that you are about to overwrite that preset. Answer YES if you wish to overwrite the preset and NO otherwise. If you answer NO, the 5750 will give you an opportunity to choose a new name for the preset you are saving.

You can save user presets from the 5750 HTML5 application. Please note that when you save presets from the HTML5 application, you save them in the 5750's memory (as if you had saved them from the 5750's front panel).

When saving presets, do not use the term "Modified". "Mod" or "modified" is OK. Attempting to save it as "Modified" (with the letter M capitalized) is not allowed.

About the Processing Structures

If you want to create your own User Presets, the following detailed discussion of the processing structures is important to understand. If you only use Factory Presets or if you only modify them with LESS-MORE, then you may still find the material interesting, but it is not necessary to understand it to get excellent sound from the 5750

In the 5750, a processing structure is a program that operates as a complete audio processing system. Only one processing structure can be on-air at a time. Just as there are many possible ways of configuring a processing system using analog components (like equalizers, compressors, limiters, and clippers), 5750's DSP hardware can realize several possible processing structures. Unlike an analog system, where creating a complete processing system involves physically wiring its various components together, the 5750 realizes its processing structures as a series of high-speed mathematical computations made by Digital Signal Processing (DSP) integrated circuit chips. In the 5750, both structures operate simultaneously so there is no delay in switching between them, which is done with a smooth cross-fade.

There are three basic structures: Two-Band, Five-Band, and Ultra-Low latency Five-Band. To select a structure, choose a factory preset having the desired structure, and, if you wish, edit it to create a user preset.

Two-Band is a versatile structure that can be configured to provide purist, phaselinear processing. When correctly configured it can be used for protection limiting and we provide two presets that use it for this. It is also used for the CLASSICAL-2 BAND presets.

Five-Band is the basic structure used for popular music in its many variations. Because it provides effective automatic re-equalization of program material, it is also used for news, talk, and sports.

The stereo enhancer, AGC, equalizer, and "back end" clippers are common to both Two-Band and Five-Band processing and therefore stay the same when the 5750 switches between two-band and five-band operation.

However, different controls appear in the screens containing dynamics processing controls, as appropriate for Two-Band or Five-Band multiband compression. The meters also change functionality to display the Two-Band or Five-Band gain reduction.

Ultra-Low-Latency Five-Band reduces the input-to-output delay of the processor to about 3.7 ms at the cost of a less favorable tradeoff between loudness, brightness, and distortion than the other presets. It is comparable in performance to OPTIMODFM 8200 version 3.0 except that the clippers run at 256 kHz sample rate and are antialiased, and it offers the same stereo enhancement, equalization section, advanced technology AGC, composite limiter, and multiplex power controller as the other 5750 structures.

The only way to create an ultra-low latency user preset is to start with a "UL" factory preset and then edit that preset. "UL" user presets cannot be directly converted to low latency or optimum latency presets because the preset customization controls are different—UL presets have fewer available controls because of the difference in processing structure.

Unused structures operate constantly in the background, so switching between structures occurs with a seamless cross-fade. Unlike older Orban processors like the 8200, no DSP code is reloaded and no audio mute is necessary.

Factory Programming Presets

Factory Programming Presets are our "factory recommended settings" for various program formats or types. The Factory Programming Presets are starting points to help you get on the air quickly without having to understand anything about adjusting the 5750's sound. You can edit any of these presets with the LESS-MORE control to optimize the trade-off between loudness and distortion according to the needs of your format. Because it is so easy to fine-tune the sound at the LESS-MORE level, we believe that many users will quickly want to customize their chosen preset to complement their market and competitive position after they had time to familiarize themselves with the 5750's programming facilities.

Start with one of these presets. Spend some time listening critically to your on-air sound. Listen to a wide range of program material typical of your format and listen on several types of radios (not just on your studio monitors). Then, if you wish, customize your sound using the information in the Protection Limiter, Two-Band and Five-Band sections that follow.

Each Orban factory preset has full LESS-MORE capability. The table below shows the presets, including the source presets from which they were taken and the nominal LESS-MORE setting of each preset. Of the Five-Band presets, several appear several times under different names because we felt that these presets were appropriate for more than one format; these can be identified by the shared source preset name.

Many of the presets come in several "flavors," like "dense," "medium," and "open." These refer to the density produced by the processing. "Open" uses a slow multiband release time "Medium" uses a medium-slow release, and "Dense" uses medium-fast. A fast release is only used in the NEWS-TALK and SPORTS presets.

Important! Factory preset names are only suggestions. Feel free to audition different presets and to choose the one whose sound you prefer. This preset may have a very different name than the name of your format. This is OK.

Try using the LESS-MORE control to trade off loudness against processing artifacts and side effects. Once you have used LESS-MORE, save your edited preset as a User Preset.

Do not be afraid to experiment with presets other than the ones named for your format if you think these other presets have a more appropriate sound. Also, if you want to fine-tune the frequency balance of the programming, feel free to enter BASIC MODIFY and make small changes to the Bass, Mid EQ, and HF EQ controls. Unlike Orban's 8200, you can make changes in EQ (and stereo enhancement) without losing the ability to use LESS-MORE settings.

Of course, LESS-MORE is still available for the unedited preset if you want to go back to it. There is no way you can erase or otherwise damage the Factory Presets. So, feel free to experiment.

- If the preset has "UL" in its name, it uses the Ultra-Low Latency Five-Band structure. "UL" presets are not as competitive as other presets and should only be used if you absolutely need the low delay (for off-air cueing of finicky talent, for example).
- Presets with LL in their names use the Hard LL bass clipper mode to achieve 13 ms input-output delay.
- Presets without LL or UL designators in their names have "optimum delay" for an 8500-style preset. This delay is approximately 18 ms delay (5-band) and 21 ms delay (2-band).

PROTECTION-0DB: PROTECTION-0DB is a two-band preset with a high amount of band coupling. It is intended for use below threshold most of the time (i.e., with 0 dB gain reduction), to provide protection limiting in the highest quality applications such as serious classical music intended for an attentive audience. Its LESS-MORE control determines the normal amount of gain reduction but does not increase distortion or other processing artifacts when turned up.

COUNTRY: The COUNTRY-MEDIUM preset uses the ROCK-SMOOTH source preset. It has a gentle bass lift and a mellow, easy-to-listen-to high end, along with enough presence energy to help vocals to stand out. The COUNTRY-LIGHT preset uses the ROCK-LIGHT source preset. Modern country stations might also find ROCK-MEDIUM or ROCK-OPEN useful if they want a brighter, more up-front sound.

Measured in third-octaves, the two presets typically produce less than 3 dB of difference with program material, so either preset will work OK (although not ideally) with all radios.

DANCE ENERGY: This 8500 preset is designed to preserve the punch and slam in dance music percussion (such as the beater click in kick drums). It uses HARD bass clipping, is loud, and has a bright high frequency texture. It was designed for 50 µs preemphasis and many user will find it to be too distorted when used at 75 µs. As LESS-MORE is turned down, this preset get quieter, yet punchier.

FOLK-TRADITIONAL: FOLK-TRADITIONAL is an alias for the ROCK-SOFT preset. It assumes that the recordings are of relatively recent vintage and require relatively subtle processing.

If the recordings you play are inconsistent in texture and equalization, you may prefer the ROCK-SMOOTH preset.

GOLD: GOLD is loud and "hi-fi"-sounding while still respecting the limitations and basic flavor of the recordings from the era of the 1950s through 1970s.

- For example, we do not attempt to exaggerate high frequency energy in the GOLD preset. The highs in recordings of this era are often noisy, disorted, or have other technical problems that make them unpleasant sounding when the processor over-equalizes them in an attempt to emulate the high frequency balance of recently recorded material.

- **GOLD OPEN**: is least sensitive to source material and is appropriate for "oldies" from the earliest rock and roll era (ca. 1954) to the late 1970s. This preset has no counterpart among the "8500-style" presets. It is a relaxed, clean, easysounding preset that does not attempt to emulate the audio processing of radio stations back in the day when this music is current.

- **GOLD HEAVY:** is appropriate for music from the mid-1960s to the late 1970s. It produces a denser sound than GOLD OPEN with more of a "classic Top-40" processing flavor.

- **GOLD HEAVY BASS**: is appropriate for carefully produced music from the mid-1960s to the late 1970s. This preset can increase the bass centered broadly around 60 Hz by as much as 6 dB, which provides "bass thump" for material whose bass was originally weak. Because GOLD HEAVY BASS can amplify bass by a large amount, it can also amplify rumble and AC line frequency hum (often from guitar amplifiers in the era before noise gating was used routinely on instrument inputs when records were mixed). Stations using GOLD HEAVY BASS should therefore make sure that their source material is cleaned up to be free from rumble and hum.

GREGG, GREGG MX: GREGG, GREGG OPEN, and GREGG LL all use a 200 Hz band1/band2 crossover frequency to achieve a bass sound similar to the classic five-band Gregg Labs FM processors designed by Greg Ogonowski. Dynamically, these presets produce a slight increase in bass energy below 100 Hz and a decrease of bass energy centered at 160 Hz. This bass sound works particularly well with radios having good bass response, such as many auto radios today.

INSTRUMENTAL: An alias for the JAZZ preset,

JAZZ: JAZZ is an 8500 preset specifically tailored toward stations that play mostly instrumental music, particularly classic jazz from the LP era (Coltrane, Mingus, Monk, etc.). It is a quiet preset with a very clean, mellow high end to prevent stridency on saxes and other horns. It preserves much of the qualities of the original recordings, doing light re-equalization. The preset produces very low listening fatigue, so it is a good choice for stations that want listeners to stay all day.

There is also an 8500-style SMOOTH JAZZ preset available.

LOUD: There are several LOUD presets, all of which use “8500-style” technology and typically have 18 ms of latency. We have not made MX versions of the LOUD presets; we retained these presets in the 5750 mainly for historical reasons. Most were first introduced with THE OPTIMOD-5750 8400 and some 5750 users may have custom presets based on them.

In order to get the punchiest and loudest sound, beyond LESS-MORE=7.0 we progressively reduce the protection provided by the distortion-controlling mechanism. So LESS-MORE settings beyond 7.0 are progressively more risky and can exhibit audible distortion.

LOUD-HOT+BASS is very bright and present, with up-front vocals. Release time is medium. It is tuned for the maximum amount of bass we could add without creating obvious distortion on some program material. For maximum punch, it uses the HARD bass clipper at higher LESS-MORE settings.

This amount of bass may be excessive with certain consumer radios (particularly “boom-boxes”) that already have substantial bass boost. Use it with care.

LOUD-HOT+BASS LL is the low-latency version of LOUD-HOT+BASS.

LOUD+SLAM is similar to LOUD-HOT+BASS, but uses HARD bass clipping mode with a SHAPE of 7.6, a BASS SLOPE of 18 dB/octave. It has modified tuning in the band-1 compressor (to control bass clipping distortion that could otherwise be introduced by Hard bass clipping). This preset provides slamming bass punch, which it trades off against bass cleanliness on certain program material. Because of the 18 dB/octave BASS SLOPE, its advantages will be appreciated most through radios with good low bass response.

LOUD-COMPRESSED retains the full 8500-style distortion-controlling mechanism for all LESS-MORE settings. Because this mechanism reduces clipper drive to prevent waveforms from being clipped excessively, it can pump audibly when being used to the extreme that it is in this preset. This is a sound texture that some people have requested.

LOUD-PUNCHY is the quietest of the “loud” preset family. It is designed for a bright, sizzling top end and very punchy lows. It is a good choice for stations that feel that the LOUD-HOT presets are too aggressive, but that think that the ROCK presets are insufficiently loud for their market position.

NEWS-TALK: This preset is quite different from the others. It is based on the fast multiband release time setting, so it can quickly perform automatic equalization of substandard program material, including telephone. It is very useful for creating a uniform, intelligible sound from widely varying source material, particularly source material that is “hot from the field” with uncontrolled quality.

It extensively exploits distortion control to achieve a very clean, highly compressed, but unclipped sound quality.

SPORTS: Similar to NEWS-TALK except the AGC Release (AGC Release Time) is slower and the Gate Thresh (Gate Threshold) is higher. This recognizes that most sports programming has very low signal-to-noise ratio due to crowd noise and other onfield sounds, so the preset does not pump this up as the NEWS-TALK preset would tend to do.

ROCK: We have included many of the 8500-style ROCK presets and have intentionally tuned them for a smooth, mellow high frequency balance, which can complement female-skewing formats.

ROCK-DENSE, ROCK-MEDIUM, and ROCK-OPEN provide a bright high end and punchy low end (although not as exaggerated as the URBAN presets). A midrange boost provides enough presence energy to ensure that vocals stand out. A modest amount of high frequency coupling (determined by the Band Clipping 3 > 4 setting) allows reasonable amounts of automatic HF equalization (to correct dull program material), while still preventing exaggerated frequency balances and excessive HF density. Dense, medium, and open refer to the compression density, which is determined by the release time settings in the AGC and multiband limiter sections.

These presets are appropriate for general rock and contemporary programming. All of these presets have distortion control implemented at their nominal levels of LESSMORE to ensure clean speech. At high LESS-MORE levels the distortion control may be relaxed somewhat to increase bass punch.

ROCK-SOFT has a mellow, easy-to-listen-to high frequency quality that is designed for female-skewing formats. It is also a candidate for “Quiet Storm” and “Love Songs” light rock or light urban formats.

ROCK-SMOOTH has the same mellow, easy-to-listen-to high frequency quality as ROCK-SOFT, but with more density. Again, it is a good choice for female-skewing formats, but where you need more compression and density than you get with ROCK-SOFT.

ROCK-MEDIUM+LOWBASS is an open-sounding preset with a lot of bass punch. Its Multiband Release control is set to Slow2 so that the sound is relaxed and not at all busy. At the same time, the preset is competitively loud. It is an excellent choice for “adult contemporary” and “soft rock” formats where long time-spent-listening is desired.

The SMOOTH JAZZ MX preset exploits the 5750’s ability to produce more transient impact and lower distortion. For either preset, if the loudness/distortion tradeoff is not to your taste use LESS-MORE to turn it down, producing lower loudness with less distortion.

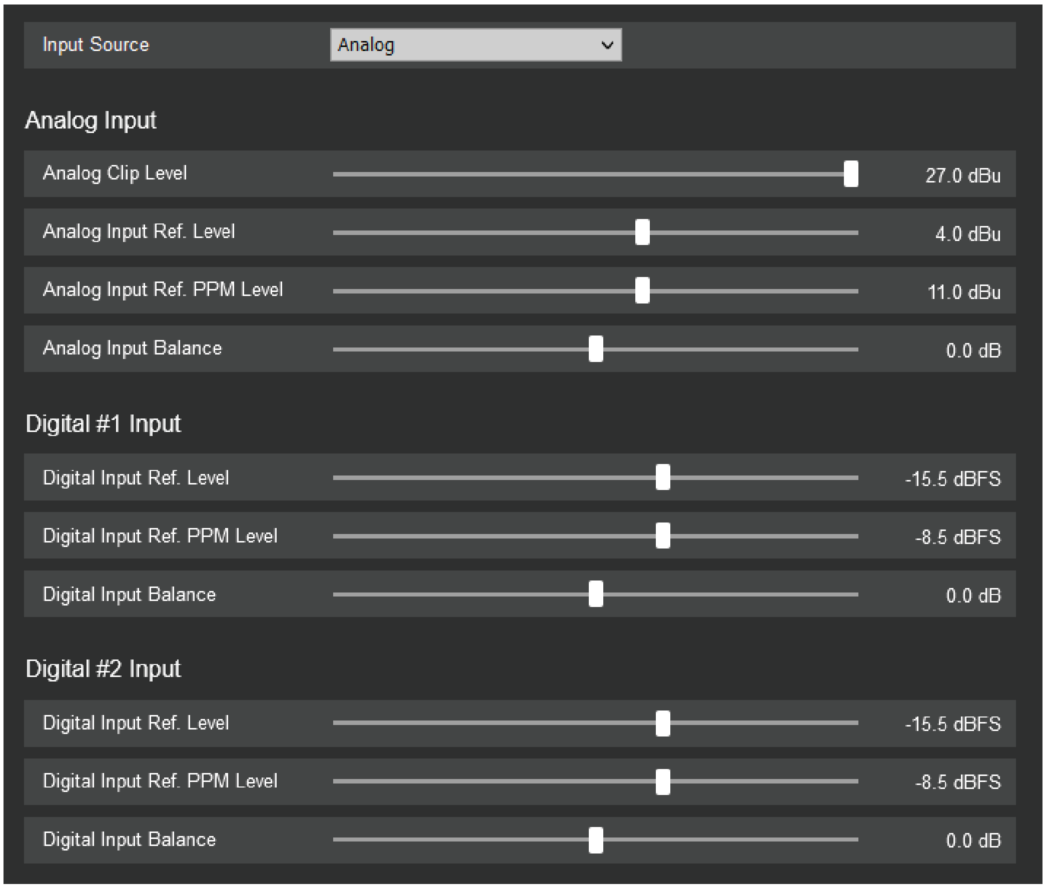
URBAN-LIGHT: This is an 8500-style preset that has been retained in the 5750 mainly for historical reasons; GREGG MX, IMPACT MX, and EDGE MX are more contemporary sounding. URBAN-LIGHT is similar to ROCK-OPEN but with a different bass sound. It uses the 3-pole (18 dB/octave) shape on the bass equalizer and is appropriate for light R&B formats.

Adjusting and using the browser based PC Remote

The 5750 HTML5 application allows you to adjust every parameter in the 5750 audio processor. Each block of processing is accessible from tabs that are aligned between the meter display and the controls. In the following pages we will step thru the controls tab-by-tab for an understanding of what you can accomplish with each.

To setup the connection of the browser with the processor, see page 2-27.

SETTING INPUT LEVELS



Navigate to the I/O Settings tab and open it. Select the INPUT tab on the right column. Here you will see options for input source (Digital or Analog) and the input reference level for both. This is not a traditional gain trim, you are setting the input gain reference level. Be certain to leave enough headroom when setting this so it is properly driven by equipment before it.

The reference level VU and PPM (Peak) settings track each other with an offset of 7 dB. This compensates for the typical indications with program material of a VU meter versus the higher indications on a PPM.

This step sets the center of the 5750's gain reduction range to the level to which your studio operators peak their program material on the studio meters. This assures that the 5750's processing presets will operate in their preferred range. You may adjust this level with a standard reference/line-up level tone from your studio or with program material.

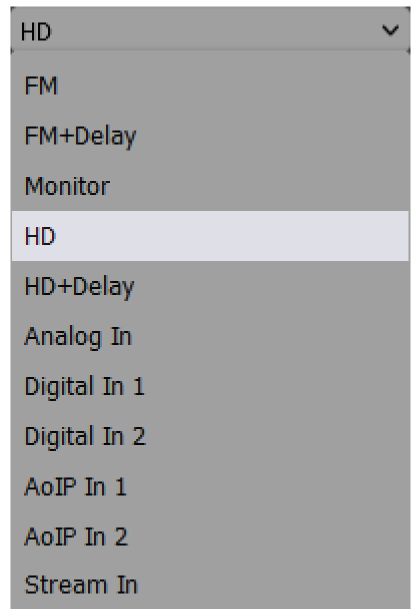
Note that in this step, you are calibrating to the normal indication of the studio meters; this is quite different from the actual peak level.

SETTING ANALOG OUTPUT LEVELS

The next tab down displays the analog output levels. If you are using the ANALOG OUTPUT to feed your transmitter, you can use the slider to get the peak level to within range of your modulation.

Also on the analog output is the option to use FLAT or PRE- EMPHASIZED audio.

If you are feeding a transmitter with a stereo encoder, it is best practice to defeat any pre-emphasis in the transmitter or stereo encoder and perform pre-emphasis in the 5750 IF the audio path is linear (non compressed). In some situations where audio needs to pass thru a codec, it is then necessary to send a FLAT audio signal across.



Please note that there may be a loudness penalty in this situation as the peak control of pre-emphasized audio in the transmitter or stereo encoder will not be as accurate as it would be in the 5750's processing.

If you are using a non linear audio path, consider moving the 5750 to the transmitter site if possible. This will restore the 5750's peak control at the transmitter point for best peak management of your audio.

You may also switch the analog output for the following options:

- FM (no HD delay can be applied)
- FM+HD (FM audio after any delay is applied)
- Monitor (A low latency signal to feed talent headphones for an "off air" experience)
- HD (The output of the 5750's HD audio path)

HEADPHONE

The drop down menu on the headphone control lets you choose what part of the 5750 audio path you want to listen to.

- FM (no HD delay can be applied)
- FM+HD (FM audio after any delay is applied)
- Monitor (A low latency signal to feed talent headphones for an "off air" experience)
- HD (The output of the 5750's HD audio path)
- HD+Delay (FM audio after any delay is applied)
- Analog input audio

- Digital 1 Input audio
- Digital 2 input audio
- AoIP 1 Input Audio
- AoIP 2 Input Audio
- Streaming Input Audio

The headphone level slider increases or decreases the headphone volume.

Digital OUTPUT 1 (and 2)

If you are using one of the DIGITAL (AES) OUTPUTS to feed your transmitter, you can use the output level slider to set the peak level within range of your modulation.

Also on the digital output is the option to use FLAT or PRE-EMPHASIZED audio.

If you are feeding a transmitter with a stereo encoder, it is best practice to defeat any pre-emphasis in the transmitter or stereo encoder and perform pre-emphasis in the 5750 IF the audio path is linear (non compressed). In some situations where audio needs to pass thru a codec, it is then necessary to send a FLAT audio signal across.

Please note that there may be a loudness penalty in this situation as the peak control of pre-emphasized audio in the transmitter or stereo encoder will not be as accurate as it would be in the 5750's processing.

If you are using a non linear audio path, consider moving the 5750 to the transmitter site if possible. This will restore the 5750's peak control at the transmitter point for best peak management of your audio.

You may also use the analog output for the following:

- FM (no HD delay can be applied)
- FM+HD (FM audio after any delay is applied)
- Monitor (A low latency signal to feed talent headphones for an "off air" experience)
- HD (The output of the 5750's HD audio path)

Synchronization determines if the sample rate appearing at the digital-channel output is synced to the 5750's internal clock, to an AES3 signal appearing at the 5750's digital input, or to an AES11 signal appearing at the 5750's sync input. Sync can be set separately for Digital Output 1 and Digital Output 2, allowing them to have different sample rates.

The selections for each of the two AES outputs are Internal, Sync In, and Input. Input sets a given AES3 output sample rate and synchronization to the same sample rates.

The selections for each of the two AES outputs are Internal, Sync In, and Input. Input sets a given AES3 output sample rate and synchronization to the same sample rate present at the 5750's AES3 (audio) input. Likewise, Sync In uses the AES11 sync input's sample rate and synchronization as the source. Internal synchronizes the given AES3 output rate to the 5750's internal clock and uses the Samp Rate setting to determine its output sample rate.

For a given AES3 output, the output sample-rate selector ("Samp Rate") has no effect in the Input and Sync In modes unless sync is lost. Then the output reverts to internal sync at the sample rate that is preset by the sample-rate selector for that output. Otherwise, the output sample rate follows the sample rate present at the selected input, regardless of the setting of the output sample rate selector.

If no signal is provided to the 5750 Input or Sync In, set SR Sync to Internal and select the desired output sample rate.

The 5750 Sample Rate can be set from 32kHz to 96kHz. Most systems will either use 44.1 or 48kHz.

Word Length sets the word length (in bits) emitted from the digital-channel output.

The largest valid word length in the 5750 is 24 bits. The 5750 can also truncate its output word length to 20, 18, 16, or 14 bits. The 5750 can also add dither, which we recommend.

Dither turns on or off addition of "high-pass" dither before any truncation of the output wordlength.

The amount of dither automatically tracks the setting of the Word Length control. This first-order noise shaped dither adds considerably less noise in the midrange than does white PDF dither. However, unlike extreme noise shaping, first-order noise shaped dither adds a maximum of 3 dB of excess total noise power when compared to white PDF dither. It is thus a good compromise between white PDF dither and extreme noise shaping.

In many cases, the source material has already been correctly dithered so you will not need to add dither and can set this control to Out. However, particularly if you use the Noise Reduction feature, the processing can sometimes attenuate input dither to a point where it is insufficient to dither the output correctly. In this case, you should add dither within the 5750 by turning this control on.

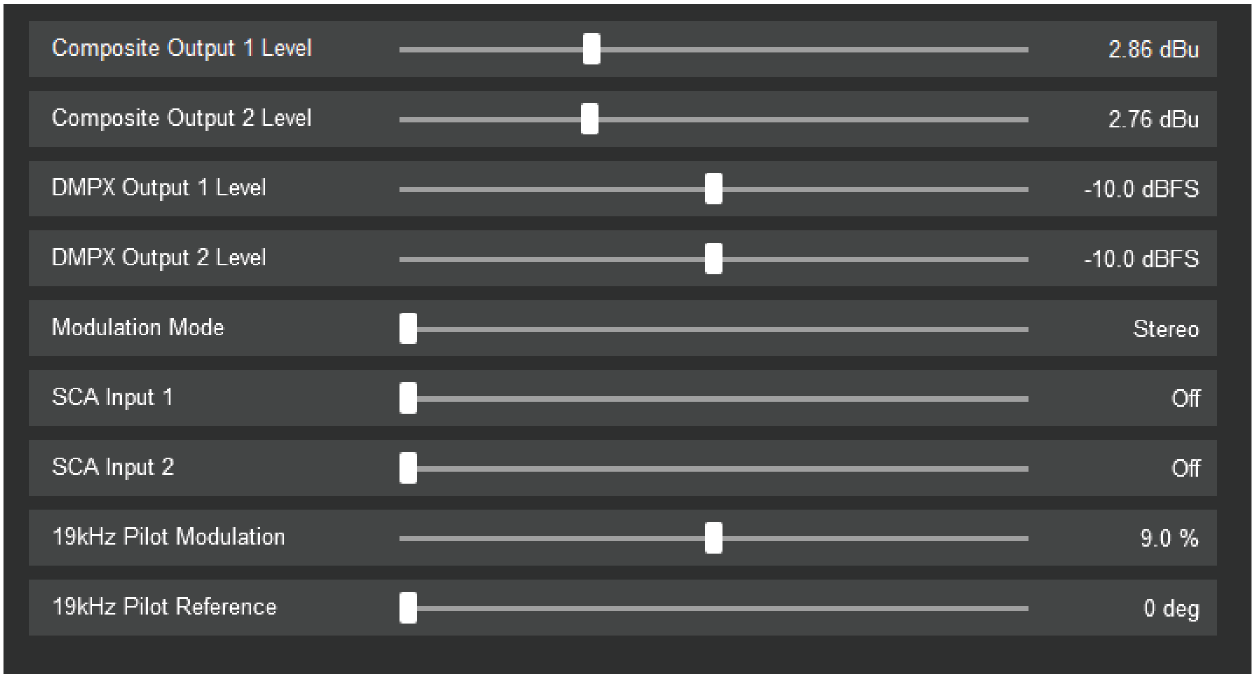
Format determines if the digital-channel output follows the professional AES3 or consumer SPDIF standard.

We expect that AES will be appropriate for almost all users, but some consumer sound cards may require SPDIF.

Output 2, just below the Output 1 controls, operates in the same manner as Output 1.

STEREO ENCODER

There are two analog composite outputs on the 5750 under the STEREO ENCODER tab. Each composite output has its own gain control.



Using normal program material and a calibrated modulation monitor, adjust the composite output so that peaks occur at 100% (75khz). If you have a backup transmitter, it is wise to adjust the composite 2 output so it is also showing peaks occurring at 100% (75kHz).

From here, you may adjust the modulation based on the number of subcarriers and how much your local governing body allows.

It should be noted that Orban uses very tight and peak controlled audio stages on its output. Any use of the output of the composite processor to make up loudness is not suggested for regulatory reasons and because of behavior of receivers when excessive modulation is used.

MODULATION MODE: You can choose between STEREO, PILOT OFF, MONO LEFT, MONO RIGHT, MONO SUM and STEREO with SSB (Single Sideband Suppressed Carrier).

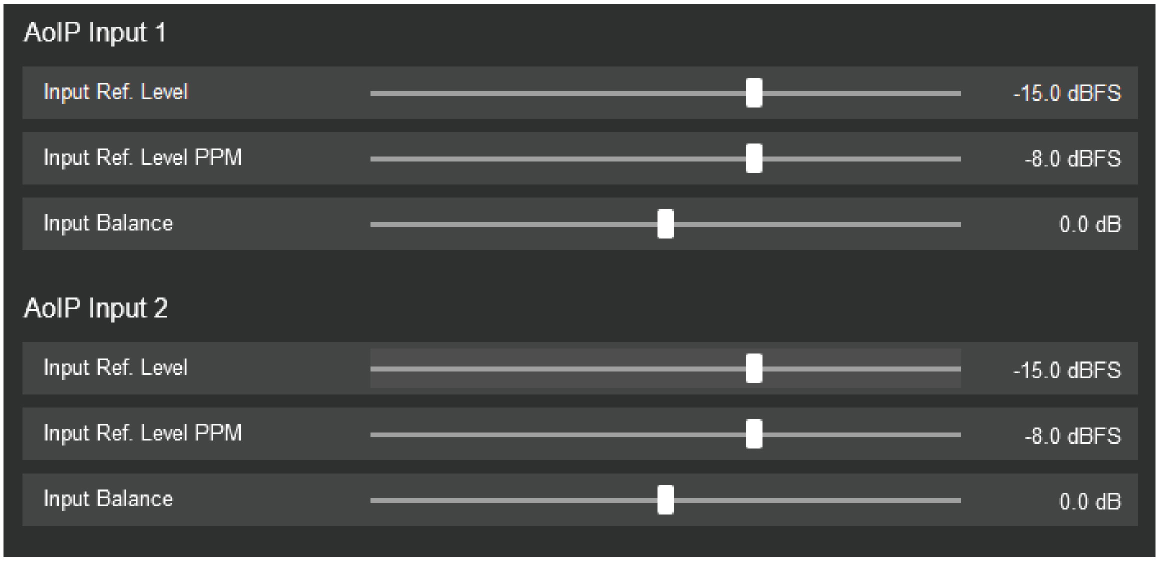
SCA 1: Sets the level of the digital SCA 1 input.

SCA 2: Sets the level of the digital SCA 2 input.

19kHz PILOT MODULATION: Sets the pilot reference level. 9% is the standard in most situations.

19kHz PILOT REFERENCE: Sets the phase of the reference output with respect to the stereo pilot tone at the composite output.

AoIP INPUT



AoIP 1 and 2 can be set here for proper operating level. This is not a traditional gain trim, you are setting the input gain reference level. Be certain to leave enough headroom when setting this so it is properly driven by equipment before it.

The reference level VU and PPM (Peak) settings track each other with an offset of 7 dB. This compensates for the typical indications with program material of a VU meter versus the higher indications on a PPM.

This step sets the center of the 5750’s gain reduction range to the level to which your studio operators peak their program material on the studio meters. This assures that the 5750’s processing presets will operate in their preferred range.

You may adjust this level with a standard reference/line-up level tone from your studio or with program material.

Note that in this step, you are calibrating to the normal indication of the studio meters; this is quite different from the actual peak level.

INPUT BALANCE: Offsets any differences from the left and right channel. Please note this should only be a temporary fix. The source of the channel imbalance should be tracked down and corrected.

AoIP 1-4 OUTPUT

If you are using one of the AoIP OUTPUTS to feed audio, you can use the output level slider to set the peak level within range of the desired level.

Also on the digital output is the option to use FLAT or PRE-EMPHASIZED audio.

If you are feeding a transmitter with a stereo encoder, it is best practice to defeat any pre-emphasis in the transmitter or stereo encoder and perform pre-emphasis in the 5750. In some situations where audio needs to pass thru a codec, it is then necessary to send a FLAT audio signal across.

Please note that there may be a loudness penalty in this situation as the peak control of pre-emphasized audio in the transmitter or stereo encoder will not be as accurate as it would be in the 5750's processing.

OUTPUT SOURCE: Can be selected from one of the following:

- FM (no HD delay can be applied)
- FM+HD (FM audio after any delay is applied)

- Monitor (A low latency signal to feed talent headphones for an “off air” experience)
- HD (The output of the 5750’s HD audio path)
- HD+Delay (FM audio after any delay is applied)

Word Length sets the word length (in bits) emitted from the digital-channel output.

The largest valid word length in the 5750 is 24 bits. The 5750 can also truncate its output word length to 20, 18, 16, or 14 bits. The 5750 can also add dither, which we recommend.

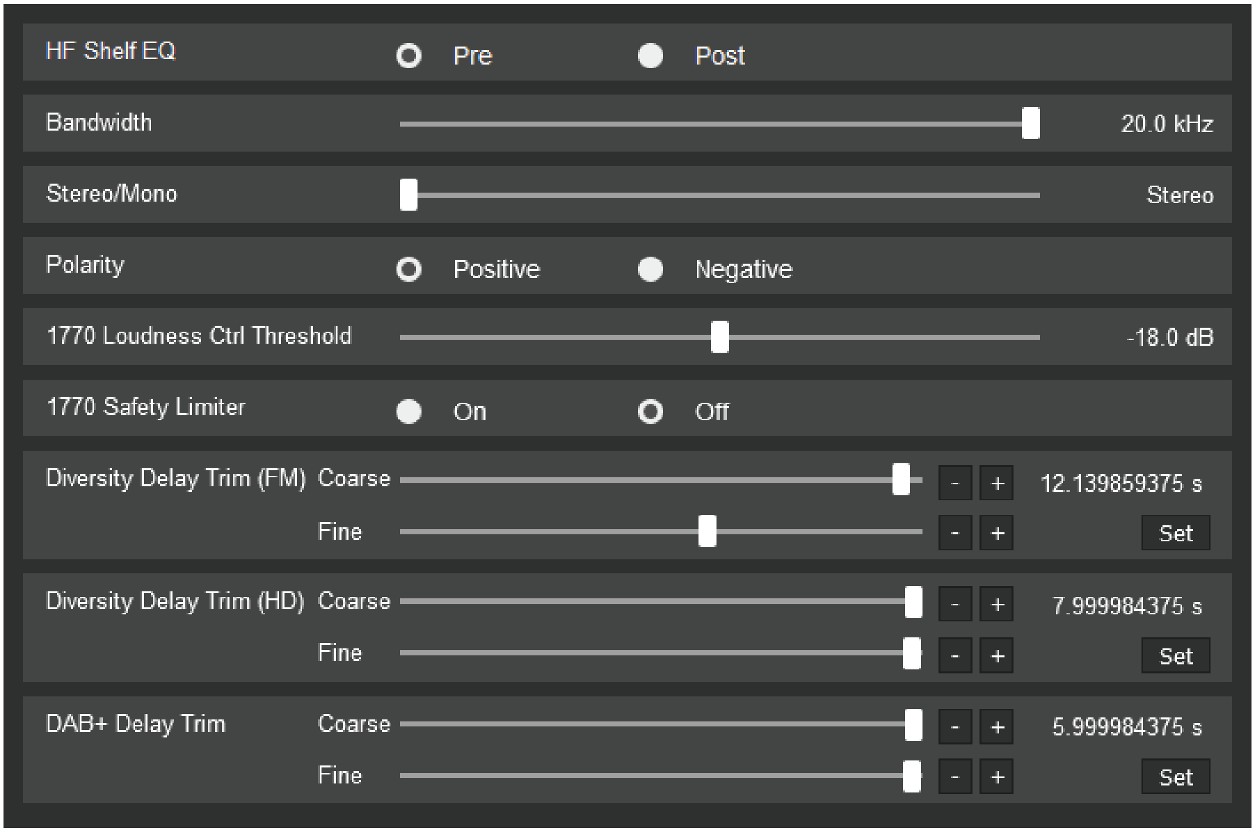
Dither turns on or off addition of “high-pass” dither before any truncation of the output wordlength.

The amount of dither automatically tracks the setting of the Word Length control. This first-order noise shaped dither adds considerably less noise in the midrange than does white PDF dither. However, unlike extreme noise shaping, first-order noise shaped dither adds a maximum of 3 dB of excess total noise power when compared to white PDF dither. It is thus a good compromise between white PDF dither and extreme noise shaping.

In many cases, the source material has already been correctly dithered so you will not need to add dither and can set this control to Out. However, particularly if you use the Noise Reduction feature, the processing can sometimes attenuate input dither to a point where it is insufficient to dither the output correctly. In this case, you should add dither within the 5750 by turning this control on.

- The same controls from AoIP 1 output apply to the other 3 outputs.

HD DIGITAL RADIO



HD HF SHELF EQ (Pre/Post) determines whether the HD HF shelving equalizer will be placed before or after the look-ahead limiter that feeds the HD output.

HD POLARITY sets the polarity of the output of the HD processing to Positive or Negative. The switch allows you to match the polarity (sometimes informally called “phase”) of the audio through the analog FM and HD transmission channels, regardless of how your facility is configured. It is important to match polarity to avoid a momentary decrease in loudness during analog/digital receiver crossfades.

BS.1770 SAFETY LIMITER ON/OFF activates or defeats the BS.1770 Safety Limiter, which is applied only to the digital radio processing. This control affects the behavior of the digital output receiving the HD feed

BS.1770 LOUDNESS CONTROL THRESHOLD sets the threshold of the BS.1770 Safety Limiter and the calibration of the BS.1770 loudness meter.

When the BS.1770 Safety Limiter is OFF, the BS.1770 Loudness Control Threshold sets the calibration of the BS.1770 Loudness Meter, such that “0” LK/LU on the meter corresponds to the loudness appearing at the Digital Output assigned to “HD.” This calibration is only correct if the Digital Output 100% Peak Level control is set to 0 dBFS.

When the BS.1770 Safety Limiter is ON, this calibration is correct regardless of the setting of the Digital Output 100% Peak Level control.

HD BANDWIDTH sets the audio bandwidth of the HD output from 15 to 20 kHz in 1 kHz steps.

The user should carefully test the codec in use to ascertain if lowering the bandwidth to 15 kHz improves subjective quality. This can occur because the codec then uses all of its bits to encode information in the most subjectively important part of the ear's bandwidth.

If the codec employs Spectral Band Replication® technology (as does the HD Codec), then you can set the bandwidth at 20 kHz without quality penalty, although the difference between 15 and 20 kHz is unlikely to be audible following the encode / decode cycle.

ST./MONO ("HD Output Stereo / Mono Mode") determines if the digital-channel output will be fed by the normal stereo output of the HD processing chain or by a mono feed from the HD processing chain's left channel, right channel, or sum of left and right channels. In all cases, the signal appears on both the left and right channels of the analog and digital outputs.

DIVERSITY DELAY TRIM allows you to trim the analog FM delay in intervals of one sample of 64 kHz (15.6 μs) so that the delays of the analog-FM and digital radio channels are matched at the receiver's crossfade point. This prevents audible comb filtering during crossfades. The setting of this control is critical to get best results and you should adjust it to one-sample accuracy.

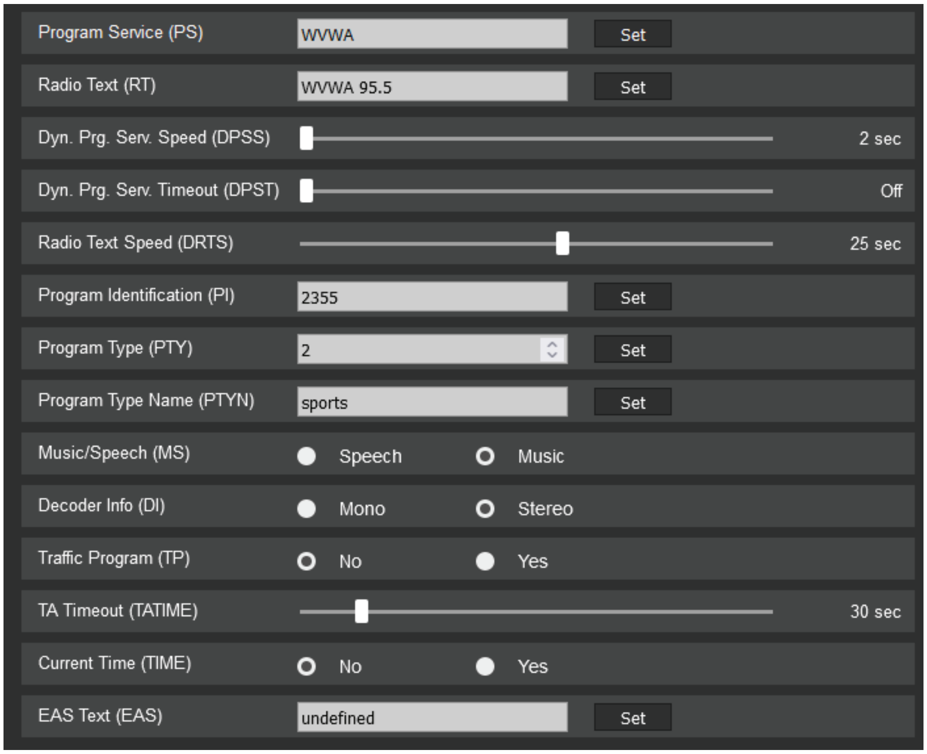
In the HTML5 application, drag the slider of the control with the mouse to set the delay coarsely, use the Page Up and Page Down keys for intermediate increments, and use the mouse wheel to fine-tune the delay in one-sample adjustment increments.

Maximum available delay is approximately 16 seconds.

See the documentation provided with your HD Radio exciter for more information on setting the delay correctly.

RDS and RBDS

Your OPTIMOD includes a full-featured RDS/RBDS generator that supports dynamic PS. We presume that you are already familiar with the basics of RDS and you wish to implement RDS via your OPTIMOD.



Program Service: (PS) 256 (max) characters for scrolling messages in the PS field

PS = Artist Goes Here
Title Goes Here

Radio Text: (RT) 128 (max) character message to be displayed by the receiver if so equipped.

RT= WBZX (800) 111-1111

website here

Dynamic Program Service Speed: (DPSS) The number of seconds the PS will pause before showing the next PS segment. Use the slider to adjust

Dynamic Program Service Timeout: (DPST) The number of minutes between receipt of the last DPS message and the transmission of the DPS Default message.

Radio Text Speed: (DRTS) Determines how fast the text scrolls on the receiver screen.

Program Identification: (PI) 4 Digit HEX number corresponding to the station's "digital address" (ex: PI=3D44)

Program Type: (PTY) Index in PTY list that describes the broadcast format. (ex: 9 is "Top 40" in the United States)

Program Type Name: (PTN) 8 characters to further define program format.

Music/Speech (MS): Set for type of format (spoken word or music)

Stereo/Mono: Set to the mode you are transmitting with.

Traffic Program: (TP) Yes if the station broadcasts routine traffic reports. No if it does not.

TA Timeout: (TATIME) Seconds between start of TA flag and automatic reset to OFF.

Current Time: (TIME) Transmits the current time to receivers that have time sync. IMPORTANT – Make sure the 5750 has the correct time before enabling this feature or receivers will display the wrong time of day. If you aren't sure you will be able to keep the system time correct, choose NO.

EAS TEXT: Displays a unique message when EAS mode has been triggered.

EAS TIME: Returns the number of seconds remaining for the current on-air EAS alert transmission.

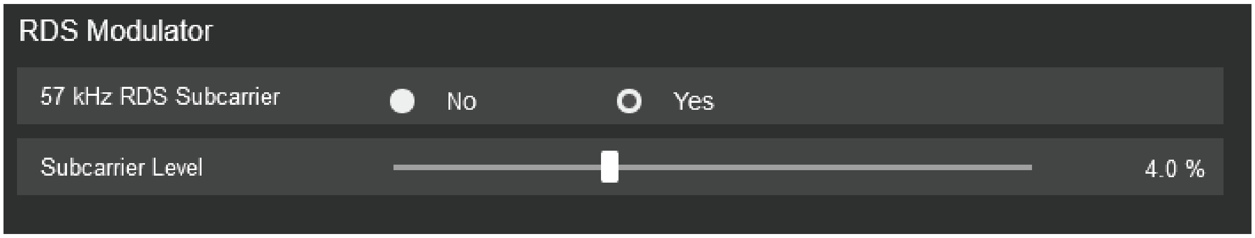
UECP CLIENT (Universal Encoder Communication Protocol)

The RDS encoder supports network-based UECP connectivity with either the TCP/IP or UDP protocol. Use the PHTML5 application to set these client settings from the RDS tab in the I/O Setup dialog: Site Address, Encoder Address, TCP/IP Port number, UDP, and UECP Active status. Use the values suggested by your UECP Server software for these fields.

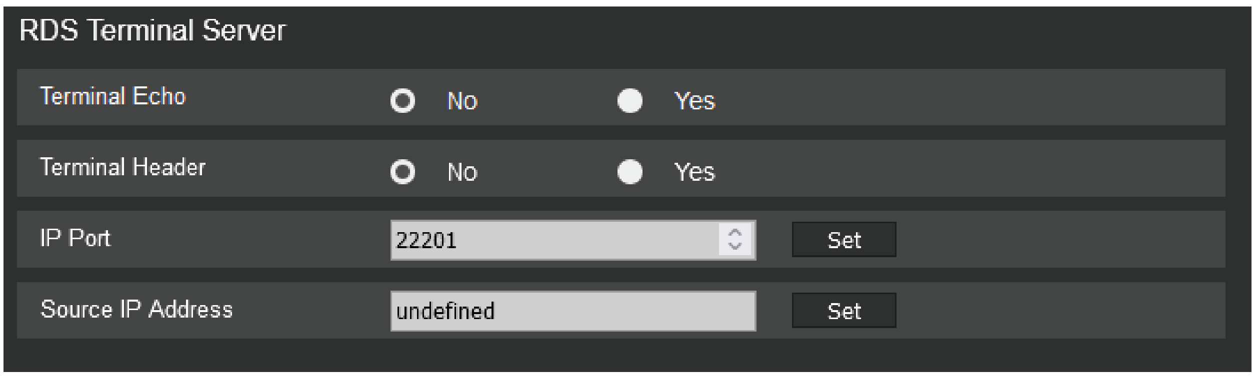
The client TCP/IP Port is only updated at the OPTIMOD when you disconnect the PC remote software from the unit.

RDS MODULATOR

Turns on or off the RDS subcarrier. The Subcarrier level controls the amount of RDS injection. Orban suggests using a modulation monitor that is capable of reading the 57 kHz subcarrier to make this adjustment.



USING TELNET TO CONTROL TERMINAL SERVER

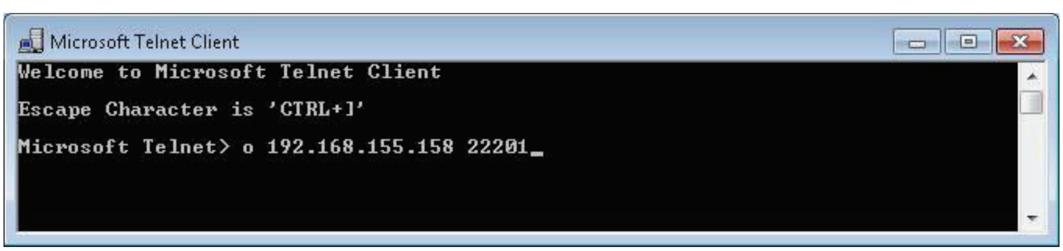


You can control RDS from Windows directly via the Windows Telnet command line utility or the free utility PuTTY.

A) Open the Windows Telnet client by typing telnet into the Windows Run box in the Start menu and hitting the Enter key on your keyboard.

In Windows 7 and higher, you must enable the Telnet client; it is not enabled by default. If you do not know how, use a search engine to find out. The general idea is to navigate to CONTROL PANEL > PROGRAMS > TURN WINDOWS FEATURES ON OR OFF and check "Telnet."

B) Connect the Telnet client to the OPTIMOD by typing open [IP address] [IP Port], where [IP address] is the IP address of the OPTIMOD and [IP Port] is the IP Port you assigned to the OPTIMOD RDS terminal server in System.



C) You may now type any of the terminal commands in the chart of RDS Terminal Commands below.

If you have checked the OPTIMOD's TERMINAL ECHO box, when you type a command, the OPTIMOD will return a status line relevant to the command to the Telnet client, which will write it to the screen.

The returned information will look similar to the following:

```
?
PS=undefined
DPS=undefined
DPSS=2 Seconds
DPST=Off
RT=KKDB MORE HIT MUSIC
DRTS=Off
PI=3D44
PTY=9
PTYN=ROCK
MS=Music
DI=Stereo
TP=No
TA=No
TATIME=30
TIME=No
RDS=Yes
RDSLEVEL= 6.0 %
AF 1=87.7 MHz
AF 2=0
AF 3=0
AF 4=0
AF 5=0
AF 6=0
AF 7=0
AF 8=0
AF 9=0
AF 10=0
AF 11=0
AF 12=0
AF 13=0
AF 14=0
AF 15=0
AF 16=0
AF 17=0
AF 18=0
AF 19=0
AF 20=0
AF 21=0
AF 22=0
AF 23=0
AF 24=0
```

Security

In the System I/O RDS control screen, you can set the RDS *Terminal control security* by specifying an RDS IP address from which to accept commands. Once set, this IP will be the only IP that can connect to the unit to update RDS. The 5750 will default to 0.0.0.0, which will allow any IP to connect to the RDS terminal control.

To prevent the OPTIMOD from disconnecting and being unable to reconnect if the terminal connection drops out temporarily, set the TIMEOUT value to the maximum expected duration of the dropout (in minutes). Default is 4 minutes. Note that the timeout reverts to the default each new connection; if you change the timeout for one connection, it is not retained for the next one.

RDS Terminal Commands

lists the terminal commands.

Note that you can fetch the status of the RDS generator as follows:

[command] ? returns current value
ST *returns current value of all controls*
HELP *returns a list of the RDS terminal commands*
↵ = CR/LF

COMMAND	PARAMETER	INFORMATION	EXAMPLE
PS= / DPS=	Dynamic PS	256 (max) characters for scrolling messages in the PS field	DPS=Artist Goes Here :: Title Goes Here↵
DPSS=	DPS Dynamic Program Service Speed	OFF = Default PS Option Disabled 2 - 9 = wait time in seconds between receipt of last incoming DPS and transmission of DPS Default. Default is the DPS control value as designated in the I/O Setup.	DPSS = 2
DPST=	DPS Dynamic Program Timeout	0 = Default PS Option Disabled 1 - 7 = wait time in minutes between receipt of last incoming DPS and transmission of DPS Default. Default is the DPS control value as designated in the I/O Setup.	DPSTIMEOUT=0
RT=	Radio Text	128 (max) character message to be displayed by the receiver if so equipped	RT=KRD :: (800) 111-1111 :: www.domain.com↵
DRTS=	RadioText Speed	0 = RadioText OFF 0,5,10,…45 (steps of 5) = Refresh rate for RadioText message transmission (15 recommended for text messaging, higher values for RT+ applications)	DRTS=15
PI=	Program Identification	**4-digit HEX number**[1] corresponding to the Station Call Letters — RDS North America ONLY	PI=3D44↵ (for KRDS)
PTY=	Program Type (Format)	1 or 2 digit number from PTY list describing the station broadcast format — RDS & RBDS are DIFFERENT	PTY=9↵ (for North American "TOP 40")
PTYN=	Program Type Name	8-character refined format definition — RDS & RBDS are DIFFERENT	PTYN=TOP 40↵
EAS=	Text of Emergency Alert System message	Text of the EAS message (64-character maximum). It will be transmitted *after* you send a non-zero EASTIME= command to the encoder. It temporarily overrides the PS and RT messages and sets the PTY code to 31.	EAS=This is an Emergency Broadcast System test.↵

[1] See section D.7 of the NRSC-4-B Standard and section 5.1 of the NRSC-G300-B RBDS Usage Guidelines.

COMMAND	PARAMETER	INFORMATION	EXAMPLE
EASTIME=	Duration of EAS message (seconds)	Countdown timer for transmission of EAS (Emergency Alert System) text. Send this command *after* the EAS= command. Range is 0 to 999 seconds. You may resend this command any time during the EAS transmission to reduce or extend the duration of the EAS message,	EASTIME=60↵
MS=	Music/Speech Switch	0 = Music 1 = Speech	MS=0↵ (Music)
DI=	Decoder Information	0 = Mono 1 = Stereo	DI=1↵ (Stereo)
TP=	Traffic Program	0 = Station does not carry traffic info 1 = Station broadcasts routine traffic info	TP=0↵ (No Traffic)
TA=	Traffic Alert ON-AIR NOW	0 = Flag Off 1 = Flag On (Flag valid only when TP=1)	TA=0↵ (No Traffic Alert)
TATIME=	TA Timeout	0 = Timer Off 1 - 255 = seconds between start of TA flag and automatic reset to OFF; 30 is recommended	TATIME=30↵ (Display TA for 30 Seconds)
AFxx=	Alternative Frequency List	Enter each AF in MHz 0 = Clear	AF1=88.1↵ AF1=0↵ (Clear AF1)
ECHO=	Terminal Echo	0 = no echo of sent data 1 = sent data echoed to Terminal window	ECHO=1↵ (Default Terminal Echo Characters)
HEAD=	Head Mode	0 = No Head 1 = Head This takes effect upon disconnect from terminal.	HEAD=1↵ (Default with Head)
TIME=	Time Data on RDS	Determines if time and date are transmitted in the RDS data stream; 0=No, 1=Yes.	TIME=0↵ (turn time transmission off))
RDS=	57kHz RDS Subcarrier	0 = RDS subcarrier On 1 = RDS subcarrier Off	RDS=1↵ (Default - Disabled)
RDSLEVEL=	Subcarrier Level	% Modulation (0…120) - 6% Default	RDSLEVEL=6.0↵ (Default - 6%)
TIMEOUT=	RDS terminal connection timeout	Timeout (in minutes) between last transmitted command and when the OPTIMOD disconnects automatically. Use it to allow the OPTIMOD to reconnect automatically if the terminal connection is lost temporarily. *This command only affects the current connection; you must reissue it each time you connect.*	TIMEOUT=15↵ (current connection stays up for 15 minutes) TIMEOUT=0↵ (no auto-disconnect occurs)
VER↵	RDS Welcome Header	{ 0=No, 1=Yes } Sets whether the RSD welcome header is sent to the Network client upon connection..	VER 0↵
INIT↵	Use System RDS parameters	Use RDS parameters from System.	INIT↵
SAVE↵	Save RDS parameters	saves the current RDS parameters to the currently active RDS control set (either the System Settings or the User Preset group) NOTE: Any changes will not appear in the PC application if it is open. You must reconnect to see the saved values.	SAVE↵

Table 3-1: RDS Terminal Commands

QUERIES		
[command] ↵	Any command and '?' returns the status of the encoder memory for that specific command..	PS?↵
ST↵	Returns all settings in encoder memory.	ST↵
HELP↵	Reports a list of available commands	HELP↵
TI↵	Returns the current time, as read from the OPTIMOD's real-time clock.	TIME=20:15:36 DATE=Jan.1, 2015
EASTIME ?	Returns the number of seconds remaining for the current on-air EAS alert transmission.	EASTIME ? ↵
EAS ?	Returns the EAS text currently in the RDS encoder's memory.	EAS ? ↵

Table 3-2: Queries

RESPONSES		
(Return Echo)	The command received was properly formatted and was received and executed by the encoder.	TATIME=30
Invalid Data Entered	Incoming data is not properly formatted and hence was not accepted and executed by the encoder.	Invalid Data Entered
(none)	Data that has been sent either has not reached the encoder or the encoder has no response for that command.	(none)

Table 3-3: Preset/Terminal RDS controls and defaults

Alternative Frequency Channel Numbers

MHz	CHAN	MHz	CHAN	MHz	CHAN	MHz	CHAN
87.6	1	92.7	52	97.8	103	102.9	154
87.7	2	92.8	53	97.9	104	103.0	155
87.8	3	93.9	54	98.0	105	103.1	156
87.9	4	93.0	55	98.1	106	103.2	157
88.0	5	93.1	56	98.2	107	103.3	158
88.1	6	93.2	57	98.3	108	103.4	159
88.2	7	93.3	58	98.4	109	103.5	160
88.3	8	93.4	59	98.5	110	103.6	161
88.4	9	93.5	60	98.6	111	103.7	162
88.5	10	93.6	61	98.7	112	103.8	163
88.6	11	93.7	62	98.8	113	103.9	164
88.7	12	93.8	63	98.9	114	104.0	165
88.8	13	93.9	64	99.0	115	104.1	166
88.9	14	94.0	65	99.1	116	104.2	167
89.0	15	94.1	66	99.2	117	104.3	168
89.1	16	94.2	67	99.3	118	104.4	169
89.2	17	94.3	68	99.4	119	104.5	170
89.3	18	94.4	69	99.5	120	104.6	171
89.4	19	94.5	70	99.6	121	104.7	172
89.5	20	94.6	71	99.7	122	104.8	173
89.6	21	94.7	72	99.8	123	104.9	174

MHz	CHAN	MHz	CHAN	MHz	CHAN	MHz	CHAN
89.7	22	94.8	73	99.9	124	105.0	175
89.8	23	94.9	74	100.0	125	105.1	176
89.9	24	95.0	75	100.1	126	105.2	177
90.0	25	95.1	76	100.2	127	105.3	178
90.1	26	95.2	77	100.3	128	105.4	179
90.2	27	95.3	78	100.4	129	105.5	180
90.3	28	95.4	79	100.5	130	105.6	181
90.4	29	95.5	80	100.6	131	105.7	182
90.5	30	95.6	81	100.7	132	105.8	183
90.6	31	95.7	82	100.8	133	105.9	184
90.7	32	95.8	83	100.9	134	106.0	185
90.8	33	95.9	84	101.0	135	106.1	186
90.9	34	96.0	85	101.1	136	106.2	187
91.0	35	96.1	86	101.2	137	106.3	188
91.1	36	96.2	87	101.3	138	106.4	189
91.2	37	96.3	88	101.4	139	106.5	190
91.3	38	96.4	89	101.5	140	106.6	191
91.4	39	96.5	90	101.6	141	106.7	192
91.5	40	96.6	91	101.7	142	106.8	193
91.6	41	96.7	92	101.8	143	106.9	194
91.7	42	96.8	93	101.9	144	107.0	195
91.8	43	96.9	94	102.0	145	107.1	196
91.9	44	97.0	95	102.1	146	107.2	197
92.0	45	97.1	96	102.2	147	107.3	198
92.1	46	97.2	97	102.3	148	107.4	199
92.2	47	97.3	98	102.4	149	107.5	200
92.3	48	97.4	99	102.5	150	107.6	201
92.4	49	97.5	100	102.6	151	107.7	202
92.5	50	97.6	101	102.7	152	107.8	203
92.6	51	97.7	102	102.8	153	107.9	204

Table 3-4:Alternative Frequency Channel Numbers:

PTY	Program Type – US	Program Type – EU
0	None	None
1	News	News
2	Information	Current Affairs
3	Sports	Information
4	Talk	Sports
5	Rock	Education
6	Classic Rock	Drama
7	Adult Hit Music	Culture
8	Soft Rock Music	Science
9	Top 40 Music	Varied
10	Country Music	Pop Music
11	Oldies Music	Rock Music
12	Soft Music	Easy Listening Music
13	Nostalgia Music	Light Classics Music
14	Jazz	Serious Classics Music
15	Classical Music	Other Music

PTY	Program Type – US	Program Type – EU
16	Rhythm and Blues Music	Weather
17	Soft R and B Music	Finance
18	Foreign Language	Children's Programs
19	Religious Music	Social Affairs
20	Religious Talk	Religion
21	Personality	Phone-In
22	Public Non-Commercial	Travel
23	College	Leisure
24	Spanish Talk	Jazz Music
25	Spanish Music	Country Music
26	Hip-Hop	National Music
27	(unassigned)	Oldies Music
28	(unassigned)	Folk Music
29	Weather	Documentary
30	Emergency Test	Alarm Test
31	Emergency!	Alarm!

Table 3-5: Program Type (PTY)

SCA/Subcarrier Phase Relationship

During stereo broadcast, the SCA subcarrier must be locked either in-phase or in quadrature to the third harmonic of the 19 kHz pilot tone. The tolerance of the phase angle is ±10° measured at the modulation input to the FM transmitter.

With no modulation other than the pilot tone, an oscilloscope trigged from the 19kHz pilot tone should display the waveform as seen to the right.

To minimize the amount of peak level that the RDS subcarrier adds to the composite baseband, your OPTIMOD's RDS generator is locked in quadrature with respect to the pilot tone, so the example shown here is for reference only.

If your transmission system is broadcasting stereo correctly, it will also correctly pass the phasing built into your OPTIMOD's SCA generator.

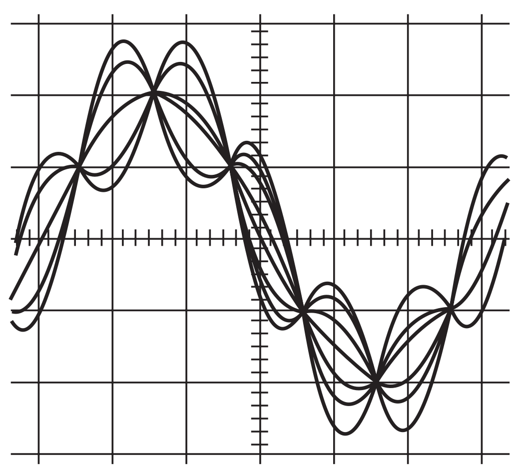


Table 3-6: Pilot/SCA Phasing Scope Trace

µMPX

µMPX in its simplest form is AES-EBU digital audio sampled at 192 kHz. AES sampling is done at 44.1 or 48 kHz in most all broadcast plants. µMPX allows the FM multiplex signal including baseband audio, pilot, stereo sub, RDS and

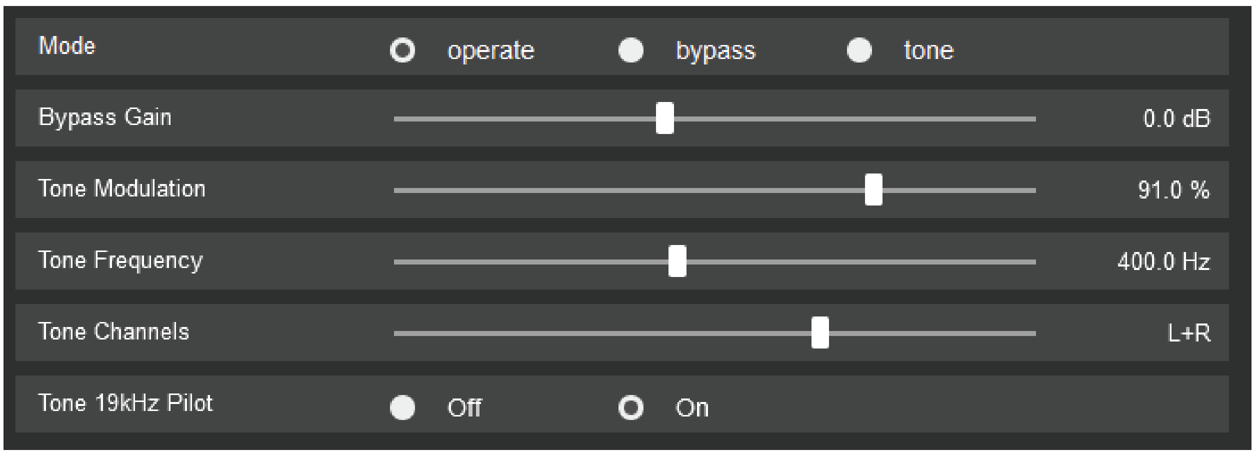
67 kHz subcarriers to be processed and transmitted either directly at the transmitter site or from the studio through a broadband wired or wireless IP STL.

As it is a composite signal, the adjustments are familiar. The 5750 offers up to four destination points for the µMPX signal. This can be useful when one processor is feeding multiple transmitter sites in a network.

You will need an audio transport method to transmit this signal and a receiver that will accept the Digital MPX signal. For more information on equipment that is compatible with µMPX, feel free to contact Orban support at support@orban.com

TEST

The Test Modes screen allows you to switch between OPERATE, BYPASS, and TONE. When you switch to BYPASS or TONE, the preset you have on air is saved and will be restored when you switch back to OPERATE.



SETUP: TEST				
PARAMETER LABELS	UNITS	DEFAULT	RANGE (CCW TO CW)	STEP

MODE	---	OPERATE	OPERATE, BYPASS, TONE	---
BYPASS GAIN	dB	0.0	–18 … +25	1
TONE FREQ	Hz	400	16, 20, 25, 31.5, 40, 50, 63, 80, 100, 125, 160, 200, 250, 315, 400, 500, 630, 800, 1000, 1250, 1600, 2000, 2500, 3150, 4000, 5000, 6300, 8000, 9500, 10000, 12500, 13586.76, 15000	LOG
TONE LVL	%	91	0 … 121	1
TONE CHAN	---	L+R	L+R, L–R, LEFT; RIGHT	---
PILOT	---	ON	ON, OFF	---

Table 3-7: Digital MPX

Facilities are self-explanatory, except for the following:

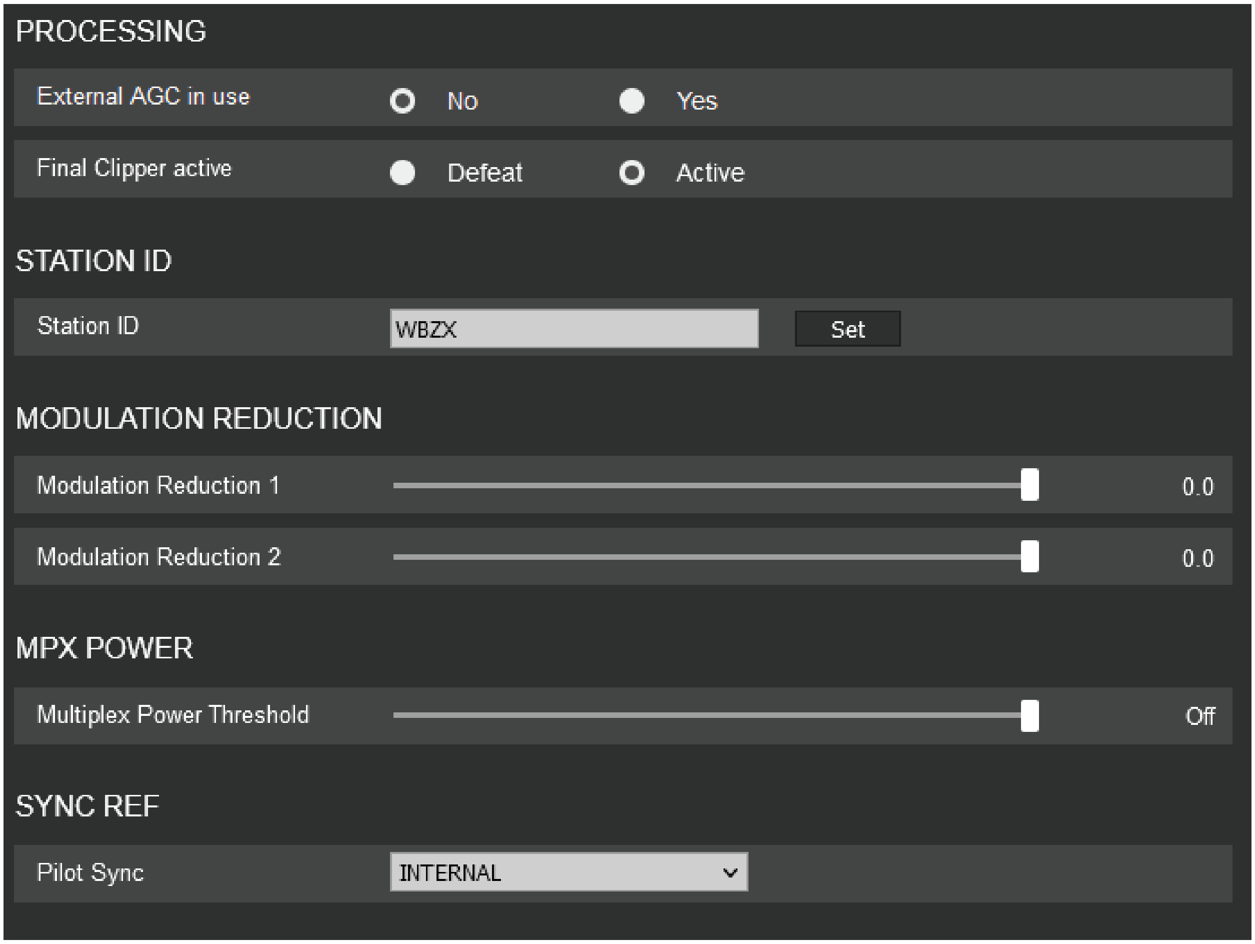
The TONE LVL control is calibrated under the assumption that the stereo pilot tone contributes 9% to the total modulation. Hence, a TONE LVL setting of 91% produces 100% modulation (91% + 9%).

Note that when the OPTIMOD is in TONE mode and its analog or digital L/R outputs are set to FLAT (instead of PRE-EMPH), these outputs apply 50µs or 75µs deemphasis to the output of the tone oscillator. Applying deemphasis this way is only correct if equipment downstream from the OPTIMOD, like a transmitter's stereo encoder, applies preemphasis before final transmission.

In BYPASS mode, preemphasis is still applied to the signal path. The BYPASS GAIN control calibration allows enough internal headroom to make swept frequency response measurements without internal clipping. When the BYPASS GAIN control is set to 0.0 dB and the AI REF VU control is set to 0.0 dBu, you will observe a gain of approximately –17 dB from the analog input to the analog output at 100 Hz. If the AO PRE-E control is set to PRE-E and the 5750 is configured for 75µs preemphasis, the gain from analog input to analog output will be approximately 0 dB at 15 kHz.

While this calibration may seem unintuitive, experience has shown that it greatly reduces calls to Orban customer service complaining that the frequency response of the transmission path is not flat when in fact the measurement in question was causing undetected clipping at high frequencies due to preemphasis.

UTILITY



The UTILITY tab allows you to customize settings in the 5750 to adapt to your airchain and regulatory requirements.

PROCESSING:

EXTERNAL AGC IN USE: If you are using an external AGC ahead of your STL, you will most likely want to select YES as you don't want dual AGC action, which will cause your audio to sound unnaturally dense. If you are not using an external AGC, this should be set for NO.

FINAL CLIPPER ACTIVE: You can defeat the FM clipper in the 5750 if so desired. This should only be done for specialized applications. For the processor to be used correctly, this control should remain on.

STATION ID: You can set the ID of your station here.

MODULATION REDUCTION: In the United States, F.C.C. Rules permit you to add 0.5% modulation for every 1% increase in subcarrier injection. For example, if your subcarrier injection totals 20%, you can set the total modulation to 110% (82.5 kHz deviation). The 5750 has the ability to reduce audio modulation to compensate for subcarriers.

The advantage of using the modulation reduction function is that the pilot injection stays constant when the audio modulation is reduced. However, using the modulation reduction function is slightly inconvenient because it requires programming and activating at least one 5750 GPI input. If you have the same subcarrier injection at all times, a more convenient alternative is to set the desired modulation level by using the Composite Level control(s). Then turn up the pilot injection control until the injection equals 9% modulation.

To comply with FCC Rules, set the modulation reduction to one-half the injection of the associated subcarrier. For example, if your subcarrier injection totals 20% from two 10% subcarriers, set Modulation Reduction 1 TO “5%” and Modulation Reduction 2 to 5%. This will reduce your audio modulation to 90% (100% – 5% – 5%). When you add back the 20% modulation due to the subcarriers, you get the required 110% total modulation.

The Modulation Reduction function is active as long as signal is applied to its associated GPI input.

MULTIPLEX POWER THRESHOLD: Set the MPX Power Threshold control to the target loudness specified by your country’s governing authority.

The BS.1770 safety limiter and meter have been included to allow your OPTIMOD to comply with government regulations in countries that enforce BS.1770-based loudness control (per EBU Recommendation R 128) in analog FM transmission. Unless your country requires this, leave the BS.1770 Safety Limiter off.

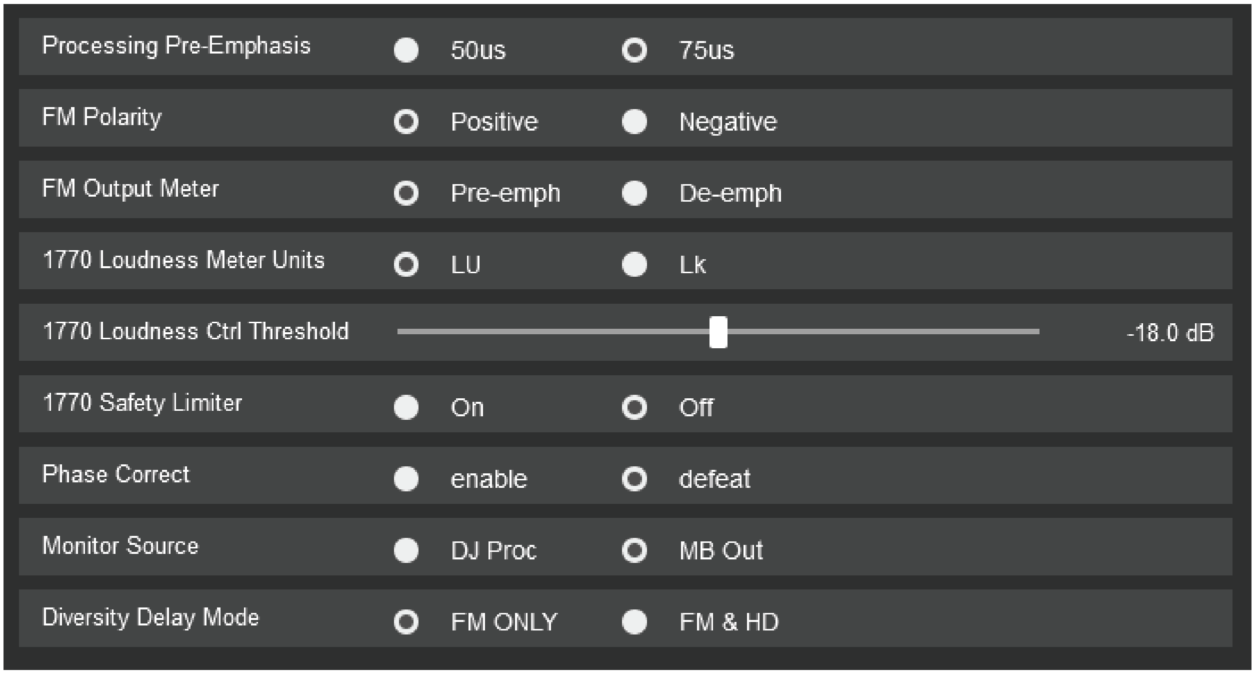
The analog-chain BS.1770 safety limiter and meter are calibrated with reference to 100% modulation with 50µs preemphasis (75 kHz deviation) per section 5.9 of EBU Tech 3344 (“Practical guidelines for distribution systems in accordance with EBU R 128”).

The BS.1770 Safety Limiter is usually used in conjunction with the BS.412 Multiplex Power Limiter. The processing chain is configured as follows:

MPX Offset Control → BS.1770 controller (0 to 3dB GR) → Peak Limiter →

BS.1770 controller (>3dB GR) → BS.412 Controller

SYNC REFERENCE: there are many options to choose from for you piolt sync reference. Default is Internal.



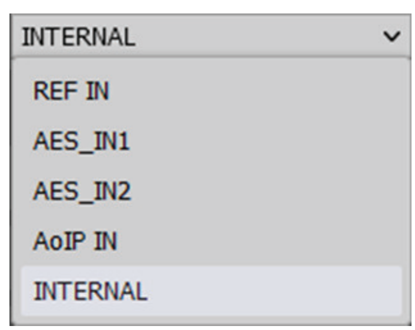
PROCESSING PRE-EMPHASIS: Sets the pre-emphasis that is required for your location. Most of the Americas use 75µs, much of Europe uses 50µs. If you are unsure, consult your local regulatory body.

FM POLARITY: Makes positive the output polarity of the 5750's FM analog channel processing. In HD Radio installations, this command is useful when switching the 5750 between transmitters if the transmitters' exciters produce opposite FM modulation polarities when driven by identical digital audio input signals. This setting affects any output emitting the analog FM processed signal, including the composite output.

FM OUTPUT METER: Selects whether the output meter displays with or without pre-emphasis in mind.

1770 LOUDNESS METER UNITS: Changes the value of the meter from LU to Lk.

1770 LOUDNESS CONTROL THRESHOLD: sets the threshold of the BS.1770 Safety Limiter and the calibration of the BS.1770 loudness meter.



When the BS.1770 Safety Limiter is OFF, the BS.1770 Loudness Control Threshold sets the calibration of the BS.1770 Loudness Meter, such that "0" LK/LU on the meter corresponds to the loudness appearing at the Digital Output assigned to "HD." This calibration is only correct if the Digital Output 100% Peak Level control is set to 0 dBFS.

When the BS.1770 Safety Limiter is ON, this calibration is correct regardless of the setting of the Digital Output 100% Peak Level control.

1770 SAFETY LIMITER: Turns on or off the 1770 safety limiter. This control should remain off unless it is required by your local regulatory body. Setting this control to on will greatly reduce loudness in regions that do not use this feature.

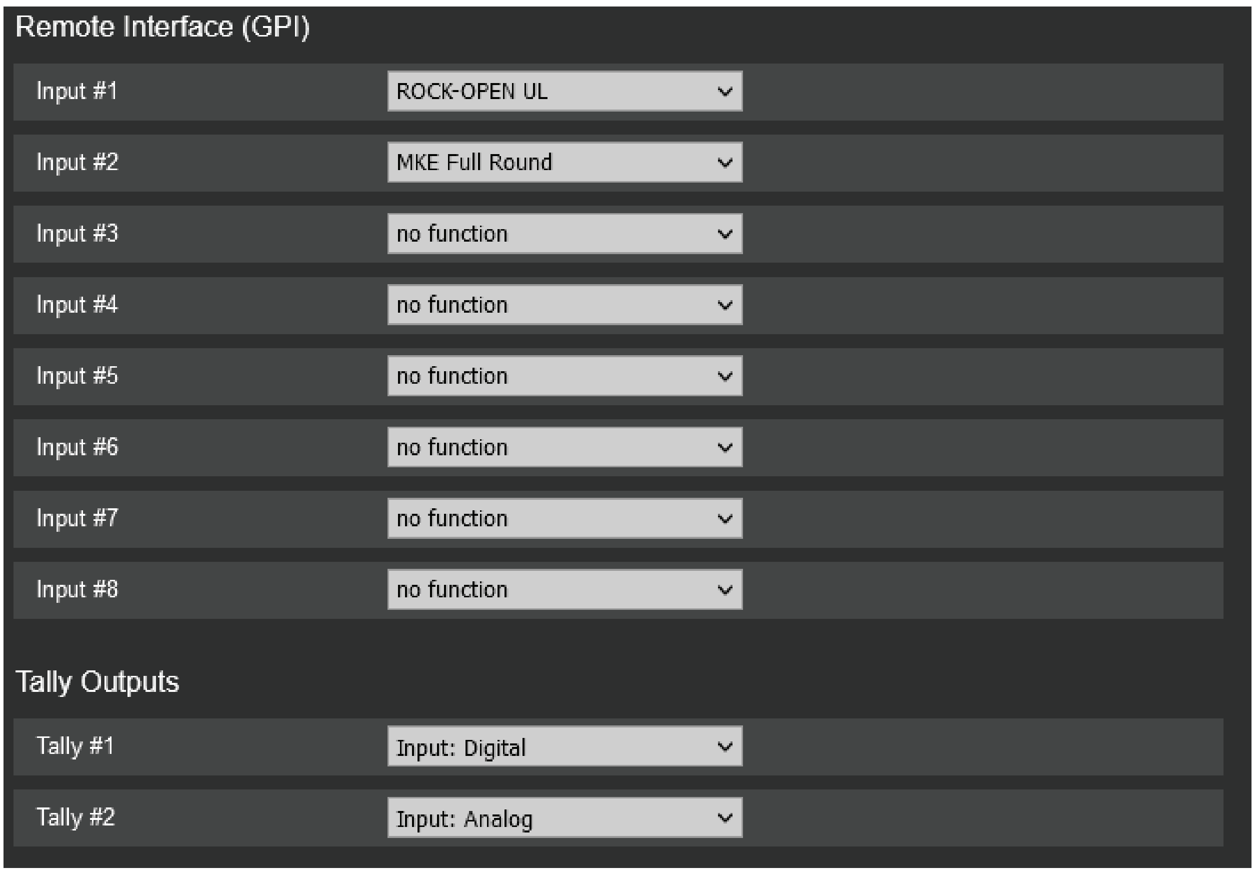
PHASE CORRECT: The phase correct option will “repair” asymmetrical voice audio so that it is more symmetrical and easier to process. Orban’s recommended setting is on.

MONITOR SOURCE: Changes the low latency monitor from the DJ Processing option to the output of the multiband processing.

DIGITAL INPUT ANALOG FAILBACK: If the AES signal is lost (or if audio falls below the user defined threshold and user defined time), the 5750 will switch to an active analog input with audio present.

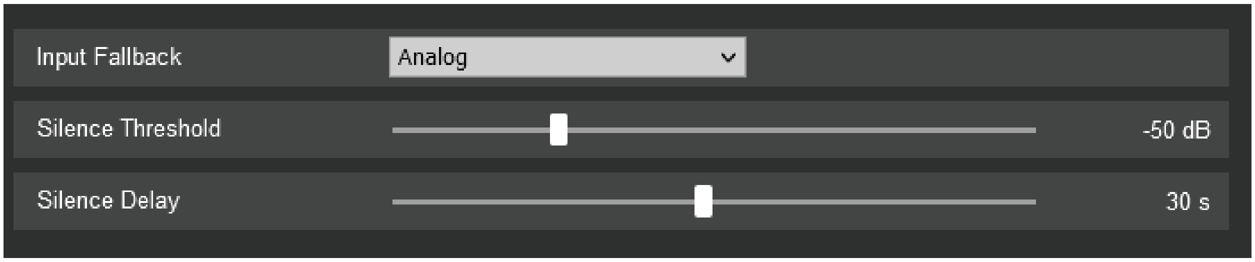
DIVERSITY DELAY MODE: Selects if the processor had a delay path in the FM or FM+HD audio path.

REMOTE INTERFACE



The GPI/GPO closures on the DB cable on the rear of the 5750 allow you to take different presets via contact closure. It also allows you to select inputs in case of failure. In the example above, Inputs #1 and #2 take different presets which, when set up with a closure, can match different types of programming on your station. The Tally Outputs allow the taking of a main and back up source. In the example above, Tally #1 takes the Digital AES input, while Tally #2 triggers the Analog backup.

SILENCE DETECT



- **INPUT FALLBACK –** When the main input fails, the 5750 will switch to the selected input on this screen. The options are ANALOG, Digital 1, Digital 2, AoIP 1, AoIP 2, or Stream.

- **SILENCE THRESHOLD –** Sets the level at which all input audio below will start the silence delay clock. If audio remains below this level longer than the silence delay time is set, the 5750 will switch to the selected backup audio source. The level range is -20dB to -60dB. For classical and jazz formats, a lower threshold is suggested.

- **SILENCE DELAY –** Adjustable from 2 seconds to 60 seconds (1 minute). If audio remains below the level selected in the SILENCE THRESHOLD setting for longer than the period of time set here, the 5750 will switch to the INPUT FALLBACK source.

If your input levels and modulation are set, it is time to pick a preset and adjust the sound. The following pages will guide you thru the various adjustments and what they mean to achieving your signature sound. You can follow along with the processing tabs from left to right.

PROCESSING PARAMETERS

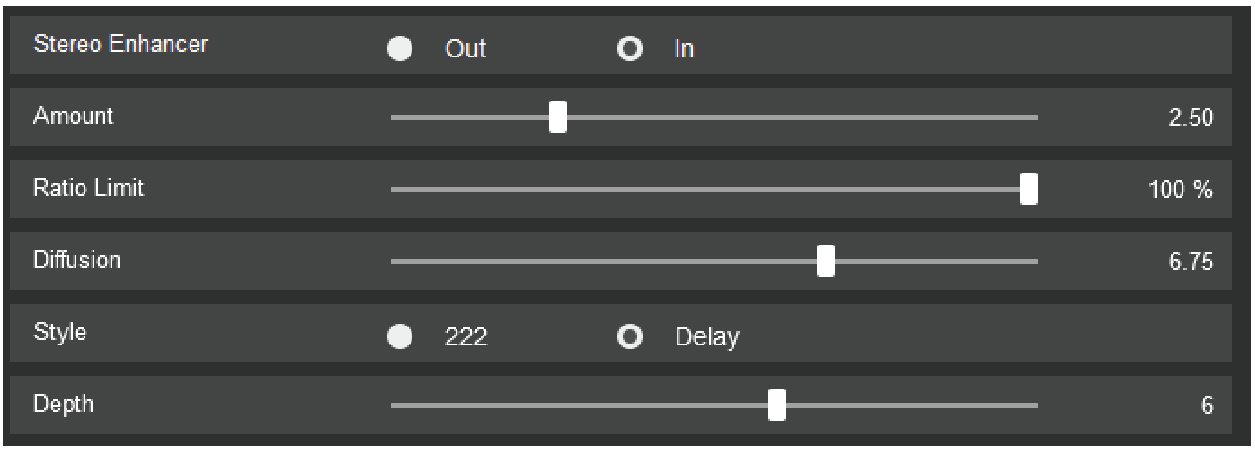
The processing parameters tab opens up a sidebar which allows the user the ability to fine tune each stage of the audio. The following section of this manual will walk you through the steps and help you understand what the various controls mean.

LESS-MORE

As you increase the setting of the Less-More control, the air sound will become louder, but (as with any processor) processing artifacts will increase. Please note that the highest Less-More setting is purposely designed to cause unpleasant distortion and processing artifacts! This helps assure you that you have chosen the optimum setting of the Less-More control, because turning the control up to this point will cause the sound quality to become obviously unacceptable.

You need not (in fact, cannot) create a sound entirely from scratch. All User Presets are created by modifying Factory Presets, or by further modifying Factory Presets that have been previously modified with a Less-More adjustment. It is wise to set the Less-More control to achieve a sound as close as possible to your desired sound before you make further modifications at the Advanced Modify level. This is because the LESS-MORE control gets you close to an optimum trade-off between loudness and artifacts, so any changes you make are likely to be smaller and to require resetting fewer controls.

STEREO ENHANCER:



The 5750 provides two different stereo enhancement algorithms. The first is based on Orban's patented analog 222 Stereo Enhancer, which increases the energy in the stereo difference signal (L–R) whenever a transient is detected in the stereo sum signal (L+R). By operating only on transients, the 222 increases width, brightness, and punch without unnaturally increasing reverb (which is usually predominantly in the L–R channel).

Gating circuitry detects "mono" material with slight channel or phase imbalances and suppresses enhancement so this built-in imbalance is not exaggerated.

It also allows you to set a "width limit" to prevent over-enhancement of material with significant stereo content, and will always limit the ratio of L–R / L+R to unity or less.

The second stereo enhancement algorithm is based on the well-known "Max" technique. This passes the L–R signal through a delay line and adds this decorrelated signal to the unenhanced L–R signal. Gating circuitry similar to that used in the "222-style" algorithm prevents over-enhancement and undesired enhancement on slightly unbalanced mono material.

You may choose to have the stereo enhancer OUT of the signal path or IN.

- **AMOUNT** – Sets the maximum special enhancement
- **RATIO LIMIT** – Limits the sum/difference ratio to help prevent multipath in receivers. However, if the original program material exceeds this limit with no enhancement, the enhancer will not reduce it.
- **DIFFUSION** – Sets the amount of delay to the L-R enhancement (available only in the DELAY option)
- **STYLE** – Switches between the 222 option and the DELAY options listed above.

- **DEPTH** – Controls the amount of L-R delay (available only in the DELAY option)

AGC (Automatic Gain Control)

The AGC is a two-band device, using Orban's patented "master / bass" band coupling. It has an additional important feature: target-zone gating. If the input program material's level falls within a user-settable window (typically 3 dB), then the release time slows to a user-determined level. It can be slow enough (0.5 dB/second) to effectively freeze the operation of the AGC. This prevents the AGC from applying additional, audible gain control to material that is already well controlled. It also lets you run the AGC with fast release times without adding excessive density to material that is already dense.

The AGC contains a compression ratio control that allows you to vary to ratio between 2:1 and essentially 20:1. Lower ratios can make gain riding subtler on critical formats like classical and jazz.

The AGC has its own silence-gating detector whose threshold can be set independently of the silence gating applied to the multiband compressor.

- **AGC ON/OFF** – Lets you decide whether the AGC is operating in the signal chain or is bypassed. If you have a leveler ahead of the 5750, it is best to turn off the AGC.

- **AGC DRIVE** – Sets the applied level to the AGC, determining the amount of overall gain reduction. This also adjusts the "idle gain"—the amount of gain reduction in the AGC section when the structure is gated. (It gates whenever the input level to the structure is below the threshold of gating.)

The total amount of gain reduction in the Five-Band structure is the sum of the gain reduction in the AGC and the gain reduction in the multiband compressor.

The total system gain reduction determines how much the loudness of quiet passages will be increased (and, therefore, how consistent overall loudness will be). It is determined by the setting of the AGC Drive control, by the level at which the console VU meter or PPM is peaked, and by the setting of the MB Drive (compressor) control.

- **AGC CROSSOVER** – Selects between LINEAR, LINEAR WITH NO DELAY or ALLPASS.

- **AGC MASTER ATTACK** – Controls the rate of gain reduction in the master band. The faster the time (in seconds) the faster the attack.

- **AGC BASS ATTACK** – Same as the MASTER ATTACK but applies to the bass band.

- **AGC RELEASE** – Controls the rate of gain increase in the master band. Settings with higher numbers will result in faster recovery times.

- **AGC BASS RELEASE** – Same as AGC RELEASE but applies to the bass band.

- **AGC BASS THRESHOLD** – Determines the threshold of the operation of the bass band. Higher numbers will allow the bass band to processor lower audio passages.

- **AGC MASTER DELTA THRESHOLD** – Allows you to offset the difference between compression thresholds of the sum and difference channels in the master band.

- **AGC BASS DELTA THRSHOLD** – Same as AGC MASTER DELTA THRESHOLD but applies only to the bass band.

- **AGC RATIO** – Determines the ratio of the AGC. Infinity:1 applies the most control over the output of the AGC block. 2:1 applies the least amount of control over the output of the AGC block. The compression ratio is the ratio between the change in input level and the resulting change in output level, both measured in units of dB.

  Previous Orban AGCs had compression ratios very close to 20:1, which produces the most consistent and uniform sound. However, the 5750 compressor can reduce this ratio to as low as 2:1. This can add a sense of dynamic range and is mostly useful for subtle formats like classical and jazz.

- **AGC GATE THRESHOLD** – Determines the lowest input level the AGC will detect as program content. All audio below this level will be ignored by the AGC.

- **AGC WINDOW SIZE** – Determines the size of the “target zone” window in the AGC. (The Bass band is not windowed.)

- **AGC WINDOW RELEASE** – Determines the rate of gain increase while program audio is within the WINDOW SIZE.

- **AGC STEREO COUPLING** – Determines the gain difference of the Left and Right channels. OFF means the channels are fully independent. 0dB means the left and right channels apply the same amount of gain reduction.

- **AGC BASS COUPLE** – Determines the maximum amount of gain difference between the MASTER and BASS bands.

- **AGC MATRIX** – Sets the AGC to operate in STEREO (L/R) or SUM/Difference (L-R) mode.

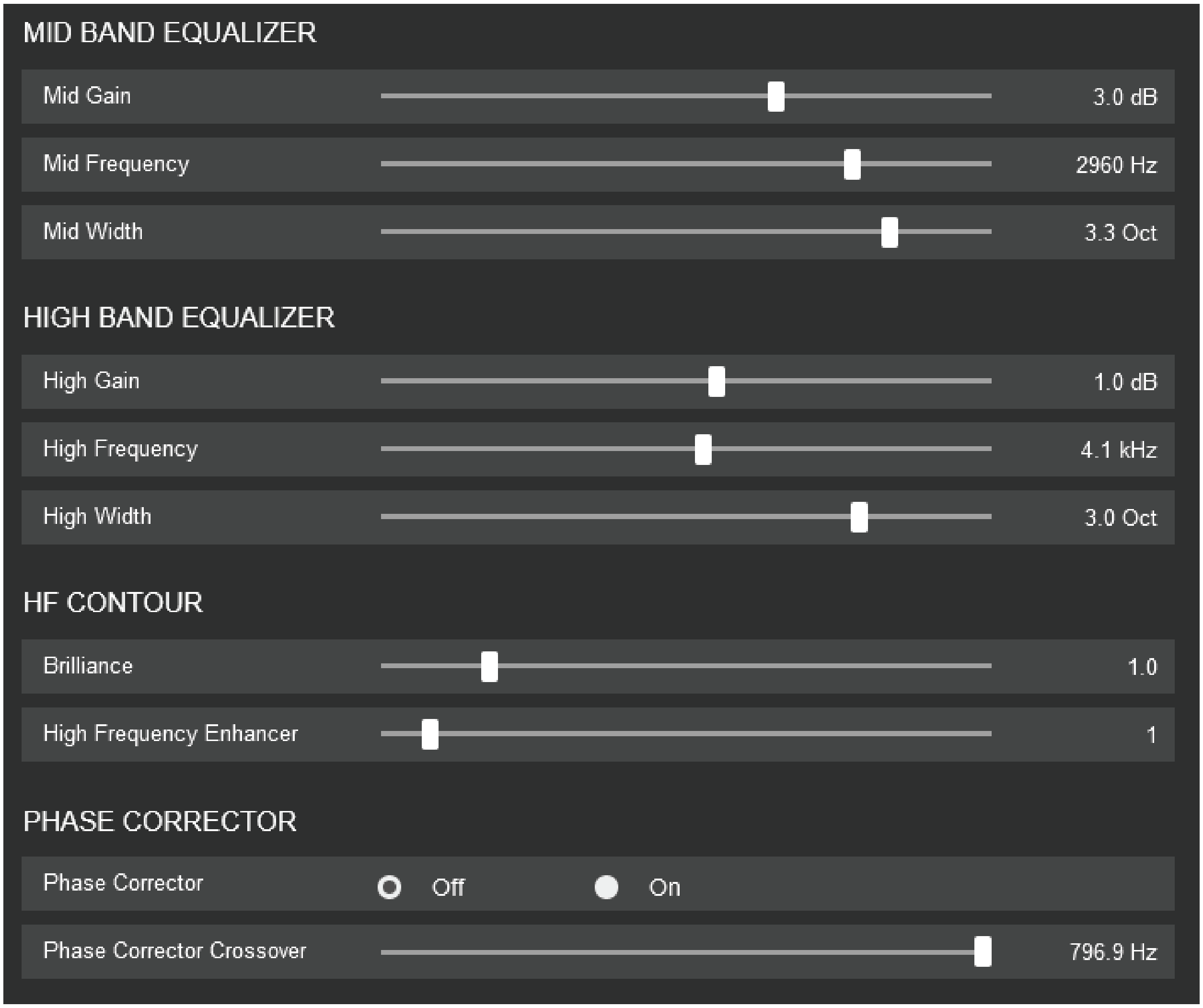
EQUALIZER (EQ)

The 5750 has steep-slope bass shelving equalizer and three bands of fully parametric bell-shaped EQ.

You can set the slope of the LF shelving EQ to 6, 12, or 18 dB/octave and adjust the shelving frequency. The bass slope can be 6, 12 or 18dB/octave.

The PHASE ROTATOR corrects asymmetrical voice energy to make it easier for the 5750 to process dry voice. Orban recommends this feature should be left on.

The **DJ BASS BOOST** (5 Band Option Only) Sets the amount of bass boost on live voice to “fill in” with more warmth to voice only audio.



In its default Off position, it causes the gain reduction of the lowest frequency band to move quickly to the same gain reduction as its nearest neighbor when gated. This fights any tendency of the lowest frequency band to develop significantly more gain than its neighbor when processing voice because voice will activate the gate frequently. Each time it does so, it will reset the gain of the lowest frequency band so that the gains of the two bottom bands are equal and the response in this frequency range is flat. The result is natural-sounding bass on male voice.

If you like a larger-than-life, "chesty" sound on male voice, set this control away from Off.

When so set, gating causes the gain reduction of the lowest frequency band to move to the same gain reduction (minus a gain offset equal to the numerical setting of the control) as its nearest neighbor when gated. You can therefore set the maximum gain difference between the two low frequency bands, producing considerable dynamic bass boost on voice.

The difference will never exceed the difference that would have otherwise occurred if the lowest frequency band was independently gated. If you are familiar with older Orban processors like the 8200, this is the maximum amount of boost that would have occurred if you had set their DJ Bass Boost controls to On.

The amount of bass boost will be highly dependent on the fundamental frequency of a given voice. If the fundamental frequency is far above 100Hz, there will be little voice energy in the bottom band and little or no audio bass boost can occur even if the gain of the bottom band is higher than the gain of its neighbor. As the fundamental frequency moves lower, more of this energy leaks into the bottom band, and you hear more bass boost. If the fundamental frequency is very low (a rarity), there will be enough energy in the bottom band to force significant gain reduction, and you will hear less bass boost than if the fundamental frequency were a bit higher.

The 5750's bass, midrange, and high frequency parametric equalizers have curves that were modeled on the curves of Orban's classic analog parametrics (like the 622B), using a sophisticated, proprietary optimization program. The curves are matched to better than 0.15 dB. This means that their sound is very close to the sound of an Orban analog parametric. They also use very high quality filter algorithms to ensure low noise and distortion.

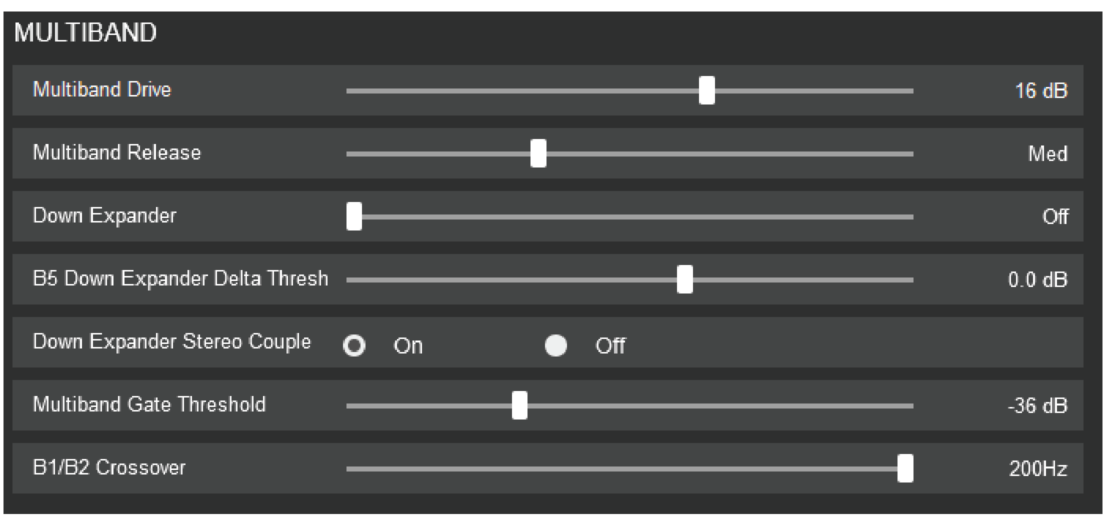
The 5750 HF Enhancer is a program-controlled HF shelving equalizer.

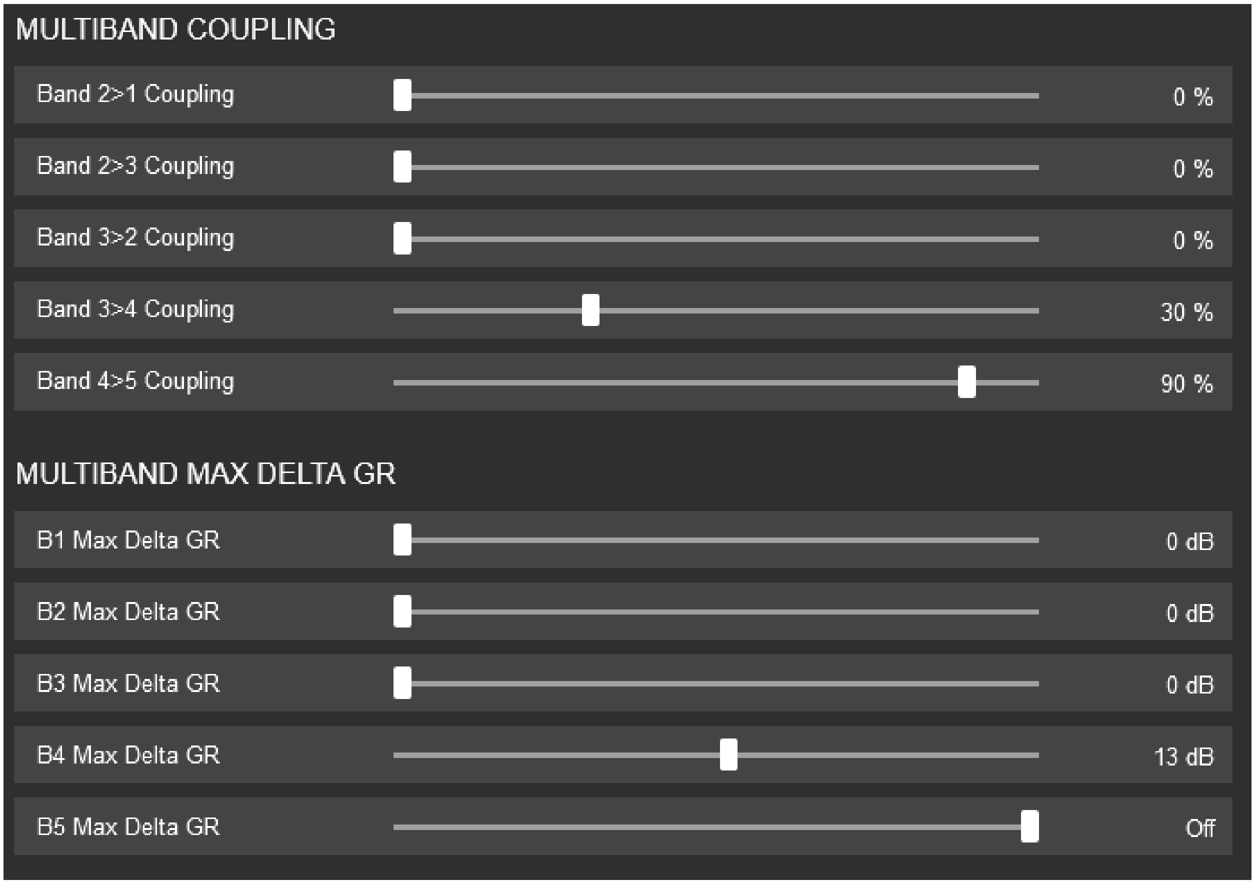
It intelligently and continuously analyzes the ratio between broadband and HF energy in the input program material and can equalize excessively dull material without over-enhancing bright material. It interacts synergistically with the five-band compressor to produce sound that is bright and present without being excessively shrill.

HF ENH ("High Frequency Enhancer") is a program-adaptive 6 dB/octave shelving equalizer with a 4 kHz turnover frequency.

It constantly monitors the ratio between high frequency and broadband energy and adjusts the amount of equalization in an attempt to make this ratio constant as the program material changes. It can therefore create a bright, present sound without over-equalizing material that is already bright.

MULTIBAND





The multiband compressor/limiter can be operated in five-band or two-band mode. The 5750 controls high frequencies with distortion-canceled clipping and, in all but 5-band MX presets, with a high frequency limiter as well. The clipper operates at 256 kHz-sample rate and is fully anti-aliased.

Usually, the gain reduction in band 5 is slaved to the gain reduction in band 4 (as determined by the setting of the B4 > B5 COUPLE control); these bands are only independent from the viewpoint of the downward expander and multiband clippers. However, a high frequency limiter causes additional gain reduction in band 5 when band 5 multiband clipping alone would be insufficient to prevent HF distortion. The HF limiter uses a sophisticated analysis of the signal conditions in the 5750's clipping system to do this.

Except in MX presets, a clipper, embedded in the crossover, protects bands 1 and 2 from transient overshoot. This clipper has a shape control, allowing you to vary the "knee" of its input/output transfer curve from hard (0) to soft (10). Instead of a clipper, MX presets use a sophisticated bass pre-limiter located immediately before the system's main distortion-controlled clipper.

In non-MX presets, the multiband compressor/limiter offers look-ahead compression to minimize overshoot and its associated clipping distortion. This look-ahead functionality can be turned on or off manually, or the 5750's speech/music detector can activate it automatically.

The Ultra-low Latency structure does not offer compressor look-ahead.

- **MULTIBAND DRIVE** – Sets the level applied to the multiband compressor and thus its average level of gain reduction.
- **MULTIBAND RELEASE** – Sets the release rate for the multiband compressors.
- **DOWN EXPANDER** – Downward Expander. Sets the threshold that triggers single ended noise reduction.
- **B5 DOWN EXPANDER STEREO COUPLE** – Fine tunes the downward expansion to be coupled or independent stereo.
- **MULTIBAND GATE THRESHOLD** – Determines the lowest input level that will be recognized as program.
- **B1/B2 CROSSOVER** – Sets the crossover frequency between bands 1 and 2.
- **BAND COUPLING** – There are 5 controls for band coupling. These controls determine the amount one band will track another. In the scenario "Band X>Y Coupling", "X" will always follow the gain reduction of "Y".
- **MAX DELTA GR** – Governs the mount of independent stereo gain reduction in each band. 0 is coupled (the channel with the most gain will reduce both channels). OFF means the channels are never coupled.

  Take care to watch the gain reduction meters while adjusting this stage. In the HTML5 application you will see the differences between left and right play out in the gain reduction of the metering.

  The same holds true for coupling. If you notice too much bass in your audio (for example), you may notice only a little gain reduction in Band 1 vs Band 2. By increasing the coupling between Band 1 and Band 2 (B2>B1 control) you will see more gain reduction in Band 1 relative to Band 2. Adjust to the desired amount of bass and set. You will now have a more consistent tonal balance between Band 1 and 2 suitable for your format. The same can be applied to the other bands of gain reduction as well.

COMPRESSORS

You can offset the adjustments made in MULTIBAND here. This will help fine tune the OPTIMOD to your tastes.

- **MULTIBAND ATTACK** – Sets the rate of speed with which the band you are setting will decrease gain at its input. Each Band (B1-B5) has it's own control. Longer attack times for bands 3 and 4 will be most transparent for horns and sustained vocals. Shorter attack times in Bands 1 and 5 will be more controlling over bass and highs.

- **MULTIBAND DELTA RELEASE** – Offsets the release of a particular band from the master release control

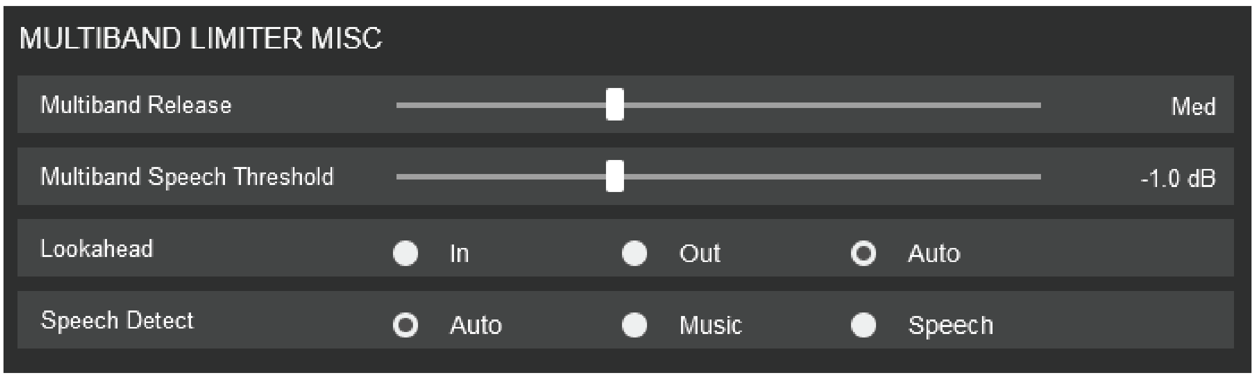
- **MULTIBAND COMPRESSOR THRESH** – Sets the compressor threshold in units of dB below the multiband clipper.

- **MULTIBAND LIMITER ATTACK** – Governs the rate of speed in which the multiband limiter will react to program audio on its input.

SPEECH MODE

You can set many of the processing parameters separately for speech signals, as detected by the 5750's speech/music detector. This allows you to tune the processing separately for speech and music.

A SPEECH DETECT control allows you to force the 5750 into Music mode, overriding the Speech/Music detector. This control is contained in the processing preset.



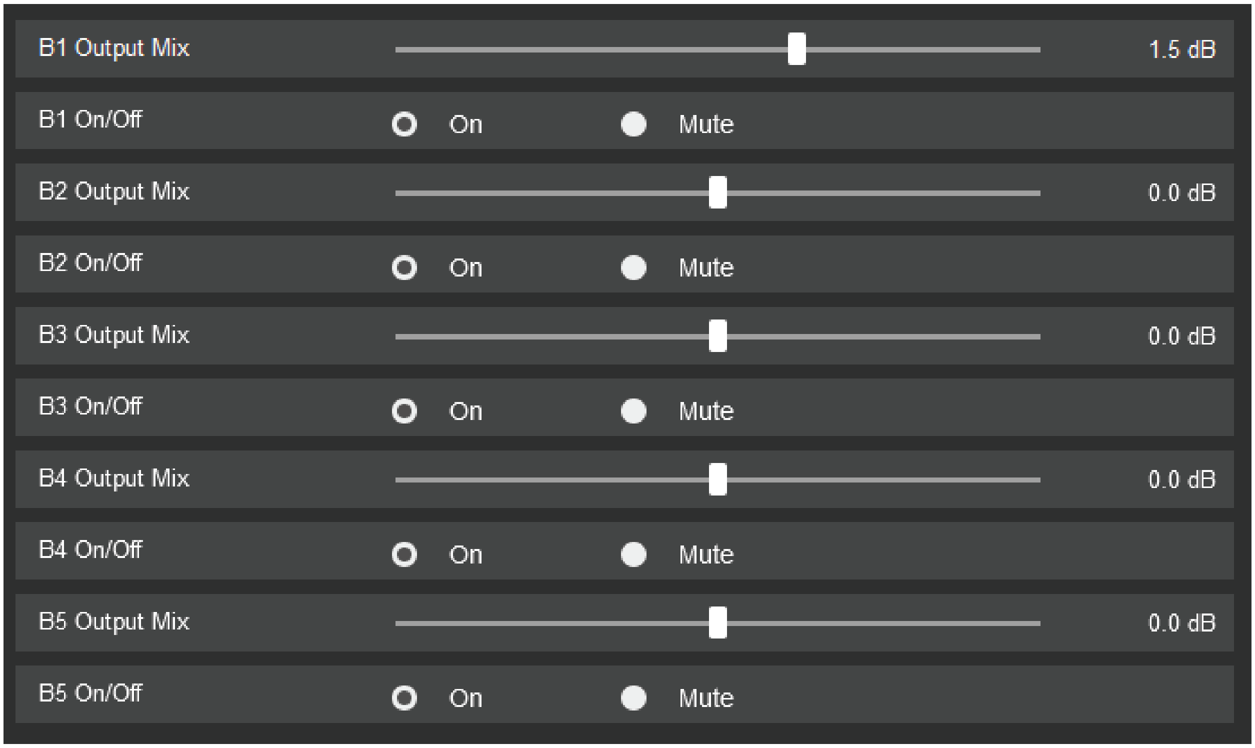
- **MULTIBAND ATTACK** – During speech, Sets the rate of speed with which the band you are setting will decrease gain at its input. Each Band (B1-B5) has it's own control.

- **MULTIBAND SPEECH THRESH** - During speech, sets the compressor threshold in units of dB below the multiband clipper.

- **LOOKAHEAD** – During speech, Activates or Deactivates the look ahead function of the limiter.

- **SPEECH DETECT** – Can force the 5750 into MUSIC only, SPEECH only or AUTO detect mode.

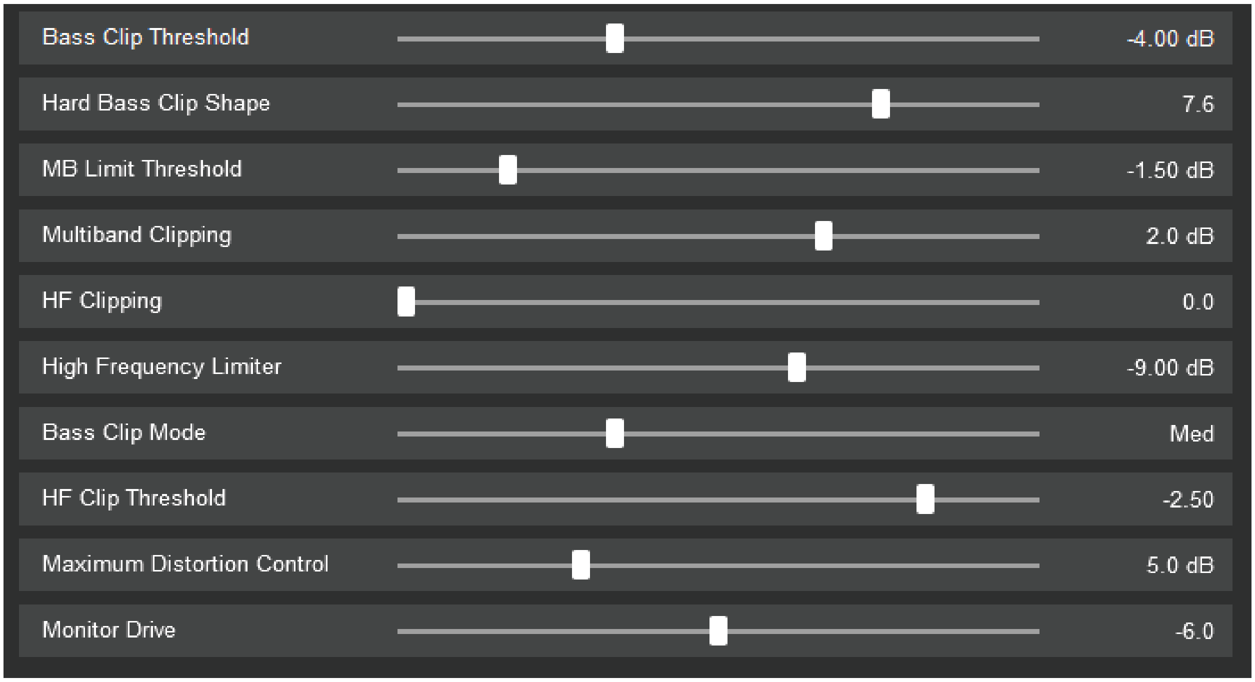
BANDMIX



Think of this section as a graphic equalizer. You can carefully add or reduce the output of each band of processing. Careful adjustments should be made here as any additional audio gain will drive the main clipper in the OPTIMOD. If you find increasing the gain of any channel adds unwanted distortion, adjust the attack time of the 5 band compressor.

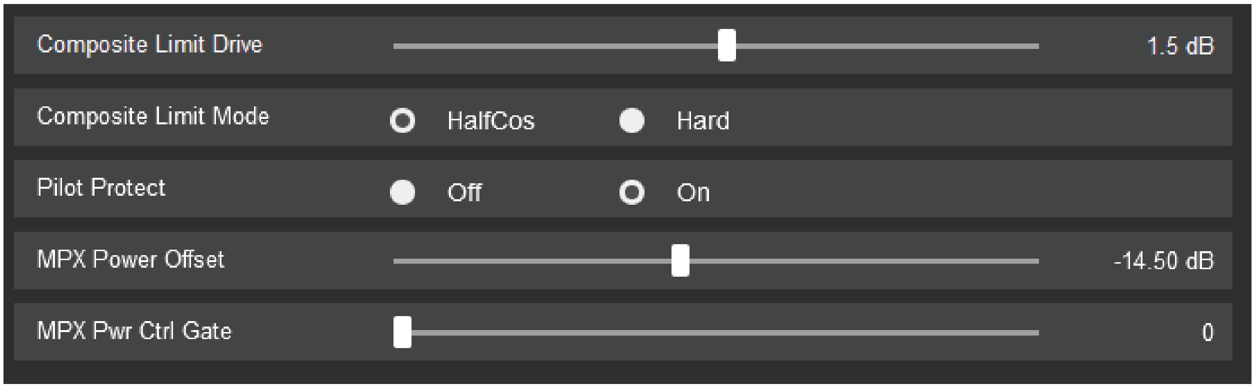
The ON/OFF control turns on and off the output of each band. Useful in some aspects, turning off a band of processing will unmask distortion from neighboring bands, especially with aggressive presets. Orban's intelligent clipper design prevents that type of distortion from being audible when all bands are operating, so it is best to adjust with all 5 options ON.

DISTORTION



- **BASS CLIP THRESHOLD** - Sets the embedded bass clipper threshold in dB below the final clipper.
- **HARD BASS CLIP SHAPE –** Changes the shape of the knee of the gain curve of the bass clipper.
- **MB (Multiband) LIMIT THRESHOLD –** Sets the threshold of the clipping distortion controller**.**
- **MULTIBAND CLIPPING –** Controls the amount of signal applied to the multiband clippers. Higher values mean less multiband compression and more clipping activity in the multiband.
- **HF (High Frequency) CLIPPING –** Normally set to 0, higher values will allow more brightness and less intelligent HF distortion control.
- **HIGH FREQUENCY LIMITER –** Sets the amount of additional gain in Band 5 of the multiband section to prevent high frequency distortion in the final clipper.
- **BASS CLIP MODE –** Sets the hardness of the embedded bass clipper
- **HF (High Frequency) CLIP THRESHOLD –** Sets the threshold of the distortion cancelled clipper in the HF limiter. Higher numbers will yield more brightness at the expense of some distortion tradeoff.
- **MAXIMUM DISTORTION CONTROL** – Limits the maximum amount of final clipper drive gain reduction (in dB) that the clipping distortion controller can apply.

DISTORTION



- **COMPOSITE LIMIT DRIVE –** Sets the amount of peak limiting in the composite limiter.

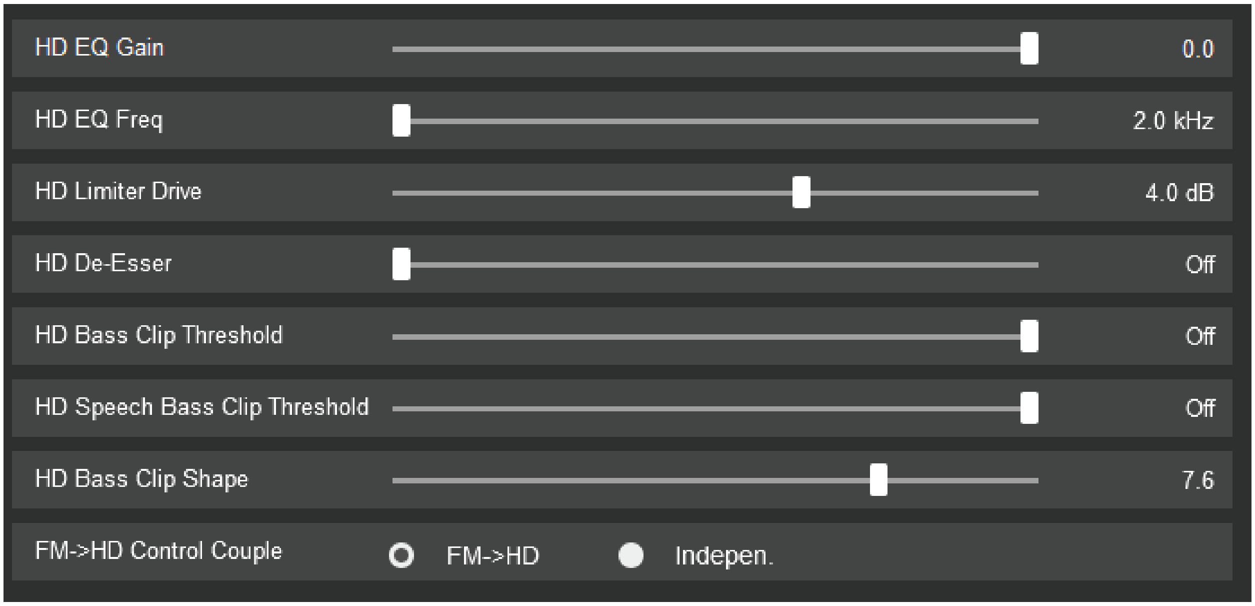
- **COMPOSITE LIMIT MODE –** Half cosine provides better separation and maintains stereo imaging. Hard mode provides a brighter sound as it creates waveforms closer to square waves.

- **PILOT PROTECT –** Protects the 19kHz pilot from composite processor produced harmonics.

- **MPX POWER OFFSET –** Reduces peak limiter drive to offset audible side effects of ITU412 controller.

- **MPX POWER CTRL GATE –** Sets the level below which gating occurs, as well as the release times above and below the threshold. Higher-numbered presets provide higher thresholds and slower release times.

HD LIMITING



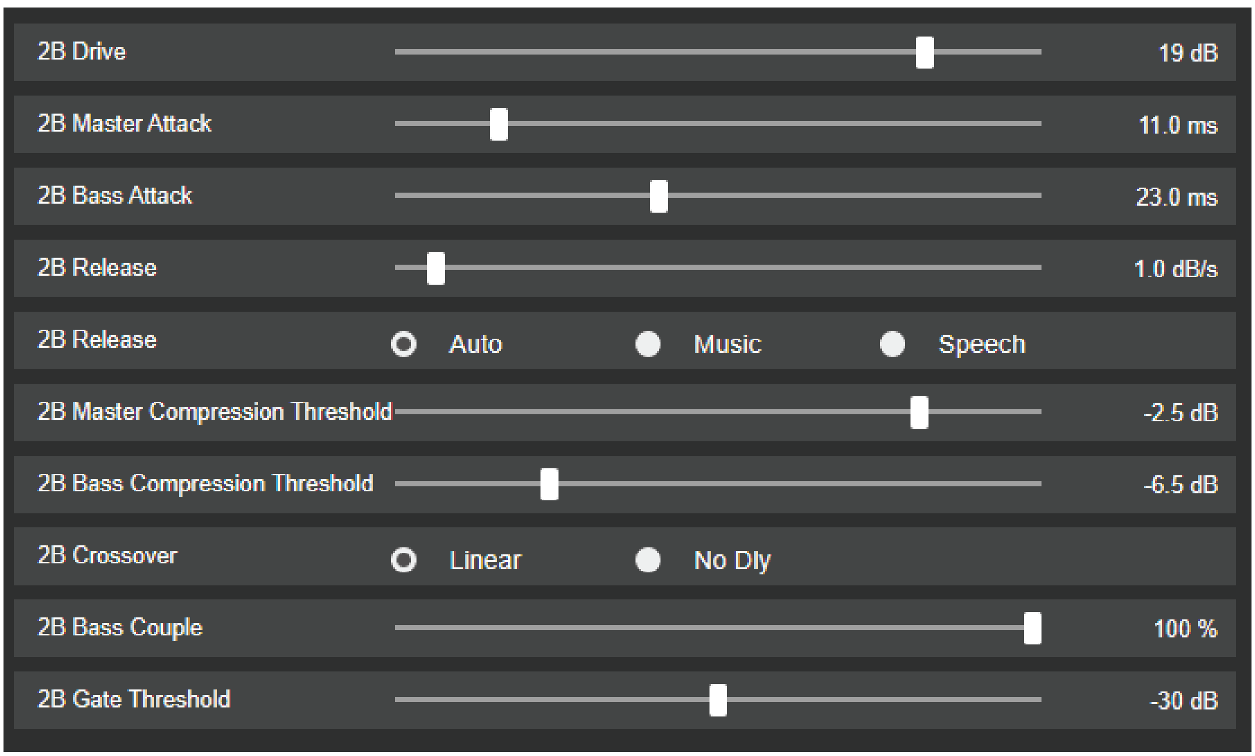
- **HD EQ GAIN** – Determines the depth of high frequency shelving equalization produced by the parametric HF shelving equalizer.
- **HD EQ FREQ (Frequency)** – Sets the corner of the parametric high frequency shelving equalizer
- **HD Limiter Drive** – Sets the drive level to the HD limiter.
- **HD De-Esser** – Reduces sibilance (‘ess sounds”) that might otherwise cause unwanted distortion,’
- **FM>HD Control Couple** – Links the HD controls to their corresponding FM controls when coupled.

2 BAND PROCESSING

In addition to five-band processing, suitable for pop music and talk formats, the 5750 offers a very high-quality two-band algorithm. This is phase-linear and features the same AGC as the five-band processor, followed by a two-band processor with look-ahead limiting. Sophisticated multiband high frequency limiting and distortion-cancelled clipping complete the chain.

We believe that this is the ideal processing for classical music because it does not dynamically re-equalize high frequencies; the subtle HF limiter only acts to reduce high frequency energy when it would otherwise cause overload because of the FM preemphasis curve. We have heard four-band, allegedly “purist” processing that caused dynamic HF lift. This created a strident, unnatural sound in strings and brass. In contrast, the 5750’s two-band phase-linear structure keeps the musical spectrum coherent and natural.

The look-ahead limiter prevents speech from being audibly clipped and prevents similar audible problems on instruments with rapidly declining overtone structures like grand piano, classical guitar, and harp.



- **2B (Two Band) DRIVE –** Sets the level applied to the 2 Band compressor and, thus, the average amount of gain reduction.
- **2B (Two Band) MASTER ATTACK –** Sets the attack time in the master band.
- **2B (Two Band) BASS ATTACK –** Sets the attack time in the bass band.
- **2B (Two Band) RELEASE –** Sets the release rate of the 2 band compressor.

- **2B (Two Band) SPEECH DETECT –** Allows the 5750 to be set to music, speech or auto mode for the speech processor. Auto will switch modes based on audio content.

- **2B (Two Band) MASTER COMPRESSOR THRESHOLD –** Sets the level where gain reduction starts to occur in the master band.

- **2B (Two Band) BASS COMPRESSOR THRESHOLD –** Sets the level where gain reduction starts to occur in the bass band.

- **2B (Two Band) CROSSOVER –** Selects the type of crossover filter between the bass and master band.

- **2B (Two Band) BASS COUPLE –** Determines how closely the bass band will follow the master band. Less coupling leads to more independent bass compression.

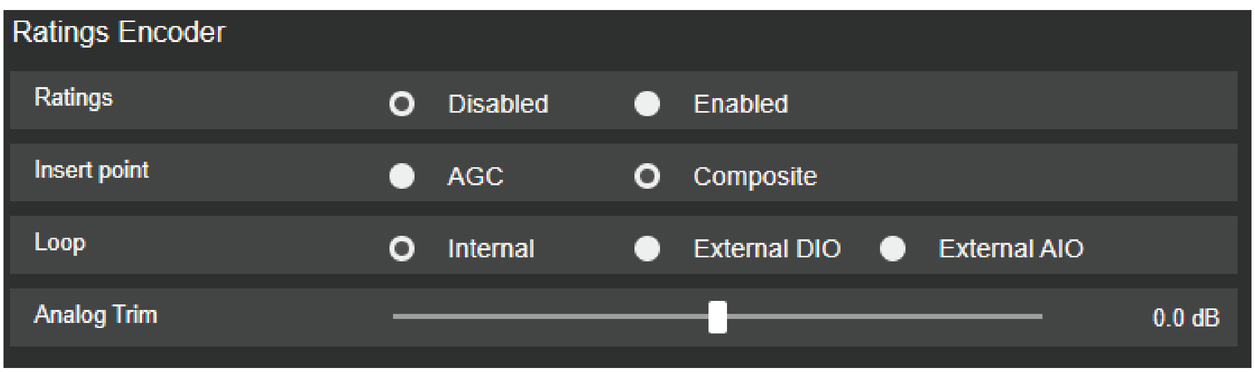
- **2B (Two Band) GATE THRESHOLD –** Determines the lowest audio input signal that will be recognized as program audio.

- **BASS CLIP THRESHOLD** – Sets the embedded bass clipper threshold in dB below the final clipper.
- **BASS CLIP MODE –** Sets the hardness of the embedded bass clipper. There are four modes: SOFT, MEDIUM, HARD and LL HARD.
- **SPEECH BASS CLIP THRESHOLD –** Sets the hard bass clipper threshold when speech is detected.
- **2B (Two Band) HIGH FREQUENCY LIMITING –** Sets the threshold of the high frequency limiter with reference to the final clipper.
- **HF (High Frequency) CLIPPING –** Normally set to 0, this control increases brightness at the expense of intelligent distortion management.
- **HF (High Frequency) CLIP THRESHOLD –** Sets the threshold of the multiband, distortion cancelled clipper in the HF limiter.
- **2B (Two Band) CLIPPING –** Sets the amount of peak limiting in the clipping system.
- **MB (Multiband) LIMIT THRESHOLD –** Sets the threshold of the clipping distortion controller with reference to the final clipper.

- **MB (Multiband) SPEECH THRESHOLD –** Sets the threshold of the clipping distortion controller for speech material.
- **MAXIMUM DISTORTION CONTROL –** Limits the maximum amount of final clipper drive reduction (in dB) that the clipping distortion controller can apply.
- **2B (Two Band) 6-15 kHz HF LIMITER –** Sets the amount of extra gain reduction in the top band of the multiband high frequency limiter.
- **2B (Two Band) LOOKAHEAD –** Sets the delay time of the lookahead 2 band compressor. Higher numbers will yield more accurate processing.
- **MONITOR DRIVE –** Sets the drive into the FM clipper in the monitor processor. Lower is cleaner. Allows users to simulate an FM processing chain for low latency monitoring by talent.

EXTERNAL ENCODER LOOP

The 5750 has a sidechain loop option to place an external ratings encoder INTO THE processing path using either an AES encoder or an analog encoder.



In the HTML5 interface, navigate to I/O Settings. Under the input tab, at the bottom you will see RATINGS ENCODER.

To insert a digital ratings encoder, you will need to connect the AES 2 output to your encoder. Next, take the encoder's output and insert it into SYNC IN on the rear of the 5950. Once complete, you will set ratings to ENABLE, choose where you want the encoding inserted (either after the AGC for stations using one encoder for FM and HD, or before the composite signals for stations that are not transmitting HD). Under LOOP you would choose EXTERNAL DIO.

To insert an analog ratings encoder, you will need to connect the analog L/R output to your encoder. Next, take the encoder's output and insert it into the ANALOG IN on the rear of the 5950. Once complete, you will set ratings to ENABLE, choose where you want the encoding inserted (either after the AGC for stations using one encoder for FM and HD, or before the composite signals for stations that are not transmitting HD). Under LOOP you would choose EXTERNAL AIO.

Section 4: Maintenance

Routine Maintenance

The 5750 Audio Processor uses highly stable analog and digital circuitry throughout. Recommended routine maintenance is minimal.

Periodically check audio level and gain reduction meter readings.

Become familiar with normal audio level meter readings, and with the normal performance of the G/R metering. If any meter reading is abnormal, see Section 5 for troubleshooting information.

Listen to the 5750's output.

A good ear will pick up many faults. Familiarize yourself with the “sound” of the 5750 as you have set it up, and be sensitive to changes or deterioration. However, if problems arise, please do not jump to the conclusion that the 5750 is at fault. The troubleshooting information in Section 5 will help you determine if the problem is with THE OPTIMOD-5750 or is somewhere else in the station's equipment.

Periodically check for corrosion.

Particularly in humid or salt-spray environments, check for corrosion at the input and output connectors and at those places where the 5750 chassis contacts the rack.

Periodically check for loss of grounding.

Check for loss of grounding due to corrosion or loosening of rack mounting screws.

Clean the front panel when it is soiled.

Wash the front panel with a mild household detergent and a damp cloth. Do not use stronger solvents; they may damage plastic parts, paint, or the silk-screened lettering. Do not use paper-based cleaning towels, or use cleaning agents containing ammonia, or alcohol. An acceptable cleaning product is “Glass Plus.” For best results when cleaning the lens, use a clean, lint-free cloth.

Section 5: Troubleshooting

Problems and Potential Solutions

Always verify that the problem is not the source material being fed to the 5750, or in other parts of the system.

RFI, Hum, Clicks, or Buzzes

A grounding problem is likely. Review the information on grounding on page 2-1. The 5750 has been designed with very substantial RFI suppression on its analog and digital input and output ports, and on the AC line input. It will usually operate adjacent to high-powered transmitters without difficulty. In the most unusual circumstances, it may be necessary to reposition the unit to reduce RF interference, and/or to reposition its input and output cables to reduce RF pickup on their shields.

It is not recommended to use a long run of coaxial cable between the 5750 and the exciter as a ground loop may inject noise into the exciter's composite input—especially if the exciter's input is unbalanced.

The AES3 inputs and output are transformer-coupled and have very good resistance to RFI. If you have RFI problems and are using analog connections on either the input or output, using digital connections will almost certainly eliminate the RFI.

Unexpectedly Quiet On-Air Levels

The ITU412 multiplex power controller may have been turned on accidentally.

The 5750 may be in stand-alone stereo encoder mode. The active on-air preset determines this.

The 5750 may not be controlling peak modulation as desired. See the next topic below.

Poor Peak Modulation Control

First, if you are using the analog or digital output to drive the transmitter, make sure that this output is not receiving the MONITOR.

The 5750 normally controls peak modulation to an accuracy of ±2%. This accuracy will be destroyed if the signal path following the 5750 has poor transient response. Almost any link can cause problems. Even the FM exciter can have insufficient flatness of response and phase-linearity (particularly at low frequencies) to disturb peak levels.

Digital STLs using lossy compression algorithms (including MPEG1 Layer 2, MPEG1 Layer 3, Dolby AC2, and APT-X) will overshoot severely (up to 3 dB) on some program material. The amount of overshoot will depend on data rate — the higher the rate, the lower the overshoot.

Even if the transmission system is operating properly, the FM modulation monitor or reference receiver can falsely indicate peak program modulation higher than that actually being transmitted if the monitor overshoots at high and low frequencies. Many commercial monitors have this problem, but most of these problem units can be modified to indicate peak levels accurately.

Orban uses the Belar “Wizard” series of DSP-based monitors internally for testing, because these units do not have this difficulty.

Unexpected Delay Between the Program Feed and the On-Air Signal

The diversity delay may have been accidentally applied to the output you are using to drive your transmitter.

Audible Distortion On-Air

Make sure that the problem can be observed on more than one receiver and at several locations. Multipath distortion at the monitoring site can be mistaken for real distortion (and will also cause falsely high modulation readings).

Verify that the source material at the 5750's audio inputs is clean. Heavy processing can exaggerate even slightly distorted material, pushing it over the edge into unacceptability.

The subjective adjustments available to the user have enough range to cause audible distortion at their extreme settings. There are many controls that can cause distortion, including MULTIBAND CLIPPING, FINAL CLIP DRIVE, and COMPOSITE CLIP DRIVE. Setting the LESS-MORE control beyond “9” will cause audible distortion of some program material with all but the Classical and Protect presets. Further, the “Loud” family of presets can sometimes cause audible distortion with certain program material; this is the price to be paid for “competitive” loudness as it is defined in certain markets.

If you are using analog inputs, the peak input level must not exceed +27 dBu or the 5750's A/D converter will clip and distort.

Unlike earlier digital OPTIMODs, there is no input peak level adjustment for the A/D converter. Instead, we have provided adequate headroom for virtually any facility. This is possible because the A/D converter in the 5750 has higher dynamic range than older designs. Therefore, without compromising the 5750’s noise level, we could eliminate a control that was frequently misadjusted.

If you are using the 5750’s stereo enhancer (which most “pop music”-oriented presets do), then this can exaggerate multipath distortion in high multipath environments. You may want to reduce the setting of the stereo enhancer’s RATIO LIMIT control. A similar problem can occur if you are using sum-and-difference processing in the 5750’s AGC. In this case, reduce the setting of the AGC’s MAXDELTAGR controls.

If you are using an external processor ahead of the 5750, be sure it is not clipping or otherwise causing problems.

Audible Noise on Air

Excessive compression will always exaggerate noise in the source material.

The 5750 has two systems that fight this problem. The compressor gate freezes the gain of the AGC and compressor systems whenever the input noise drops below a level set by the threshold control for the processing section in question, preventing noise below this level from being further increased.

In the Multiband structure, dynamic single-ended noise reduction can be used to reduce the level of the noise below the level at which it appears at the input.

If you are using the 5750's analog input, the overall noise performance of the system is usually limited by the overload-to-noise ratio of the analog-to-digital converter used by the 5750 to digitize the input. (This ratio is better than 108 dB.) It is important to drive the 5750 with professional levels (more than 0 dBu reference level) to achieve adequately low noise. (Clipping occurs at +27 dBu.)

The 5750's AES3 input is capable of receiving words of up to 24 bits. A 24-bit word has a dynamic range of approximately 144 dB. The 5750's digital input will thus never limit the unit's noise performance even with very high amounts of compression.

If an analog studio-to-transmitter link (STL) is used to pass unprocessed audio to the 5750, the STL's noise level can severely limit the overall noise performance of the system because compression in the 5750 can exaggerate the STL noise. For example, the overload-to-noise ratio of a typical analog microwave STL may only be 70-75 dB. In this case, it is wise to use the Orban 8200ST Studio AGC to perform the AGC function prior to the STL transmitter and to control the STL's peak modulation. This will optimize the signal-to-noise ratio of the entire transmission system. An uncompressed digital STL will perform much better than any analog STL.

Whistle on Air, Perhaps Only in Stereo Reception

The most likely cause is oscillation in the analog input or output circuitry. If the oscillation is in the output circuitry and is between 23 and 53 kHz, it will be detected in a receiver’s stereo decoder and translated down into the audible range.

If you encounter this problem, check the analog or digital outputs with a spectrum analyzer to see if the spurious tone can be detected here. If it appears at both outputs, it is probably an input problem. If it only appears at the analog output, then it is likely a problem with the left/right D/A converter or other analog circuitry. If it appears only when you use the composite output, then it is likely a problem in the composite D/A converter or output amplifiers.

A whistle could also be caused by power supply oscillation, STL problems, or exciter problems.

Interference from stereo into SCA

A properly operating 5750 generates an immaculately clean baseband, with program-correlated noise below –80 dB above 57 kHz even when the composite limiter is used aggressively. If the 5750 and the rest of the transmission system are operating correctly, subcarriers should experience no interference.

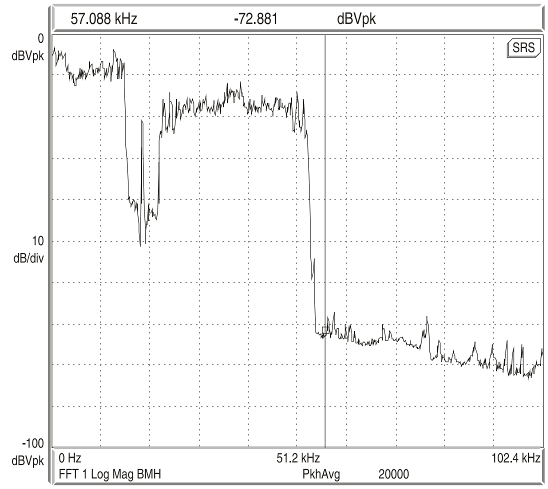


Figure 5-1: Typical 5750 baseband spectrum with heavy processing, 0-100 kHz.

Interference from the stereo into a subcarrier is best diagnosed with a spectrum analyzer. First examine the spectrum of the 5750's composite output to verify that program correlated noise is less than –80 dB below 100% modulation from 57 to 100 kHz. Any inadvertent composite clipping will dramatically degrade this protection. Make sure that the link between the 5750's composite output and the transmitter has sufficient headroom.

If the exciter is nonlinear, this can cause crosstalk. In general, a properly operating exciter should have less than 0.1% THD at high frequencies to achieve correct operation with subcarriers.

To prevent truncation of the higher-order Bessel sidebands of the FM modulation, the RF system following the exciter must be wideband (better than ±500 kHz) and must have symmetrical group delay around the carrier frequency. An incorrectly tuned transmitter can exhibit an asymmetrical passband that will greatly increase crosstalk into subcarriers.

Amplitude modulation of the carrier that is synchronous with the program ("synchronous AM") can cause program-related crosstalk into subcarriers. Synchronous AM should be better than 35 dB below 100% modulation as measured on a synchronous AM detector with standard FM de-emphasis (50µs or 75µs).

The subcarrier receiver itself must receive a multipath-free signal, and must have a wide and symmetrical IF passband and a linear, low-distortion FM demodulator to prevent program-related crosstalk into subcarriers.

Shrill, Harsh Sound

If you are using the Five-Band structure, this problem can be caused by excessive HF boost in the HF Equalizer and HF Enhancer.

It could also be caused by an excessively high setting of the BAND 4 THRESH control, or by excessively high settings of the BAND 4 MIX and BAND 5 MIX controls (located in Intermediate and Advanced Modify).

If you are driving an external stereo encoder with built-in pre-emphasis, you must set the 5750's output to Flat in the System Setup > Output screen to prevent double pre-emphasis, which will cause very shrill sound (and very poor peak modulation control).

You will always achieve better peak control by defeating the pre-emphasis and input filters of an external stereo encoder, permitting the 5750 to perform these functions without overshoot.

Dull Sound

If you are using the Two-Band structure, dull-sounding source material will sound dull on the air. The Five-Band structure will automatically re-equalize such dull-sounding program material to make its spectral balance more consistent with other program material.

If the 5750's output is set to Flat in System Setup > Output, there will be no pre-emphasis unless it is supplied somewhere else in the system. This will cause very dull sound.

System Will Not Pass Line-Up Tones at 100% Modulation

This is normal. Sine waves have a very low peak-to-average ratio by comparison to program material. The processing thus automatically reduces their peak level to bring their average level closer to program material, promoting a more consistent and well-balanced sound quality.

The 5750 can generate test tones itself. The 5750 can also be put into Bypass mode (locally or by remote control) to enable it to pass externally generated tones at any desired level.

System Will Not Pass Emergency Alert System ("EAS" USA Standard) Tones at the Legally Required Modulation Level

See System Will Not Pass Line-Up Tones at 100% Modulation (directly above) for an explanation. These tones should be injected into the transmitter after the 5750, or the 5750 should be temporarily switched to BYPASS to pass the tones.

System Receiving 5750's Digital Output Will Not Lock

Be sure that the 5750's output sample rate is set match the sample rate that the driven system expects. Be sure that the 5750's output FORMAT (AES3 or SPDIF) is set to match the standard expected by the driven system.

You See A "Communications Board Error" Message

This can be caused by applying an AES3, AES3id, AES11, or AES11id to the 10 MHz/wordclock input. Use the XLR AES3 input as an AES3 sync source.

19 kHz Frequency Out-of-Tolerance

First, verify that a problem really exists by using a second frequency-measuring device and/or verifying the problem with a monitoring service. If the problem is real, contact Orban Customer Service for a crystal replacement; there is no frequency trim available.

L–R (Stereo Difference Channel) Will Not Null With Monophonic Input

This problem is often caused by relative phase shifts between the left and right channels prior to the 5750's input. This will cause innocuous linear crosstalk between the stereo main and subchannels. Such crosstalk does not cause subjective quality problems unless it is very severe.

Audio Mute Occurs When Switching Between UL and Non-UL Presets

This is normal. It occurs because the DSP code must be reloaded.

HD Output Sounds Too Bright

Enable adjustment of the HD output's spectral balance by putting the on-air preset into INDEPENDENT mode. Do this by setting the FM→HD CONTROL COUPLING control in the HD LIMITING page of ADVANCED CONTROL to INDEPENDENT. You can also toggle this control via two buttons on the button bar in HTML application.

In this mode, the HD processing channel's equalizer, five-band compressor, and band-mix controls are independent of the corresponding controls in the FM channel and can be adjusted separately. You can therefore fine-tune high frequencies by adjusting the equalizer, parameters in the band 4 and band 5 compressors, the band 4 and band 5 BAND MIX controls, and the HD DE-ESS control. One of the most effective ways to tame harsh high frequencies dynamically is to activate the HD channel's band 5 compressor.

Harsh Sibilance ("Ess" Sounds) in the HD Channel

Adjust the HD DE-ESS control and/or activate the HD channel's band 5 compressor.

HD and FM Levels Do Not Match When the Receiver Crossfades

Adjust the HD LIMIT DR control in the on-air preset to match levels.

Do not match levels by adjusting the output level of the output driving the HD exciter. Only use this control to match the peak levels of the OPTIMOD output to the exciter.

In other words, if the exciter is set to clip when it receives at –3 dBFS, adjust the OPTIMOD output level to –3 dBFS. This uses all of the headroom available in the transmission channel, minimizing the amount of look-ahead limiting that the OPTIMOD needs to do.

If you have accidentally turned on the BS.1770 Safety Limiter, this will cause the loudness to be lower than expected in HD Radio applications. Turn it off.

Loudness is unexpectedly low from the analog FM processing chain

The MPX Power Controller and/or analog FM processing chain BS.1770 Safety Limiter may have been turned on accidentally.

Digital Radio Loudness Cannot be Set Using the Digital Output 100% Peak Level Control

When the BS.1770 Safety Limiter is ON, adjusting the HD-assigned 100% PEAK LEVEL CONTROL sets only the output headroom, not loudness.

Loudness Drops Momentarily During HD Radio Analog/Digital Crossfades

The analog and digital channels in your transmission path have reversed polarity with respect to each other and a phase cancellation is occurring in the radio during crossfades. You can correct this with the HD POLARITY or FM POLARITY controls.

BS.1770 Safety Limiter produces too much gain reduction

Turn down the HD Final Limiter Drive control in the processing preset and save the result as a User Preset.

HD Frequency Response is Limited to 15 kHz

The HD BANDWIDTH control might be set to 15 kHz.

Even if the HD BANDWIDTH control is set to 20 kHz, bandwidth will be limited to 15 kHz if the HD output's sample rate is set to 32 kHz and/or if the sample rate of the audio applied to the 5750's digital input is 32 kHz. 20 kHz response requires sample rates to be set to 44.1 kHz or higher.

General Dissatisfaction with Subjective Sound Quality

The 5750 is a complex processor that can be adjusted for many different tastes. For most users, the factory presets, as augmented by the gamut offered by the LESS-MORE control for each preset, are sufficient to find a satisfactory "sound." However, some users will not be satisfied until they have accessed other Modify Processing controls and have adjusted the subjective setup controls in detail to their satisfaction. Such users must fully understand the material in Section 3 of this manual to achieve the best results from this exercise.

By comparison to competitive processors, the 5750 offers a uniquely favorable set of trade-offs between loudness, brightness, distortion, and build-up of program density.

If your radio station does not seem to be competitive with others in your market, the cause is usually source material (including excess use of lossy digital compression), overshoot in the transmission link (including the FM exciter) following the 5750, or an inaccurate modulation monitor that is causing you to under-modulate the carrier. A station may suffer from any combination of these problems, and they can have a remarkable effect on the overall competitiveness of a station's sound..

Technical Support

If you require technical support, contact Orban customer service.
See http://www.orban.com/contact/ for contact information.

Be prepared to describe the problem accurately. Know the serial number of your 5750. This is printed on the rear panel of the unit.

Please check Orban's website, www.orban.com, for Frequently Asked Questions and other technical tips about 5750 that we may post from time to time. Manuals (in .pdf form) and 5750 software upgrades will be posted there too — click "Downloads" from the home page.

Factory Service

Before you return a product to the factory for service, we recommend that you refer to this manual. Make sure you have correctly followed installation steps and operation procedures. If you are still unable to solve a problem, contact our Customer Service for consultation. Often, a problem is relatively simple and can be quickly fixed after telephone consultation.

If you must return a product for factory service, please notify Customer Service by telephone, before you ship the product; this helps us to be prepared to service your unit upon arrival. Also, when you return a product to the factory for service, we recommend you include a letter describing the problem.

Please refer to the terms of your Limited 5 year Standard Warranty, which extends to the first end user. After expiration of the warranty, a reasonable charge will be made for parts, labor, and packing if you choose to use the factory service facility. Returned units will be returned C.O.D. if the unit is not under warranty. Orban will pay return shipping if the unit is still under warranty. In all cases, the customer pays transportation charges to the factory (which are usually quite nominal).

Shipping Instructions

Use the original packing material if it is available. If it is not, use a sturdy, double-walled carton no smaller than 7″ (H) x 15.5″ (D) x 22″ (W) □ 18 cm (H) x 40 cm (D) x 56 cm (W), with a minimum bursting test rating of 200 pounds (91 kg). Place the chassis in a plastic bag (or wrap it in plastic) to protect the finish, then pack it in the carton with at least 1.5 inches (4 cm) of cushioning on all sides of the unit. "Bubble" packing sheets, thick fiber blankets, and the like are acceptable cushioning materials; foam "popcorn" and crumpled newspaper are not. Wrap cushioning materials tightly around the unit and tape them in place to prevent the unit from shifting out of its packing.

Close the carton without sealing it and shake it vigorously. If you can hear or feel the unit move, use more packing. Seal the carton with 3-inch (8 cm) reinforced fiberglass or polyester sealing tape, top and bottom in an "H" pattern. Narrower or parcel-post type tapes will not withstand the stresses applied to commercial shipments.

Mark the package with the name of the shipper, and with these words in red:

DELICATE INSTRUMENT, FRAGILE!

Insure the package properly. Ship prepaid, not collect. Do not ship parcel post. Your Return Authorization Number must be shown on the label, or the package will not be accepted.

Section 6: Technical Data

J613:

As shipped from the factory, a jumper is kept across pins 1 and 2 of J613, and a second jumper is kept across its pins 3 and 4. The jumpers engage the direct connection from AES/EBU DIGITAL INPUT 1 to AES/EBU DIGITAL OUTPUT 1 made by the relay bypass function when OPTIMOD processing is unavailable.

To disengage the direct connection just described, the user should remove both of those jumpers. Removal of the jumpers will be necessary in most configurations involving use of AES/EBU DIGITAL OUTPUT 1 as a DMPX source.

J614:

As shipped from the factory, a jumper is kept across pins 1 and 2 of J614, and a second jumper is kept across its pins 3 and 4. The jumpers engage the direct connection from AES/EBU DIGITAL INPUT 2 to AES/EBU DIGITAL OUTPUT 2 made by the relay bypass function when OPTIMOD processing is unavailable.

To disengage the direct connection just described, the user should remove both of those jumpers. Removal of the jumpers will be necessary in most configurations involving use of AES/EBU DIGITAL OUTPUT 2 as a DMPX source.

J501/J502:

As shipped from the factory, J501 (composite 1) and J502 (composite 2) are across pins 3-4 for 0Ω impedance on their respective composite output. Switching to pins 1-2 sets the impedance to 75Ω.

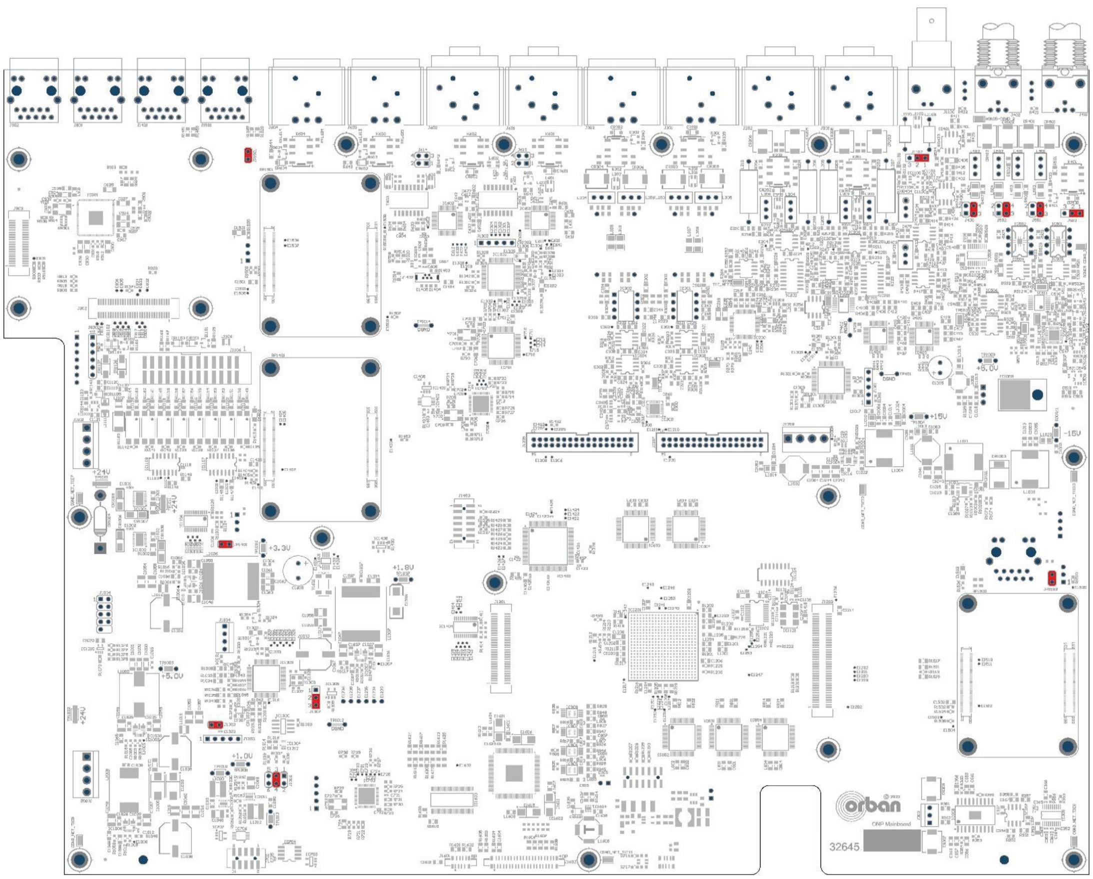
CAUTION

TO PREVENT ELECTRICAL SHOCK, DO NOT REMOVE COVER

NO USER SERVICABLE PARTS INSIDE.

REFER SERVICING TO QUALIFIED SERVICE PRESEONNEL.

PROTECT AGHAINST HUMIDITY



Default Jumper Settings

J400 3-4
J500 1-2
J501 3-4
J502 3-4
J1102 1-2
J1300
J1306 3-4
J1307 2-3
J1401
J1501
J1502

orban ®

7209 Browning Road
Pennsauken, NJ 08110
Phone 1-856-719-9900

MODEL NO.	5950	TITLE JUMPER SETTINGS Mainboard 5950				
APPROVALS	DATE					
DWN D. Franz	05/04/23					
CKD *	*	SIZE	DWG NO.	VER	REV	SHEET
APVR		*	32645	595	03.1	1 OF 1

Using Lossy Data Reduction in the Studio

Many stations are now using lossy data reduction algorithms like MPEG-1 Layer 2 or Dolby AC2 to increase the storage time of digital playback media. In addition, source material is often supplied through a lossy data reduction algorithm, whether from satellite or over landlines. Sometimes, several encode / decode cycles will be cascaded before the material is finally presented to THE OPTIMOD-5750's input.

All such algorithms operate by increasing the quantization noise in discrete frequency bands. If not psychoacoustically mask`ed by the program material, this noise may be perceived as distortion, an "underwater sound," or other perceptual degradation. Psychoacoustic calculations are used to ensure that the added noise is masked by the desired program material and not heard. Cascading several stages of such processing can raise the added quantization noise above the threshold of masking, such that it is heard.

At least one other mechanism can cause the noise to become audible at the radio. THE OPTIMOD-5750's multiband limiter performs an "automatic equalization" function that can radically change the frequency balance of the program. This can cause noise that would otherwise have been masked to become unmasked because the psychoacoustic masking conditions under which the masking thresholds were originally computed have changed.

Accordingly, if you use lossy data reduction in the studio, you should use the highest data rate possible. This maximizes the headroom between the added noise and the threshold where it will be heard. Also, you should minimize the number of encode and decode cycles, because each cycle moves the added noise closer to the threshold where the added noise is heard.

About Transmission Levels and Metering

Metering

Studio engineers and transmission engineers consider audio levels and their measurements differently, so they typically use different methods of metering to monitor these levels. The VU meter is an average-responding meter (measuring the approximate RMS level) with a 300ms rise time and decay time; the VU indication usually under-indicates the true peak level by 8 to 14dB. The Peak Program Meter (PPM) indicates a level between RMS and the actual peak. The PPM has an attack time of 10ms, slow enough to cause the meter to ignore narrow peaks and under-indicate the true peak level by 5 dB or more. The absolute peak-sensing meter or LED indicator shows the true peak level. It has an instantaneous attack time and a release time slow enough to allow the engineer to read the peak level easily. Figure 1-2: Absolute Peak Level, VU and PPM Indications shows the relative difference between the absolute peak level and the indications of a VU meter and a PPM for a few seconds of music program.

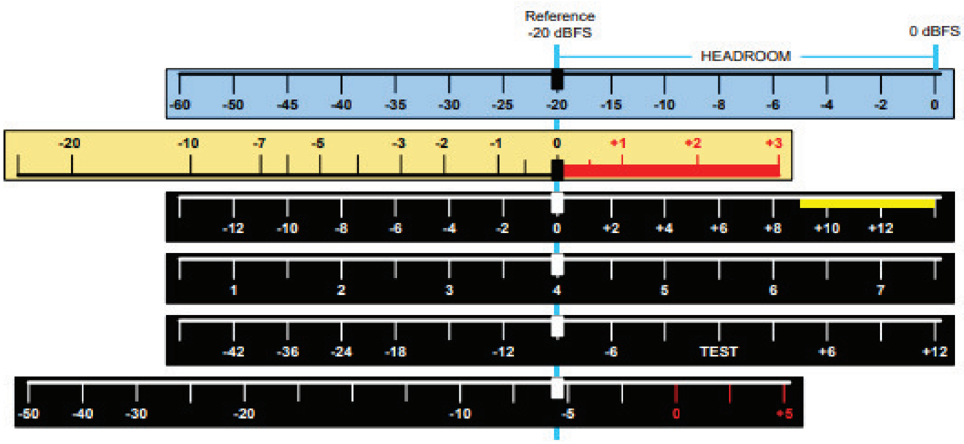
Studio Line-up Levels and Headroom

The studio engineer is primarily concerned with calibrating the equipment to provide the required input level for proper operation of each device so that all devices operate with the same input and output levels. This facilitates patching devices in and out without recalibration.

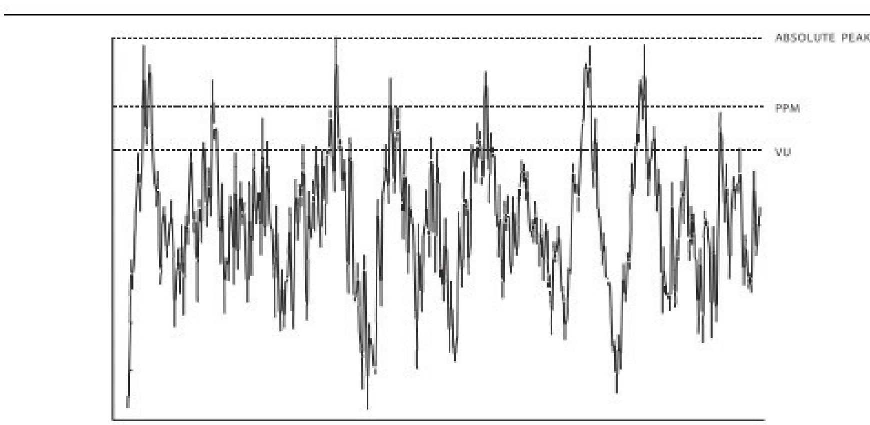
For line-up, the studio engineer uses a calibration tone at a studio standard level, commonly called line-up level, reference level, or operating level. Metering at the studio is by a VU meter or PPM (Peak Program Meter). As discussed above, the VU or PPM indication under-indicates the true peak level.

Most modern studio audio devices have a clipping level of no less than +21dBu and often +24dBu or more. The studio standardizes on a maximum program indication on the meter that is lower than the clipping level, so peaks that the meter does not indicate will not be clipped. Line-up level is usually at this same maximum meter indication. In facilities that use VU meters, this level is usually at 0VU, which corresponds to the studio standard level, typically +4 or +8dBu.

For facilities using +4dBu standard level, instantaneous peaks can reach +18dBu or higher (particularly if the operator overdrives the console or desk). Older facilities with +8dBu standard level and equipment that clips at +18 or +21dBu will experience noticeable clipping on some program material. In facilities that use the BBC-standard PPM, maximum program level is usually PPM4 for music, PPM6 for speech. Line-up level is usually PPM4, which corresponds to +4dBu. Instantaneous peaks will reach +17dBu or more on voice. In facilities that use PPMs that indicate level directly in dBu, maximum program and line-up level is often +6dBu. Instantaneous peaks will reach +11dBu or more.



Common Audio Meter Scales, Aligned to the Same Reference Level



Absolute Peak Level, VU and PPM Indications

Figure 6.2: Studio Line-up Levels and Headroom

Transmission Levels

The transmission engineer is primarily concerned with the peak level of a program to prevent overloading or over-modulation of the transmission system. This peak overload level is defined differently, system to system. In FM modulation (FM / VHF radio and television broadcast, microwave or analog satellite links), it is the maximum-permitted RF carrier frequency deviation.

In AM modulation, it is negative carrier pinch-off. In analog telephone / post / PTT transmission, it is the level above which serious crosstalk into other channels occurs, or the level at which the amplifiers in the channel overload. In digital, it is the largest possible digital word. For metering, the transmission engineer uses an oscilloscope, absolute peak-sensing meter, calibrated peak-sensing LED indicator, or a modulation meter. A modulation meter usually has two components—a semi-peak reading meter (like a PPM) and a peak-indicating light, which is calibrated to turn on whenever the instantaneous peak modulation exceeds the overmodulation threshold.

Line Up Facilities

Metering of Levels and Subjective Loudness

The meters on the 5750 show left/right input and output levels and composite modulation. Left and right input level is shown on a VU-type scale (0 to –40 dB), while the metering indicates absolute instantaneous peak (much faster than a standard PPM or VU meter). The input meter is scaled so that 0 dB on the scale corresponds +27 dBu, which is the absolute maximum peak level that the 5750 can accept. If you are using the AES3 digital input, a full-scale digital word corresponds to the 0 dB point on the 5750's input meter.

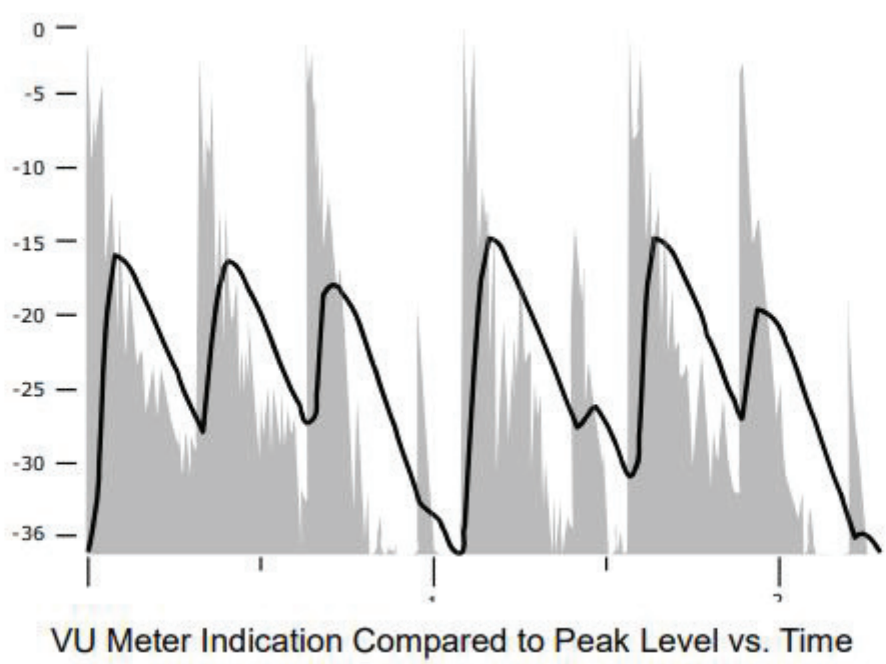


Figure 6-3: Metering of Levels and Subjective Loudness

Left/right Output Level

Left and right output level is shown on a VU-type scale. The metering indicates absolute instantaneous peak (much faster than a standard PPM or VU meter). The meter is scaled so that 0 dB is calibrated to the highest left and right peak modulation level, before de-emphasis, that the processing will produce, under any program, processing, or setup condition (except when the processing is switched to BYPASS). The meter indication is not affected by the setting of the output level control.

Composite Output Level

The Orban 5750 Audio Processor controls instantaneous, absolute peak levels to a tolerance of approximately 0.1 dB. Composite modulation is indicated in percentage modulation, absolute instantaneous peak indicating. 100% is calibrated to the highest composite peak modulation level that the processing will produce, including the pilot tone, under any program, processing, or setup condition (except when the processing is switched to BYPASS). 100% ordinarily corresponds to ±75 kHz-carrier deviation.

Note that if the 5750's subcarrier inputs are used, the meter will not indicate the subcarriers' effect on composite modulation because the subcarriers are mixed into the composite signal in the analog domain, after the composite signal is metered. Therefore, you must mentally add the subcarriers to the meter indication or refer to an external, calibrated modulation monitor.

BS.1770 Loudness Level

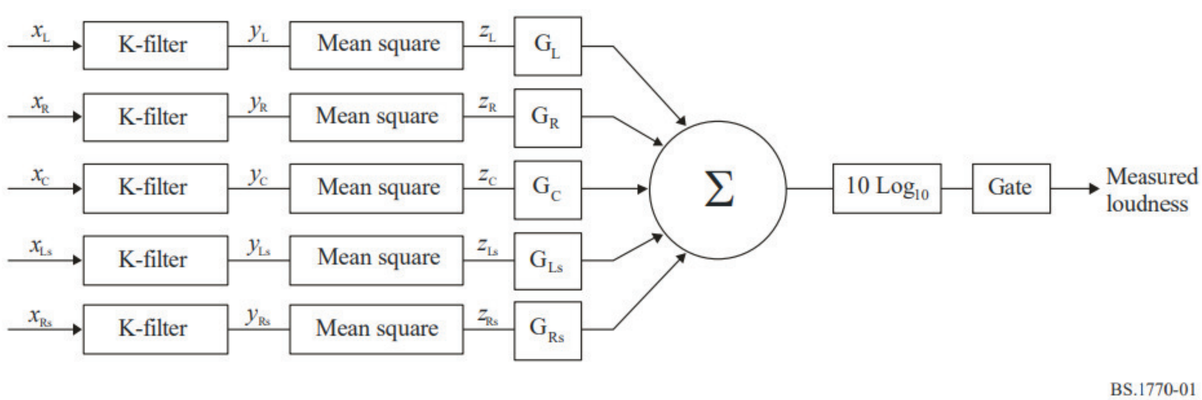


Figure 6-4: BS.1770 Loudness Meter Block Diagram (from ITU-R document)

The subjective loudness meters, labeled LOUDNESS in the 5750's GUI, display the loudness at the output of the digital radio processing chain and analog FM processing chains, measured by the ITU-R BS.1770-3 algorithm.

Recommendation ITU-R BS.1770-3 (08/2012): "Algorithms to measure audio programme loudness and true-peak audio level." BS.1770-3 first introduced gating to the loudness meter. For further information about the BS.1770-3 meter, please refer to the following standards: ITU-R BS.1770, ATSC A/85 EBU R 128, EBU Tech 3341, EBU.

The loudness meter indicates both BS.1770 Short-Term and Integrated Loudness on the same scale. The solid bar indicates the Integrated Loudness, while the floating bar segment indicates the Short-Term Loudness. Per the BS.1770-3 specification, the integration time of the Short-Term Meter is always 3 seconds, while the Integrated

Meter uses silence gating and its integration time is 10 seconds. Because the meter is always monitoring program material, it integrates the previous 10 seconds of program material and weights all program material equally within the specified time window.

For example, material occurring 3 seconds in the past and 8 seconds in the past both contribute equally to the meter's current indication; newer program material in the specified time window is not favored over older program material.

Because loudness perception combines the contributions of all acoustic sources, there is only one Loudness Level meter indication for both stereo channels.

The unit of measure in the BS.1770 meter is LKFS or LUFS, which are the same, differing only in nomenclature. A change of 1 LUFS is the same as a change of 1 dB.

In the digital radio chain, "LKFS" and "LUFS" are absolute loudness measurements with respect to digital full scale. "LK" and "LU" (without the "FS") are relative loudness measurements, where "0" on the meter corresponds to a user-preset "BS.1770 Reference Level," which you set via your OPTIMOD's BS.1770 LOUDNESS CONTROL THRESHOLD control. The BS.1770 meter on your OPTIMOD indicates "LK" or "LU"; you can choose which label to use via a control available on the HD DIGITAL RADIO tab in I/O SETUP. The other BS.1770-associated controls for digital radio are also there.

The meter is scaled so that the loudness level at the consumer's receiver is correct when the 5750's digital radio processing chain is adjusted to make the dominant program material indicate "0 dB" on the 5750's Loudness Level meter and the BS.1770 REFERENCE LEVEL (which you must enter manually) in I/O SETUP > HD RADIO is equal to that specified by the regulatory authority in your country.

In the analog radio processing chain, the BS.1770 meter and Safety Limiter are calibrated per EBU Tech 33444, Section 5.9 titled "Practical guidelines for distribution systems in accordance with EBU R 128; Supplementary information for EBU R 128," which is available for free download. Use a search engine to find the latest version. This calls for a 1 kHz sinewave at –23 LUFS to produce an FM carrier deviation of ±14 kHz without pilot tone. This corresponds to 18.67% modulation without pilot tone. This calibration includes 50µs transmission pre-emphasis. The loudness meter is calibrated so that "0" corresponds to the setting of the BS.1770 LOUDNESS CONTROL THRESHOLD control in INPUT/OUTPUT > UTILITY.

Built-in Calibrated Line-up Tones

To facilitate matching the output level of the 5750 to the transmission system that it is driving, the 5750 contains an adjustable test tone oscillator that produces sine waves at 5750's (analog or digital) left, right, and composite outputs. The frequency and modulation level of the line-up tones can be adjusted from the front panel.

The stereo encoder is calibrated so that 100% left or right modulation will provide 100% modulation of the stereo composite signal, including pilot tone, but excluding any SCA subcarriers. The pilot tone stereo system has an interleaving property, which means that the stereo composite modulation is approximately equal to the higher of the left or right channels.

Because the pilot tone is phase-synchronous with the stereo subcarrier, the composite modulation will actually increase about 2.7% when the modulation is changed from pure single-channel to L+R modulation while the peak audio level is held constant.

When the 5750's left/right analog output is switched to FLAT, a de-emphasis filter is inserted between output of the 5750's audio processing and its line output. Thus, as the frequency of the Test Tone is changed, the level at the 5750's line output will follow the selected de-emphasis curve. In most cases, the pre-emphasis filter in the driven equipment will undo the effect of the 5750's internal de-emphasis, so the 5750's output level should be adjusted such that the tone produces 100% modulation of the transmission link as measured after the link's pre-emphasis filter. At 100Hz, switching the de-emphasis out or in will have negligible effect on the level appearing at the 5750's left and right audio outputs. You can adjust the frequency and modulation level of the built-in line-up tone. You can use the HTML5 PC connection, or the opto-isolated remote control interface ports to activate the Test Tone.

Built-in Calibrated Bypass Test Mode

A BYPASS Test Mode is available to transparently pass line-up tones generated earlier in the system. It will also pass program material, applying no gain reduction or protection against overmodulation. It can transparently pass any line-up tone applied to its input up to about 130% output modulation, at which point clipping may occur.

ITU-R MULTIPLEX POWER CONTROLLER

The ITU-R recommends that the power in the composite baseband signal (including the pilot tone), integrated over any 60-second interval, not exceed the power in a sinewave that modulates the FM carrier to ±19 kHz (25.3% modulation). Many European countries are now enforcing this recommendation. (See *ITU-R 412 Compliance* on page 3-**Error! Bookmark not defined.** for more information.)

The BS.1770 Safety Limiter for the analog radio processing chain is located immediately before the MPX power controller. Normally, both are used simultaneously, but when the target loudness is –23 LUFS, the BS.1770 Safety Limiter typically produces enough gain reduction to cause the MPX Power Controller to produce no gain reduction. See step **Error! Reference source not found.** on page 2-**Error! Bookmark not defined.** for instructions on setting up the BS.1770 Safety Limiter.

MPX Power Meter

The MPX POWER meter indicates MPX power according to the ITU-R BS.412 standard. All samples are weighted equally in a 60-second sliding window.

BS.412 requires limiting the integrated power of the composite signal so that it does not exceed the power in a sinewave that deviates the FM carrier by ±19 kHz (25.333% modulation with reference to ±75 kHz deviation). The 5750's MPX POWER meter is therefore calibrated so that it indicates 0 dB when the composite output of the 5750 is a sinewave at 25.333% modulation, which is –11.92615 dB with reference to a sinewave at 100% modulation.

The meter is calibrated with reference to the 5750's 100% peak modulation level. This calibration is only correct if the transmitter and/or studio-transmitter link do not add overshoots to program material processed by the 5750. Such overshoots necessitate turning down the 5750's output level control after it has been calibrated with tone using an FM modulation meter and the 5750's built-in line-up tone oscillator. If the output level is turned down after a tone calibration, the MPX POWER LEVEL meter will read high compared to the actual on-air MPX power. The error will be equal to the amount that the 5750's output level control was turned down.

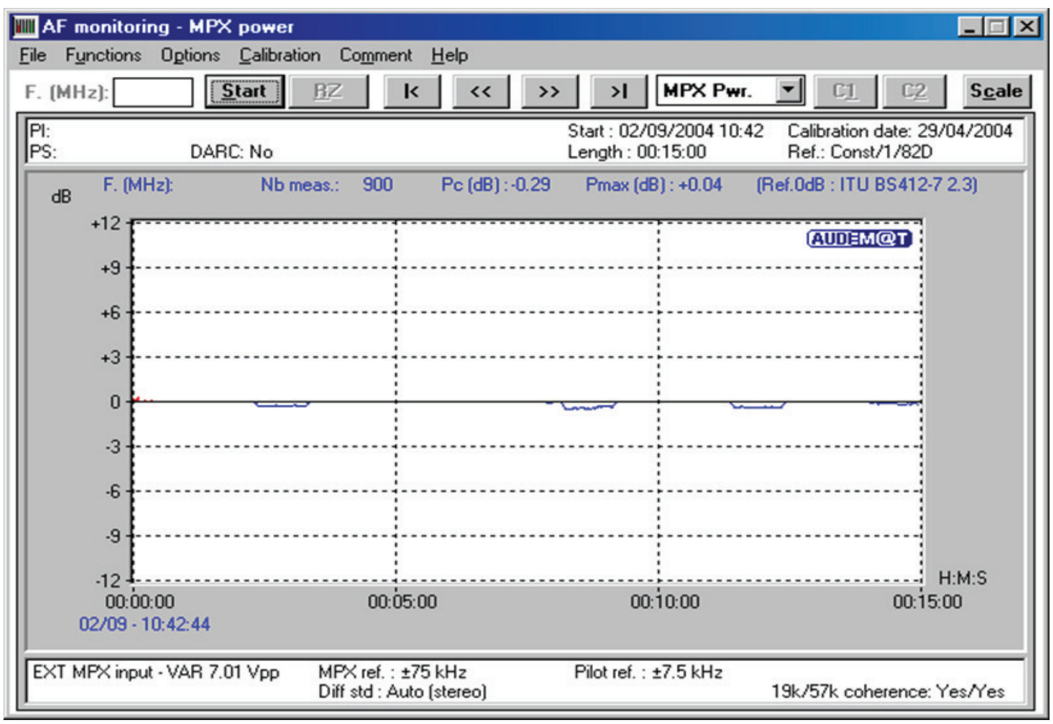
See ***Error! Reference source not found.*** starting on page 1-**Error! Bookmark not defined.** for a discussion of overshoots and how they force the average modulation to be reduced to prevent peak overmodulation of the FM carrier.

Because the 5750 does not digitize subcarriers applied to its subcarrier inputs, the 5750's MPX POWER meter (which operates in the DSP domain) cannot indicate the power added by such subcarriers. These usually have constant power, so it is easy to compensate for them. For example, if an FM subcarrier is injected at 4% modulation, it adds power that can be calculated with an R.M.S. summation of the subcarrier and the rest of the composite signal.

Assuming that the subcarrier and composite signal are uncorrelated and that the composite signal is limited so that its power is equivalent to a sinewave at 25.3% modulation, we calculate their R.M.S. sum as follows:

$$\sqrt{0.25333^2 + 0.04^2} = \sqrt{0.06418 + 0.0016} = 0.25647$$
$$20\log_{10}(0.25647) = -11.81921\text{ dB}$$

Recalling that the MPX POWER LEVEL meter is calibrated so that it indicates 0 dB when the composite output of the 5750 is a sinewave at –11.92615 dB below 100% modulation, we conclude that our subcarrier at 4% injection will add 0.10694 dB to the multiplex power. Another calculation (not shown) indicates that 10% injection will add 0.62889 dB to the MPX power.



Multiplex Power Threshold: The 5750 provides a means to limit the integrated multiplex power to the ITU standard by a technique that allows you to use any preset and to create customized presets freely. The multiplex power controller is adjusted in the INPUT/OUTPUT > UTILITIES screen by the MULTIPLEX POWER THRESHOLD control. Set it OFF if your country does not enforce the standard.

The control is located in the INPUT/OUTPUT > UTILITIES screen because the regulation applies to operation of the processor in a given installation.

Figure 6-11: Multiplex power over 15 minute observation interval
with Multiplex power controller active, measured at the Optimod's composite output

If your country enforces the standard, you should set the control to complement the amount of peak overshoot in the transmission system following the 5750. Setting the control at "0" will correctly control the multiplex power when there is no overshoot after the 5750. This will typically be true when you are using your Optimod's built-in stereo encoder to drive the transmitter directly.

Many paths have overshoot and this forces you to reduce the average modulation to avoid overmodulating the transmitter. This would reduce the multiplex power by the same amount, forcing the multiplex power below the ITU requirement.

To compensate for this, match the MULTIPLEX POWER THRESHOLD control to the peak overshoot of the transmission system following the 5750. For example, if RF peak deviation exceeds the peak deviation produced by the 5750's sinewave oscillator (set for 100% modulation) by 3 dB, set the MULTIPLEX POWER THRESHOLD to "+3."

Audio Processing and the Multiplex Power Threshold Control

The multiplex power controller reduces multiplex power by applying gain reduction after the Optimod's FM peak limiting system, which reduces the tendency of the MPX power controller to produce unnatural-sounding gain reduction because the standard forces MPX power to be measured after preemphasis and without psychoacoustic weighting.

With no power control, some of the louder 5750 presets can exceed the ITU standard by as much as 16 dB. This means that the controller must reduce gain by as much as 16 dB depending on the dynamics and spectral content of the input program material. To prevent unnatural loudness variations, your Optimod applies a static loss (preset-dependent and set by the MULTIPLEX POWER OFFSET control) before the FM peaks limiters when the multiplex power controller is activated. This complements the dynamic gain reduction produced by the multiplex power controller.

The MPX offset is applied before the peak limiters. Turning it up (for example, from –12 to –9 dB) increases both the amount of peak limiting and the amount of wideband gain reduction performed by the MPX Power Controller

The multiplex power controller does not use the output of the 5750's stereo encoder as its reference. Instead, it computes the multiplex power directly from the left and right audio signals, the setting of the PILOT LEVEL control, and the setting of the COMPOSITE LIMIT DRIVE control. Hence, the multiplex power controller does not take into account the effect of any composite limiting on the multiplex power. This is not a problem because a BS412-compliant broadcast does not cause enough composite limiting to affect the multiplex power measurably. The purpose for this change was to allow the multiplex power controller to work even when diversity delay is applied to the stereo encoder.

The multiplex power controller is operational with all of the Two-Band and Five-Band processing structures. *It is not active in Test mode and will not prevent the 5750's test oscillator from producing illegal modulation.* It is the responsibility of the operator to make sure that the test oscillator does not violate the ITU requirements.

(To ensure this, never modulate the carrier with a single L+R tone that produces total carrier modulation, including pilot tone, of more than 24%.)

About the Multiplex Power Controller's Time Constants

Although the BS412 specification calls for a 60-second integration time, the integration time of the Optimod's MPX power controller is about 10 seconds. The problem with making the integration time longer is that the BS412 standard states that the integrated MPX power in any *arbitrary 60-second time period* cannot exceed the average power of the sinewave that produced ±19 kHz carrier deviation. In other words, *whenever you start measuring*, you must not exceed the total integrated power limit over the following 60 seconds.

This makes it impractical to "bank" power over the full 60-second window. For example, at first glance one might think that a classical music station could exploit a period of quiet music to allow a crescendo to get louder than it would using the 5750's relatively fast integration time. However, what happens if someone starts an arbitrary 60-second measurement period not at the beginning of the quiet passage but at the beginning of the crescendo?

Because an automatic MPX power controller does not know what is coming after the crescendo, it must reduce the level of the crescendo so that it complies with the MPX power requirement over an integration time that is shorter than 60 seconds. Otherwise, it might have to dramatically reduce the level of following (as yet unknown) program material in order to ensure that the MPX power limit is not exceeded over the 60-second measurement period in question. This kind of gain pumping would be far worse than the pumping produced by using a relatively short integration time.

MPX Pwr Ctrlr Gate: To minimize audible side effects of the MPX power controller's gain reduction, its release time is dual-speed and changes as a function of the audio level: if the audio level is below a preset threshold, the slower time constant activates. There are five preset values for the gating, which set the level below which gating occurs, as well as the release times above and below the threshold. Higher-numbered presets provide slower release times both above and below the gating threshold.

Unlike the MPX POWER THRESHOLD control (a System control), the MPX PWR CTRLR GATE control is part of the active processing preset.

Preset 0: No gating: Works like Orban's older, non-gated MPX power controller.

Preset 1: Only the quieter passages are gated and the gated release is faster than it is in the other presets. Created to maximize loudness within the BS.412 limit while providing more on-air dynamics and preventing unnecessary gain pump-ups.

Preset 2: A compromise between Preset 2 and 3. Works well with more dynamic, more open-sounding presets.

Preset 3: Recommended for most CHR-style presets. Quieter parts of the music are effectively frozen to achieve less audible BS412 control. It is still possible to stay at the BS.412 limit most of the time.

Preset 4: A general-purpose preset that works well with most processing presets. When on-air processing preset is designed well, it is still possible to stay at the limit with nearly inaudible BS412 control up to 2dB gain reduction.

Preset 5: The controller gates on nearly every power-drop so that the release rate is almost always very slow. Designed to act only as a protection limiter to sound nearly like no BS412 controller is working at all while still getting loudness benefits from it. BS.412 gain reduction of up to 3dB is possible without objectionable side effects.

Test Modes

Setup: Test				
Parameter Labels	**Units**	**Default**	**Range (CCW to CW)**	**Step**
Mode	—	Operate	Operate, Bypass, Tone	—
Bypass Gain	dB	0.0	–18 … +25	1
Tone Frequency	Hz	400	16, 20, 25, 31.5, 40, 50, 63, 80, 100, 125, 160, 200, 250, 315, 400, 500, 630, 800, 1000, 1250, 1600, 2000, 2500, 3150, 4000, 5000, 6300, 8000, 9500, 10000, 12500, 13586.76, 15000	LOG
Tone Mod. Level	%	91	0 … 100	1
Tone Mod. Type	—	L+R	L+R, L–R, LEFT; RIGHT	—
Pilot	—	ON	ON, OFF	—

Table 6-1: Test Modes

The Test Modes screen allows you to switch between OPERATE, BYPASS, and TONE. When you switch to BYPASS or TONE, the preset you have on air is saved and will be restored when you switch back to OPERATE.

The MODULATION MODE setting in the INPUT/OUTPUT > COMPOSITE screen determines the stereo/mono mode. The choices are STEREO, MONO-L, MONO-R, MONO-SUM, AND SSB.

Table 6-1: Test Modes shows the facilities available, which should be self-explanatory.

Section 7: Specifications

It is impossible to characterize the listening quality of even the simplest limiter or compressor based on specifications, because such specifications cannot adequately describe the crucial dynamic processes that occur under program conditions. There- fore, the only way to evaluate the sound of an audio processor meaningfully is by subjective listening tests.

Certain specifications are presented here to assure the engineer that they are reasonable, to help plan the installation, and make certain comparisons with other processing equipment.

Performance

Except as noted in the text, specifications apply for measurements from analog left/right input to stereo composite output and to FM analog left/right output.

Frequency Response (Bypass Mode): Follows standard 50µs or 75µs pre-emphasis curve ±0.10 dB, 2.0 Hz–15 kHz. Analog left/right output and digital output can be user- configured for flat or pre-emphasized output.

Noise: Output noise floor will depend upon how much gain the processor is set for (Limit Drive, AGC Drive, Two-Band Drive, and/or Multi-Band Drive), gating level, equalization, noise reduction, etc. The dynamic range of the A/D Converter, which has a specified overload-to–noise ratio of 110 dB, primarily governs it. The dynamic range of the digital signal processing is 144 dB.

Total System Distortion (de-emphasized, 100% modulation): <0.01% THD, 20 Hz–1 kHz, rising to <0.05% at 15 kHz. <0.02% SMPTE IM Distortion.

Total System L/R Channel Separation: >50 dB, 20 Hz – 15 kHz; 60 dB typical.

Polarity (Two-Band and Bypass Modes): Absolute polarity maintained. Positive-going signal on input will result in positive-going signal on output when HD Polarity and FM polarity controls are set to POSITIVE.

Processing Sample Rate: The 5750 is a “multirate” system, using internal rates from 64 kHz to 512 kHz as appropriate for the processing being performed. Audio clippers operate at 256 kHz (and are anti-aliased), while the composite limiter operates at 512 kHz.

Peak Control at HD Output: The peak limiter is oversampled at 256 kHz, yielding a worst- case overshoot of 0.5 dB at the analog output and for all output sample rates. (To achieve this performance at 32 kHz output sample rate, it is necessary to set the 5700i’s HD lowpass filter cutoff frequency to 15 kHz.)

Processing Resolution: Internal processing has 24 bit (fixed point) or higher resolution.

Delay

Defeatable Analog FM Diversity Delay: 0.27 to 12.0 seconds, adjustable in one-sample increments at 64 kHz sample rate to allow the delays of the analog and digital paths in the HD Radio system to be matched at the receiver output. When the Diversity Delay Mode = FM_HD, to avoid introducing artifacts in the analog FM path, its delay is held constant and HD delay is varied if the adjustment does not exceed ±1.3 seconds. Larger adjustments change the analog FM delay and re-center the HD delay within the ±1.3 second window. When the Diversity Delay Mode = FM, the delay adjustments occur in the FM path with minimum delay in the HD path.

DAB+ Mode Delay: 0.365015625 to 8.0 seconds (FM path); 0 to 6.0 seconds (HD path). DAB+ mode is triggered by setting any output to HD+DELAY. FM_HD Diversity Delay Mode is unavailable when the delay is in DAB+ mode.

Minimum Processing Delay: Processing structure dependent. Typically 17 ms for normal latency Five-band, 13 ms for low-latency Five-band, 3.7 ms for ultra-low-latency Five- band, and 17 or 22 ms for 2-band, depending on crossover structure chosen. MX presets have approximately 270 ms delay. The multipath mitigator adds 146 ms of additional delay, and can be bypassed in situations like outside broadcasts where talent needs to monitor off-air. The defeatable subharmonic synthesizer adds 67.5 ms of delay when active.

Headphone Monitor Processor Delay: The low-delay, dedicated headphone monitor processor has 5 ms of delay and provides a complete FM processing chain, including 5- band compressor and distortion-cancelled FM clipper. The clipper drive is adjustable to allow dialing in the preferred amount of “FM clipper sound.”

Analog Audio Input

Configuration: Stereo.

Impedance: >10kΩ load impedance, electronically balanced[1].

Nominal Input Level: Software adjustable from –9.0 to +13.0 dBu (VU) / –2.0 t- +20.0 dBu (PPM)

Maximum Input Level: +27 dBu.

Connectors: Two XLR-type, female, EMI-suppressed. Pin 1 chassis ground, Pins 2 (+) and 3 (-) (electronically balanced, floating and symmetrical.

A/D Conversion: 24 bit 128x oversampled delta sigma converter with linear-phase anti- aliasing filter.

Filtering: RFI filtered, with high-pass filter at 0.15 Hz.

Analog Audio Output

Configuration: Stereo. Flat or pre-emphasized (at 50µs or 75µs), software-selectable.

Source Impedance: 50Ω, electronically balanced and floating.

Load Impedance: 600Ω or greater, balanced or unbalanced. Termination not required or recommended.

Output Level (100% peak modulation): Adjustable from –6 dBu to +24 dBu peak, into 600Ω or greater load, software-adjustable.

Signal-to-Noise: >= 90 dB unweighted (Bypass mode, de-emphasized, 20 Hz–15 kHz bandwidth, referenced to 100% modulation).

L/R Crosstalk: <= –70 dB, 20 Hz–15 kHz.

Distortion: <= 0.01% THD (Bypass mode, de-emphasized) 20 Hz–15 kHz bandwidth.

Connectors: Two XLR-type, male, EMI-suppressed. Pin 1 chassis ground, Pins 2 (+) & 3 (-) electronically balanced, floating and symmetrical.

D/A Conversion: 24 bit 128x oversampled, with high-pass filter at 0.15 Hz (–3 dB).

Filtering: RFI filtered.

Digital Audio Input

Configuration: Stereo per AES3 standard, 24 bit resolution, software selection of stereo, mono from left, mono from right or mono from sum.

Sampling Rate: 32, 44.1, 48, 88.2, or 96 kHz, automatically selected. Can be used as a sync reference.

Connector: XLR-type, female, EMI-suppressed. Pin 1 chassis ground, pins 2 (+) and 3 (-) trans- former balanced and floating, 110Ω impedance.

Input Reference Level: Variable within the range of –30 dBFS to –7 dBFS (VU)./ -23 dBFS to 0.0 dBFS

Filtering: RFI filtered.

Digital Audio Outputs

Configuration: Two outputs, each stereo per the AES3 standard. The outputs can be independently set to emit the analog FM processed signal, the digital radio processed signal, the low-delay monitor signal or a 384/192 kHz digital composite baseband output.

The FM processed signal can be configured in software as flat or pre-emphasized to the chosen processing pre-emphasis (50µs or 75µs). The digital radio processing chain receives the output of the multiband limiter and processes it through a look-ahead peak limiter that operates in parallel with the main FM peak limiting system. The DR and FM signals are always simultaneously available.

The digital composite baseband is internally sampled at 384 kHz and alternate samples are placed on the left and right audio channels. This is equivalent to complex (I/Q) sampling and, with compatible receiving hardware, allows the full frequency baseband frequency range to be accommodated without aliasing. If the frequency range of the SCAs applied to the digital output is limited to less than 96 kHz, this is fully compatible and interoperable with the existing system adopted by the industry, which uses only the left channel at 192 kHz.

Sample Rate: Internal free running at 32 kHz, 44.1 kHz, 48 kHz, 88.2 kHz, or 96 kHz, se- lected in software. (Use 44.1 kHz or higher for best peak control.) Can also be synced to AES_IN1, AES_IN2 at 32 kHz, 44.1 kHz, 48 kHz, 88.2 kHz, and 96 kHz or Wordclock/10 MHz Sync Reference input as configured in software. Digital composite baseband configuration runs standard at 192 kHz

Word Length: Software selected for 24, 20, 18, 16 or 14-bit resolution. First-order highpass noiseshaped dither can be optionally added, dither level automatically adjusted appropriately for the word length.

Connector: XLR-type, male, EMI-suppressed. Pin 1 chassis ground, pins 2 and 3 trans- former balanced and floating, 110Ω impedance.

Output Level (100% peak modulation): –23.0 to 0.0 dBFS software controlled.

Filtering: RFI filtered.

Frequency Response (Digital Audio Output (from Digital Radio Processing Chain)):

For output sample rates of 44.1 kHz and above, the frequency response from input to DR-configured output is ±0.10 dB, 2.0 Hz - 20 kHz; flat. The user may specify lowpass filtering to constrain the bandwidth to 15, 16, 17, 18, or 19 kHz.

Relative Time Delay between FM and HD Outputs: Depends on setting of analog processing channel diversity delay. Once set, this delay is constant regardless of processing preset in use.

Wordclock/10 MHz Sync Reference

Configuration: Accepts 1x wordclock or 10 MHz reference signals, automatically selected. The DSP master clock can be phase-locked to these signals, which in turn phase-locks the 19 kHz pilot tone frequency, facilitating single-frequency network operation. The digital output sample frequency can also be locked to these signals.

Level: Unit will lock to 1x wordclock and 10 MHz squarewaves and sinewaves having a peak value of 0.5 V to 5.0 V.

Connector: BNC female, grounded to chassis, non-terminating to allow reference signals to be looped through via an external BNC “tee” connector (not supplied).

Audio-Over-IP I/O (AoIP)

Standard:

- With Dante Module: Fully supports Dante networks. SMPTE ST2110-30 RTP AES67 compliant
- With AES67 Module: AES67 SMPTE ST-2110-30/31, NMOS4, NMOS5 and seamless switching ST2022-7, RAVENNA compliant

Number of Input Channels Supported: Two (2) stereo pairs.

Number of Output Channels Supported: Four (4) stereo pairs.

Sample Rate: 44.1, 48, 88.2 and 96 kHz.

Networking: Two RJ45 Ethernet connectors for connection to dedicated audio-over-IP LANs (supports DANTE redundancy if Dante module is used). These connections are independent of the Optimod’s main Remote Computer Interface and have their own IP and MAC addresses. These are automatically assigned and can be discovered in Dante Controller (Dante Module) or JSON API, NMOS IS-05, ANEMAN & Web UI (AES67 Module)

Composite Baseband Outputs

Configuration: Two outputs, each with an independent software-controlled output level control, output amplifier and connector.

Source Impedance: 0Ω voltage source or 75Ω, jumper-selectable.

Load Impedance: 37Ω or greater. Termination not required or recommended.

Maximum Output Level: +16.0 dBu (13.82Vp-p).

Pilot Level: Adjustable from 6.0% to 12.0%, software controlled.

Pilot Stability: 19 kHz, ±1.0 Hz (10 degrees to 40 degrees C).

D/A Conversion: 24-bit

Signal-to-Noise Ratio: >= 85 dB (Bypass mode, de-emphasized, 20 Hz – 15 kHz band- width, referenced to 100% modulation, unweighted).

Distortion: <= 0.05% THD (Bypass mode, de-emphasized, 20 Hz – 15 kHz bandwidth, referenced to 100% modulation, unweighted).

Stereo Separation: At 100% modulation = 3.5Vp-p, >60 dB, 30 Hz – 15 kHz. At 100% modulation = 1.0 - 8.0 Vp-p, >55 dB.

Crosstalk-Linear: <= –80 dB, main channel to sub-channel or sub-channel to main channel (referenced to 100% modulation).

Crosstalk-Non-Linear: <= –80 dB, main channel to sub-channel or sub-channel to main channel (referenced to 100% modulation).

38 kHz Suppression: >= 70 dB (referenced to 100% modulation).

76 kHz & Sideband Suppression: >= 80 dB (referenced to 100% modulation).

Pilot Protection: >-90 dB relative to 9% pilot injection, ±250 Hz (up to 2 dB composite processing drive).

Subcarrier Protection (60-100 kHz): with up to 2 dB composite limiting drive: -80 dB, -90 dB without composite limiting (referenced to 100% modulation; measured with 4K FFT analyzer using "maximum peak hold with 3 averages" display).

57 kHz (RDS/RBDS) Protection: 50 dB relative to 4% subcarrier injection, ±2.0 kHz (up to 2 dB composite processing drive). 55dB without composite limiting.

Connectors: Two BNC, shell connected to chassis ground, EMI suppressed.

Maximum Load Capacitance: 0.047 microfarad (0Ω source impedance). Maximum cable length of 100 feet/30 meters RG–58A/U.

Filtering: RFI filtered.

Subcarrier (SCA) Inputs

Number of Inputs: 2 x digitized analog.

Resolution: 16-bit conversion.

Impedance: > 600Ω

SCA Sensitivity: Variable from Off, -30 dB … +10 dB to produce 10% injection assuming 100% modulation = 4 V p-p at the analog composite outputs.

Connectors: Two BNC, shell connected to chassis ground, EMI suppressed.

Configuration: Two subcarrier inputs each with an independent software-controlled input level are summed into the digital composite and analog composite outputs. Composite level control settings have no effect on the absolute subcarrier levels.

19 kHz Pilot Reference: SCA2 input can be re-jumpered to provide a 19 kHz pilot reference output.

µMPX Codec (optional)

Supported bitrates: 320, 384, 448, 576 kbps. Supports error correction, redundant connections, multicast/broadcast connections

Streaming Audio Monitor (optional)

Audio Codecs: standard MPEG Layer 3 (MP3) and OPUS.

Audio Bitrate: 32, 64, 96, 128,192, 256, 320 kbps

Audio Sample Rate: 32, 44.1, 48 kHz

Audio Channels: Mono, Stereo

Streaming Servers: SHOUTcast2 and Icecast2 streaming protocols.

Streaming Transports: Ultravox 2.1 protocol (SHOUTcast2 Server), HTTP protocol (Icecast2 Server)

Local: Built-in Icecast2 streaming server.

Emergency Player (optional)

Internal Storage: 2 GB Flash Memory

Audio Storage: 2 hours linear audio (.wav), 12 hours of (MP3 or OPUS)

Ratings Encoder:

Supported watermark encoders: 2 (one for FM, one for Digital Media)

Supported systems: Nielsen, Kantar, IPSOS

Remote Computer Interface

Configuration: TCP/IP configured and controlled via any modern HTML5 web browser via Ethernet interface.

Ethernet Connector: Female RJ45 connector for 10 Mbps and higher networks using CAT5 cabling. Native rate is 100 Mbps. Provides for connection to any modern computer through either a network, or, using a crossover Ethernet cable, directly to a computer.

Ethernet Networking Standard: TCP/IP, HTTP port 80 (user configurable), HTTPS port 443 (user configurable).

Remote Control (GPI) Interface

Configuration: Eight (8) inputs, opto-isolated and floating.

Voltage: 6–15V AC or DC, momentary or continuous. +12VDC provided to facilitate use with contact closure.

Connector: DB–25 male, EMI-suppressed.

Control: User-programmable for any eight of user presets, factory presets, bypass, test tone, stereo or mono modes, analog input, digital input.

Filtering: RFI filtered.

Tally Outputs

Circuit Configuration: Two NPN open-collector outputs.

Voltage: +15 volts maximum. Do not apply negative voltage. When driving a relay or other inductive load, connect a diode in reverse polarity across the relay coil to protect the driver transistors from reverse voltage caused by inductive kickback.

Current: 30 mA maximum

Indications: Tally outputs can be programmed to indicate a number of different operational and fault conditions, including Input: Analog, Input: Digital, Analog Input Silent, AES In- put Silent, and AES Input Error.

Power

Voltage: 80–264 VAC, 50–60 Hz, <65 VA.

Connectors: Two (2) IEC, EMI-suppressed. Detachable 3-wire power cord supplied.

Fuse: T2A Quick Acting HBC, mounted on the power supply circuit board.

Grounding: In order to meet EMI standards, circuit ground is hard-wired to chassis ground.

Safety Standards: ETL listed to UL standards, CE marked.

Environmental

Operating Temperature: 32° to 122° F / 0° to 50° C for all operating voltage ranges.

Humidity: 0–95% RH, non-condensing.

Dimensions (W x H x D): 19” x 1.75” x 14.25” / 48.3 cm x 4.44 cm x 36.2 cm. One rack unit high.

Humidity: 0–95% RH, non-condensing.

RFI/EMI: Tested according to Cenelec procedures. FCC Part 15 Class A device.

Shipping Weight: 17.64 lbs / 8 kg

Warranty

Five Years, Parts and Service: Subject to the limitations set forth in Orban's Standard Warranty Agreement. See page 1-5.

Because engineering improvements are ongoing, specifications are subject to change with- out notice.