OPERATING MANUAL

OPTIMOD TRIO

Multi-Mode Digital Audio Processor

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

Model Number:
OPTIMOD TRIO-AM TRIO

Description:
AM or FM or DM Digital Audio Processor OPTIMOD TRIO, 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 May 2024.

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 TRIO-AM only)
   - The COAX Connecting Cable (THE OPTIMOD TRIO-AM-TRIO and OPTIMOD TRIO-AM-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 TRIO-AM 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 TRIO-AM-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 TRIO-AM TRIO is a 1 RU (rack unit) broadcast audio processor designed for use with analog AM or FM broadcasts, or 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 for all three modes (AM, FM or HD/DAB/Streaming)

The OPTIMOD TRIO-AM TRIO 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 TRIO-AM TRIO. A small investment of your time now will yield large dividends in audio quality.

Features of the OPTIMOD TRIO-AM TRIO 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.
- An RDS/RBDS on-board encoder that supports dynamic PS scrolling and IP access in the FM version.
- 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 TRIO 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.
- Neilsen and Kantar watermarking encoders are available, allowing the processing to be independently watermarked.
- The 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 for FM.

- High resolution front panel touch screen for ease of navigation.

Input/output Configuration

THE OPTIMOD TRIO-AM-TRIO 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
- Analog and AES3 composite stereo outputs.
- Subcarrier (SCA and RDS/RBDS) input for FM.
- 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 TRIO-AM-TRIO simultaneously accommodates digital and analog inputs and outputs.

You can switch any of the TRIO's outputs between the analog-channel processing, and a low-delay monitor signal. You select whether THE OPTIMOD TRIO-AM-TRIO 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 TRIO'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 TRIO-AM'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 10k□;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 (FM Oonly)

The TRIO 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 (FM Only)

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 TRIO 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 TRIO'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 (FM Only)

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 TRIO 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 TRIO-AM'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 TRIO.

Computer Interface

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

- **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 TRIO 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.

Installing the TRIO

Allow about 2 hours for installation.

Installation consists of: (1) unpacking and inspecting the TRIO, (2) mounting the TRIO in a rack, (3) connecting inputs, outputs and power, (4) optional connecting of remote control leads and (5) optional connecting of computer interface control leads.

When you have finished installing the TRIO, proceed to "Quick Setup," on page 2-**Error! Bookmark not defined.**.

Unpack and inspect.

If you note obvious physical damage, contact the carrier immediately to make a damage claim. Packed with the OPTIMOD TRIO-AM-TRIO are:

2ea. Line Cords (domestic US, European)

4ea. Rack-mounting screws, 10-32 x ½ — with washers, #10

Save all packing materials! If you should ever have to ship the TRIO (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.

Complete the Registration Card and return it to Orban. (please)

The Registration Card enables us to inform you of new applications, performance improvements, software updates, and service aids that may be developed, and it helps us respond promptly to claims under warranty without our having to request a copy of your bill of sale or other proof of purchase. Please fill in the Registration Card and send it to us today.

Customer names and information are confidential and are not sold to anyone.

Install the appropriate power cord.

Check the power cord.

AC power passes through an IEC-standard mains connector and an RF filter designed to meet the standards of all international safety authorities.

The power cord is terminated in a "U-ground" plug (USA standard), or CEE7/7 plug (Continental Europe), as appropriate to your TRIO's Model Number. The green/yellow wire is connected directly to the TRIO chassis.

If you need to change the plug to meet your country's standard, we suggest purchasing a new mains cord with the correct line plug attached.

The TRIO uses a universal switching power supply that will operate without readjustment from 95 to 264 volts AC, 50 or 60 Hz.

Always place surge protection circuits as close as possible to the OPTIMOD TRIO-AM-TRIO. 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 OPTIMOD TRIO-AM-TRIO), 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?

Mount the TRIO in a rack.

The TRIO requires one standard rack unit (1¾ inches / 4.45 cm).

There should be a good ground connection between the rack and the TRIO chassis — check this with an ohmmeter 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.

PIN ASSIGNMENT

1.	COMMON	
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.	POWER COMMON	
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	

REMOTE INTERFACE

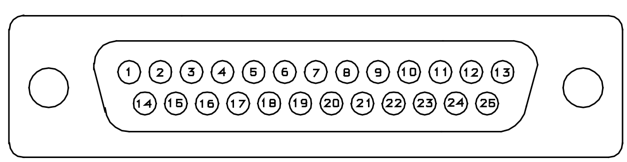
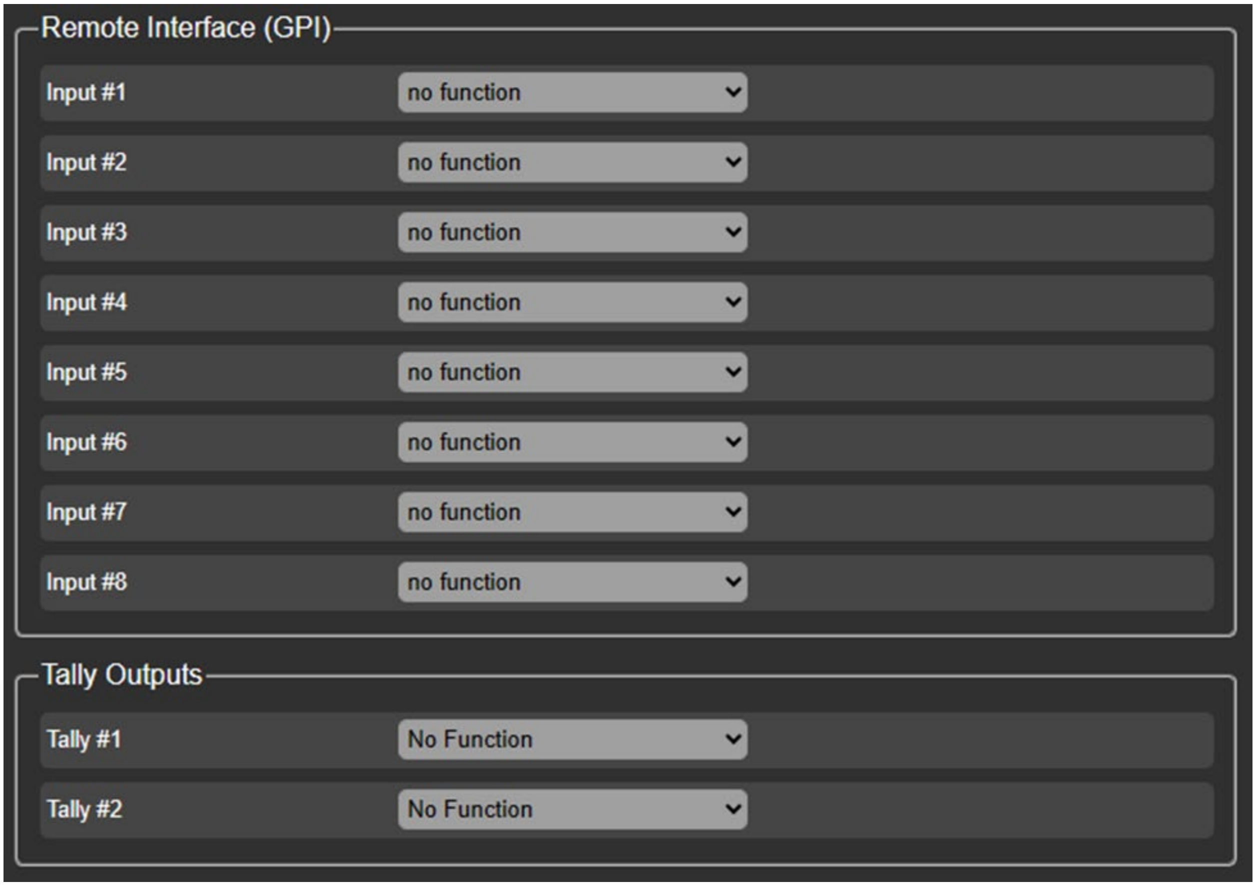


Figure 1-1: Wiring the 25-pin Remote Interface Connector.



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 OPTIMOD TRIO-AM TRIO-AM, 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) In the HTML5 browser connection, 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.

ENERGIZE YOUR TRIO

All the way to the right of the rear panel, you will find two power inlets. For setup mode it is not necessary to energize both, however, you will want to perform this task before you place the processor on air.

THE TRIO FRONT PANEL

The TRIO 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 TRIO. 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 failure. The type of failure is indicated by the cadence of the LED
- 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 TRIO 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) Monitoring current TRIO mode (AM or FM or HD)

  B) Analog In

  C) Digital In 1

  D) Digital in 2

  E) AoIP In 1

  F) AoIP In 2

  G) Stream In

Use the up and down dobby 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.

MAIN MENU

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

INFO

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

- System Version
- I/O Board Revision
- Aux PIC
- Power Pic
- Serial Number
- MAC Address
- IP, Subnet and Gateway addresses
- DNS
- IPv6 Address
- Hostname

Touching the > box will reveal the following information

- CPU Temperature
- MB Temperature
- Fan Speed
- Free Memory Flash #1
- Free Memory Flash #2
- Free Memory
- Voltages
- Power Supply Power Supply 1 and 2 voltages
- Power Supply Status

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 TRIO.
- 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.

The TRIO Rear Panel

The OPTIMOD TRIO-AM-TRIO rear panel lays out I/O for all three modes it is capable of performing.

From left to right, these are

- SCA 1 INPUT (FM Only)
- SCA 2 INPUT/PILOT REFERENCE (FM Only)

- COMPOSITE OUT 1 (FM Only)
- COMPOSITE OUT 2 (FM Only)
- 10MHz REFERENCE IN
- ANALOG LEFT INPUT
- ANALOG RIGHT INPUT
- ANALOG OUT LEFT
- ANALOG OUT RIGHT
- DIGITAL 1 INPUT
- DIGITAL 2 INPUT (AES Sync)
- DIGITAL 1 OUTPUT
- DIGITAL 2 OUTPUT
- STREAMING PORT/µMPX
- ETHERNET CONTROL
- AOIP (Audio over IP)
- AC MAINS (There are two power supplied, when ready for use on air, both should be energized. Front panel LED's will alarm if only one supply is active or has power.

CONNECTING VIA COMPUTER

Now that you have setup the the IP address, time and input source, it's time to connect to the TRIO using the Ethernet port.

The default address for the TRIO is 192.168.254.254 with a subnet of 255.255.255.0 and the gateway address 0.0.0.0. You may assign a static IP address of your own using standard network protocols. Please use a switch between the PC and TRIO when connecting to set up a proper network. See page 1-12 for information on how to set the IP address from the front panel.

Section 2: OPTIMOD TRIO-AM

When you first energize your OPTIMOD Trio

During the boot process, you will be asked to pick which mode you want to use. For this section, choose AM. The TRIO will ask to confirm you selection. Press OK. The TRIO will notw enter AM mode.

The OPTIMOD TRIO-AM TRIO-AM Digital Audio Processor

Orban's all-digital OPTIMOD TRIO-AM TRIO-AM Audio Processor can help you achieve the highest possible audio quality in monophonic AM shortwave, medium wave and long wave broadcasts. OPTIMOD TRIO-AM TRIO-AM delivers louder, cleaner, brighter, FM-like audio with an open, fatigue-free quality that attracts listeners and holds them. Because all processing is performed by high-speed mathematical calculations, the processing has cleanliness, quality, and stability over time and temperature that is unmatched by analog processors.

OPTIMOD TRIO-AM TRIO-AM is descended from the industry-standard 9000 series of OPTIMOD TRIO-AM TRIO-AM audio processors. Thousands of these processors are on the air all over the world. They have proven that the “OPTIMOD TRIO-AM sound” attracts and keeps an audience even in the most competitive commercial environment.

Take a little time now to familiarize yourself with OPTIMOD TRIO-AM TRIO-AM. A small investment of your time now will yield large dividends in audio quality.

OPTIMOD TRIO-AM TRIO-AM was designed to deliver a high-quality FM-like sound to the listener's ear by pre-processing for the limitations of the average car or table radio (while avoiding audible side effects and compromises in loudness or coverage). Because such processing can make audible many defects ordinarily lost in the usual sea of AM mud, it is very important that the source audio be as clean as possible.

For best results, *feed OPTIMOD TRIO-AM TRIO-AM unprocessed audio.* No other audio processing is necessary or desirable.

Making the Most of the AM Channel

- OPTIMOD TRIO-AM TRIO-AM **rides gain** over an adjustable range of up to 25dB, compressing dynamic range and compensating for operator gain-riding errors and for gain inconsistencies in automated systems.
- OPTIMOD TRIO-AM TRIO-AM **increases the density and loudness of the program material** by multiband limiting and multiband distortion-canceling clipping, improving the consistency of the station's sound and increasing loudness and definition without producing audible side effects.
- OPTIMOD TRIO-AM TRIO-AM **precisely controls peak levels** to prevent overmodulation.
- OPTIMOD TRIO-AM TRIO-AM **compensates for the high- and low-frequency rolloffs of typical AM receivers** with a fully adjustable program equalizer providing up to 20dB of high-frequency boost (at 5 kHz) without

producing the side effects encountered in conventional processors. This equalizer can thus produce extreme preemphasis that is appropriate for very narrow-band radios. OPTIMOD TRIO-AM TRIO-AM's fully parametric low- and mid-frequency equalizers allow you to tailor your air sound to your precise requirements and desires. OPTIMOD TRIO-AM TRIO-AM also fully supports the NRSC standard preemphasis curve.

Controllable and Adjustable

- OPTIMOD TRIO-AM TRIO-AM comes with a **wide variety of factory presets** to accommodate almost any user requirement. A single LESS-MORE control easily modifies any factory preset. The user (via FULL CONTROL) can further customize the presets, and these can be stored and recalled on command. Advanced Control (accessible from the PC Remote application) facilitates detailed sound design using the same controls that were available to the factory programmers.

- An **LCD** and **full-time LED meters** make setup, adjustment and programming of OPTIMOD TRIO-AM TRIO-AM easy — you can always see the metering while you're adjusting the processor. Navigation is by dedicated buttons, soft buttons (whose functions are context-sensitive), and a large rotary knob. The LEDs show all metering functions of the processing structure (Two-Band or Five-Band) in use.

- OPTIMOD TRIO-AM TRIO-AM contains a versatile **real-time clock**, which allows automation of various events (including recalling presets) at pre-programmed times.

- A Bypass Test Mode can be invoked locally, by remote control (from either the TRIO's GPI port or the TRIO PC Remote application), or by automation to permit broadcast system **test and alignment** or "proof of performance" tests.

- OPTIMOD TRIO-AM TRIO-AM contains a built-in **line-up tone generator** that offers sine, square, and triangle waves, facilitating quick and accurate level setting in any system.

- **OPTIMOD TRIO-AM TRIO-AM** Uses an HTML5 web browser interface and offers a complete tool set to monitor and measure your audio.

- The TRIO **can be remote-controlled** by 5-12V pulses applied to eight programmable, optically isolated "general-purpose interface" (GPI) ports.

Versatile Installation

- OPTIMOD TRIO-AM TRIO-AM controls the **transmitter bandwidth as necessary to meet government regulations**, regardless of program material or equalization. OPTIMOD TRIO-AM TRIO-AM's high-frequency bandwidth can be switched instantly in 500Hz increments between 4.5 kHz and 9.5 kHz. The lower cutoff frequencies meet the output power spectral density requirements of ITU-R 328-5 without further low-pass filtering at the transmitter, while the 9.5 kHz filter meets the requirements of the NRSC-1 standard (North America).

- OPTIMOD TRIO-AM TRIO-AM **compensates for inaccuracies in the pulse response (tilt, overshoot, ringing) of transmitters and antenna systems** with a powerful four-parameter transmitter equalizer. A built-in square-wave generator makes adjustment easy. Four sets of equalizer parameters can be stored and recalled, allowing you to program day and night variations for two transmitters.

- **Two mono analog outputs** and **one AES3 output** accommodate as many as three transmitters.

- OPTIMOD TRIO-AM TRIO-AM is **usually installed at the transmitter**, replacing all processing normally employed at the transmitter site, including compressor, protection peak limiters, clippers, and high- and low-pass filters normally included within the transmitter. It can also **be installed at the studio** if an uncompressed digital STL is available.

- The TRIO includes **analog** and **AES3 digital** inputs and outputs. Both the digital input and the digital output are equipped with sample-rate converters and can operate at 32 kHz, 44.1 kHz, 48, 88.2, and 96 kHz sample rates. The digital output can be pre-emphasized to the J.17 standard.

- The analog inputs are **transformerless, balanced 10kΩ instrumentation-amplifier circuits**, and the analog outputs are transformerless balanced, and floating (with 351 Ω impedance) to ensure highest transparency and accurate pulse response.

- All input, output, and power connections are **rigorously RFI-suppressed** to Orban's traditional exacting standards, ensuring trouble-free installation.

- The TRIO is designed and certified to **meet all applicable international safety and emissions standards**.

Presets in OPTIMOD TRIO-AM TRIO-AM

There are two distinct kinds of presets in OPTIMOD TRIO-AM TRIO-AM: **factory presets** and **user presets**.

Factory Presets

The Factory Presets are our "factory recommended settings" for various program formats or types. The description indicates the processing structure and the type of processing. Each Factory Preset on the Preset list is really a library of more than 20 separate presets, selected by navigating to MODIFY PROCESSING > > BASIC CONTROLS > LESS-MORE and using the LESS-MORE control to adjust OPTIMOD TRIO-AM TRIO-AM for less or more processing. The factory presets are listed and described on page 3-2-32.

Factory Presets are stored in OPTIMOD TRIO-AM TRIO-AM's non-volatile memory and cannot be erased. You can change the settings of a Factory Preset, but you must then store those settings as a User Preset, which you are free to name as you wish. The Factory Preset remains unchanged.

User Presets

User Presets permit you to change a Factory Preset to suit your requirements and then store those changes.

You can store more than 100 User Presets, limited only by available memory in your TRIO (which will vary depending on the version of your TRIO's software). You can give your preset a name up to 18 characters long.

User Presets cannot be created from scratch. You must always start by recalling a Factory Preset. Make the changes, and then store your modified preset as a User Preset. You can also recall a previously created user preset, modify it, and save it again, either overwriting the old version or saving under a new name. In all cases, the original Factory Preset remains for you to return to if you wish.

Input/Output Configuration

OPTIMOD TRIO-AM TRIO-AM simultaneously accommodates:

A digital AES3 left/right input and an AES3 mono output.

A stereo analog left/right input and two mono outputs.

A Dante AES67 left/right input and two mono outputs.

Digital AES3 Input/Output

The digital input and output conform to the professional AES3 standard. They have sample rate converters to allow operation at 32, 44.1, 48, 88.2, and 96 kHz sample frequency.

The left/right digital input is on one XLR-type female connector on the rear panel; the mono digital output is on an XLR-type male connector on the rear panel.

You can select whether OPTIMOD TRIO-AM TRIO-AM uses its digital, analog or AES67 input either locally or by the HTML5 web server. If OPTIMOD TRIO-AM TRIO-AM is set to accept a digital input and the feed fails, you can configure OPTIMOD TRIO-AM TRIO-AM so that it automatically switches back to one of the other inputs.

The TRIO-AM can process the signal from the left, right, or sum of the left and right channels of either the analog or the digital input.

Level control of the AES3 input is accomplished via the HTML5 web server using your browser.

Analog and digital outputs are active continuously.

The TRIO-AM's output sample rate can be locked either to the TRIO-AM's internal crystal clock or to the sample rate present at its AES3 input.

Analog Input/Outputs

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 +27dBu (0dBu = 0.775Vrms). Although the TRIO-AM's processing is monophonic, we have supplied stereo inputs so that the TRIO-AM can process the L+R, L-only, or R-only signals without needing an external mixer.

The two mono analog outputs are on XLR-type male connectors on the rear panel. Output impedance is 351 Ω; balanced and floating. The outputs can drive 600Ω or higher impedances, balanced or unbalanced. The peak output level is adjustable from –6 dBu to +20 dBu.

Level control of the analog inputs and outputs is accomplished via the HTML5 web server via a browser.

Dante Audio Input/Output

The following instructions describe how to set up your OPTIMOD TRIO-AM'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 TRIO-AM'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 TRIO-AM and configure its IP address. Then the OPTIMOD TRIO-AM 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 TRIO-AM, 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 OPTIMOD TRIO-AM TRIO-AM 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 OPTIMOD TRIO-AM TRIO-AM.

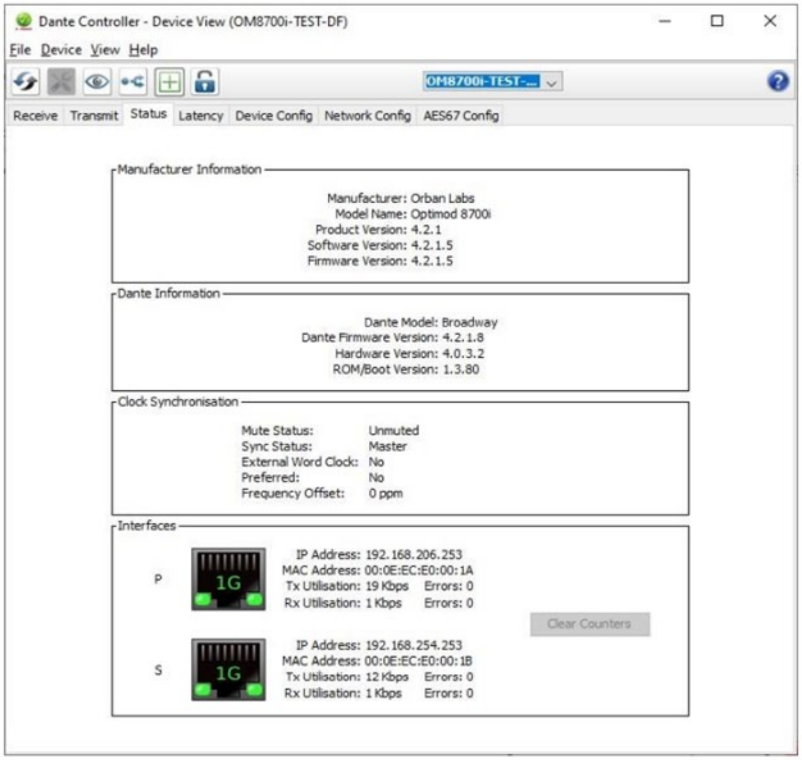


Figure 2-1: A REDUNDANT setup with static IP addresses.

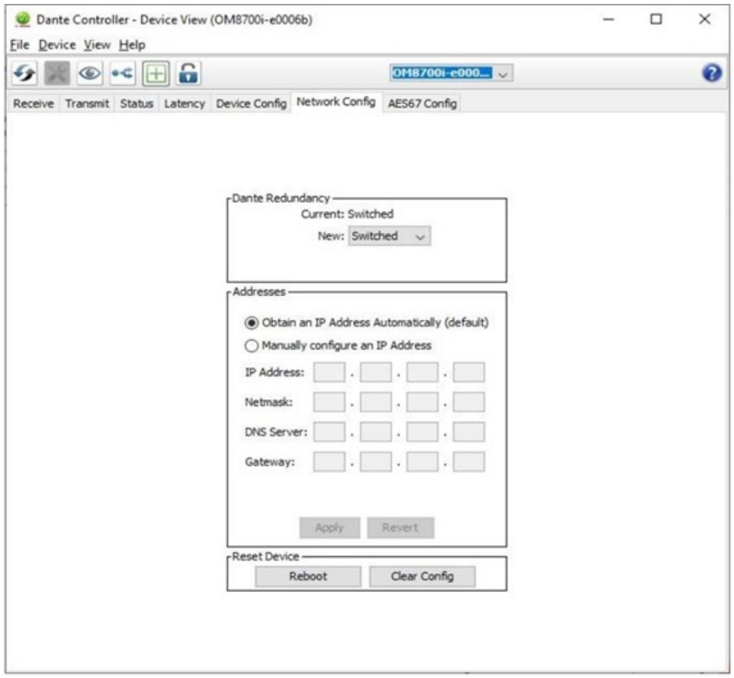


Figure 2-2: A SWITCHED setup with a DHCP-sourced IP address

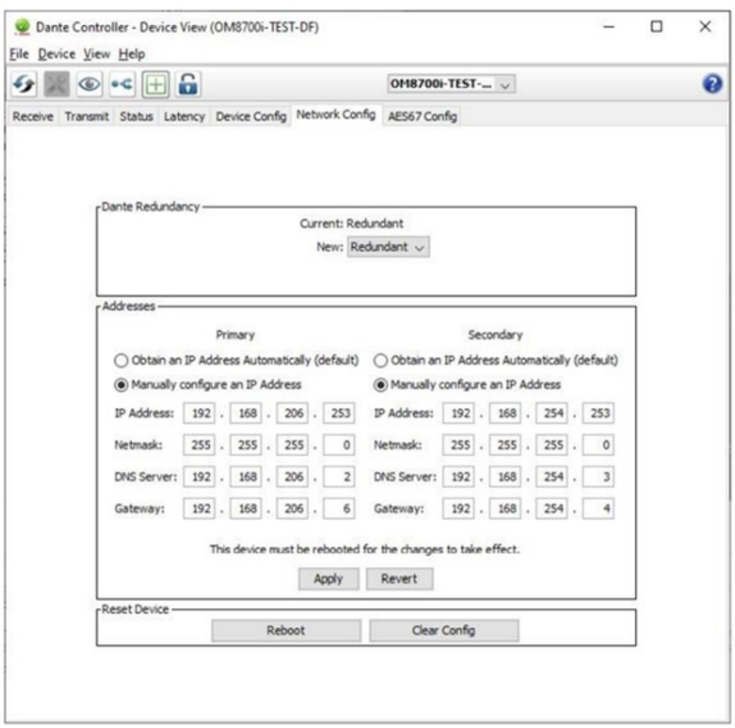
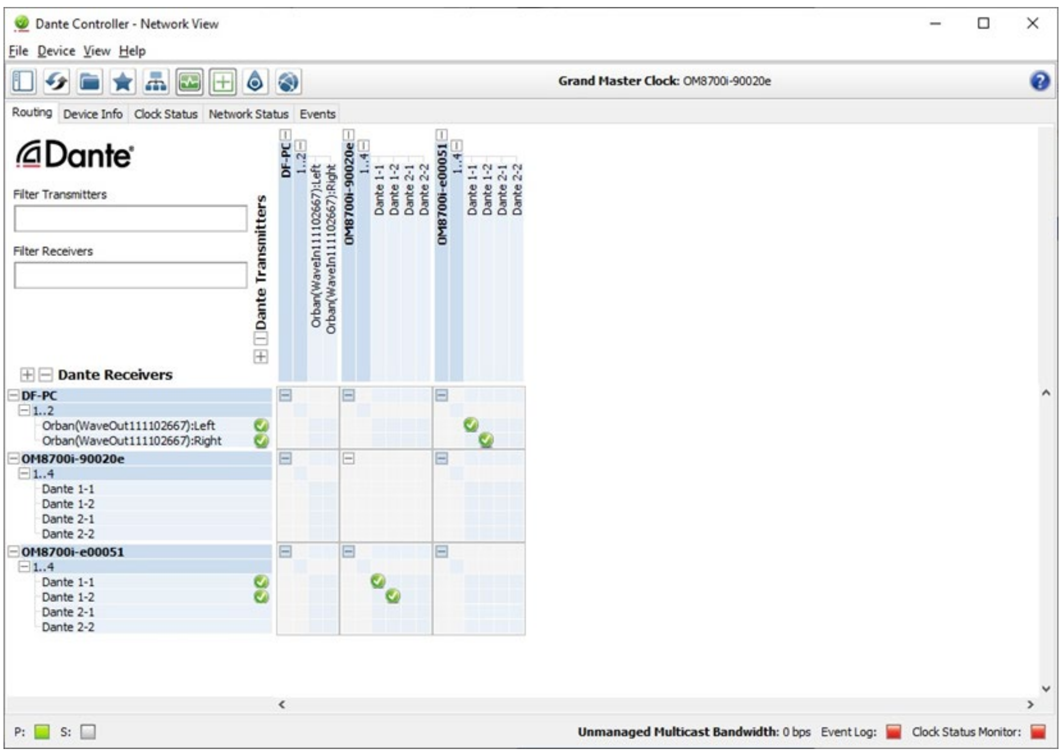


Figure 2-3: A Dante Controller reports the OPTIMOD TRIO-AM TRIO-AM network status when a OPTIMOD TRIO-AM TRIO-AM has been configured and connected for dual-redundant networks.

3) In the Dante network, route audio to and from the OPTIMOD TRIO-AM TRIO-AM.

A Dante device has a number of channels associated with it. These are either transmit (TX) or receive (RX) channels. Your OPTIMOD TRIO-AM 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 OPTIMOD TRIO-AM TRIO-AM'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 OPTIMOD TRIO-AM TRIO-AM'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 TRIO-AM'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 TRIO-AM's output.

4) Configure the OPTIMOD TRIO-AM TRIO-AM's Dante inputs and outputs.

Unlike the sample rate of the OPTIMOD TRIO-AM's AES3 outputs, the OPTIMOD TRIO-AM'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 TRIO-AM 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 OPTIMOD TRIO-AM TRIO-AM 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 or MONITOR,

- 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 TRIO-AM'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 TRIO-AM 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.

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

b) 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 OPTIMOD TRIO-AM TRIO-AM 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.

Remote Control Interface

The Remote Control Interface is a set of eight optically isolated GPI inputs on a DB-25 connector, which can be activated by 5-12V DC. They can control various functions of the TRIO-AM via closures through automation systems or remote controls.

Location of OPTIMOD TRIO-AM TRIO-AM

Optimal Control of Peak Modulation Levels

The audio processing circuitry in OPTIMOD TRIO-AM TRIO-AM produces a waveform that is precisely peak-controlled to prevent overmodulation, and is lowpass filtered to protect adjacent channels and to conform to government regulations. Severe changes in the shape of the waveform can be caused by passing it through a circuit with non-constant group delay and/or non-flat frequency response in the 30-9500Hz range. Deviation from flatness and phase linearity will cause spurious modulation peaks because the shape of the peak-limited waveform is changed. Such peaks add nothing to average modulation. Thus, the average modulation must be lowered to accommodate those peaks so that they do not overmodulate. Transformers can cause such problems.

Landline equalizers, transformers, and low-pass filters in transmitters typically introduce frequency response errors and non-constant group delay. There are three criteria for preservation of peak levels through the audio system:

1) The system group delay must be essentially constant throughout the frequency range containing significant energy (30-9,500Hz). If low-pass filters are present, this may require the use of delay equalization. The deviation from linear-phase must not exceed ±10° from 30-9,500Hz.

2) The low-frequency -3 dB point of the system must be placed at 0.15Hz or lower (this is not a misprint!). This is necessary to ensure less than 1% overshoot in a 50Hz square wave and essentially constant group delay to 30Hz.

3) Any preemphasis used in the audio transmission system prior to the transmitter (such as in an STL) must be canceled by a precisely complementary deemphasis: Every pole and zero in the preemphasis filter must be complemented by a zero and pole of identical complex frequency in the deemphasis network. An all-pole deemphasis network (like the classic series resistor feeding a grounded capacitor) is not appropriate.

In this example, the network could be corrected by adding a second resistor between ground and the capacitor, which would introduce a zero.

Low-pass filters (including anti-aliasing filters in digital links), high-pass filters, transformers, distribution amplifiers, and long transmission lines can all cause the above criteria to be violated, and must be tested and qualified. It is clear that the above criteria for optimal control of peak modulation levels are met most easily when the audio processor directly feeds the transmitter. While OPTIMOD TRIO-AM TRIO-AM's transmitter equalizer can mitigate the effects of group delay and frequency response errors in the signal path, an accurate path will still achieve the best results.

Best Location for OPTIMOD TRIO-AM TRIO-AM

The best location for OPTIMOD TRIO-AM TRIO-AM is as close as possible to the transmitter so that OPTIMOD TRIO-AM TRIO-AM's output can be connected to the transmitter through a circuit path that introduces the least possible change in the shape of OPTIMOD TRIO-AM TRIO-AM's carefully peak-limited waveform. This connection could be short lengths of shielded cable (for transmitters with analog inputs) or a direct AES3 connection (if the transmitter has a digital input available). If this is impossible, the next best arrangement is to feed the TRIO-AM's AES3 digital output through an all-digital, uncompressed path to the transmitter's exciter.

If the programming agency's jurisdiction ends at the link connecting the audio facility to the transmitter, a variety of problems can occur downstream. (The link might be telephone/post lines, analog microwave radio, or various types of digital paths.) The link, the transmitter, the transmitter peak limiters, or the transmitter itself can all introduce artifacts that a studio-located audio processor cannot control.

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 an uncompressed AES3 digital link is available to the transmitter, this is an excellent, accurate means of transmission. However, if the digital link employs lossy compression, it will disturb peak levels by up to 4 dB. Lossy compression is also inappropriate for another reason: it cannot accommodate pre-emphasized audio (like OPTIMOD TRIO-AM TRIO-AM's output) without introducing serious artifacts.

Unlike FM, where the transmitter usually can be set up to provide preemphasis, AM transmitters are universally "flat." Therefore, unlike FM, there is no option when using lossy compression to de-emphasize at the output of OPTIMOD TRIO-AM TRIO-AM and then to restore the preemphasis at the transmitter. The best one can do is to use NRSC preemphasis, apply NRSC deemphasis before the lossy link's input, and then re-apply NRSC preemphasis at the link's output.

If only an analog link is available, use a TRIO-AM's audio output and feed the audio directly into the link. If possible, request that any transmitter protection limiters be adjusted for minimum possible action — OPTIMOD TRIO-AM TRIO-AM 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 OPTIMOD TRIO-AM TRIO-AM at the transmitter site or by connecting it to the transmitter through an uncompressed digital STL.

Because OPTIMOD TRIO-AM TRIO-AM controls peaks, it is irrelevant whether the audio link feeding OPTIMOD TRIO-AM TRIO-AM's input terminals is phase-linear. However, the link should have low noise, the flattest possible frequency response from 30-9,500, and low nonlinear distortion.

Transmission from Studio to Transmitter

There are several 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 in AM service are used in two fundamentally different ways. They can either:

pass unprocessed audio for application to the TRIO-AM's input, or

pass the TRIO-AM's peak-controlled analog or digital audio outputs for application to the transmitter.

These 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 passing processed audio does not 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. Such processing will almost invariably distort peak levels; such links must therefore be carefully qualified before you use them to carry the peak-controlled output of the TRIO-AM to the transmitter. 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.

As stated above, links using lossy data reduction cannot pass an OPTIMOD TRIO-AM TRIO-AM–processed signal without distorting it. However, it is practical (though not ideal) to use lossy data reduction to pass *unprocessed* audio to the TRIO-AM'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 TRIO-AM 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 TRIO-AM’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 384 kB/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 if they do not truncate the digital words produced by the devices driving their inputs. Because the TRIO-AM’s output is tightly band-limited to 9.5 kHz or lower (depending on the TRIO-AM’s lowpass filter setting), any link with 32 kHz or higher sample frequency can pass either output without additional overshoot.

Currently available sample rate converters use phase-linear filters. These have constant group delay at all frequencies. Sample rate conversion, whether upward or downward, will not add overshoot to the signal if it does not remove spectral energy from the original signal.

If the link does not have an AES3 input, you must drive its analog input from the TRIO-AM’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.

Analog Microwave STLs

Potential problems include overloads induced by preemphasis and requirements that the audio applied to the microwave transmitter be processed to prevent overmodulation of the microwave system.

Lack of transparency in the path will cause overshoot. Unless carefully designed, analog 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 TRIO-AM 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 TRIO-AM 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 TRIO-AM-PC 1101e is placed before the STL. Discontinued Orban products that would work would be the OPTIMOD TRIO-AM 6200 / 6300 or another OPTIMOD TRIO-AM TRIO in HD/Streaming mode.

Some microwave links can be modified such that the deviation from linear phase is less than $\pm 10°$ from 20 Hz to 9.5 kHz and frequency response is less than 3 dB down at 0.15Hz and less than 0.1 dB down at 20 kHz. 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.

Nevertheless, in an analog microwave system, the TRIO-AM is usually located at the main AM transmitter and is driven by the microwave receiver. One of Orban’s studio level control systems (like our PC 1101e) protects the microwave transmitter at the studio from overload. This unit also performs the gain riding function ordinarily executed by the AGC section of the TRIO-AM's processing and it optimizes the signal-to-noise ratio obtainable from the dual-microwave link.

Further, it is common for a microwave STL to bounce because of a large infrasonic peak in its frequency response caused by an under-damped automatic frequency control (AFC) phase-locked loop. This bounce can increase the STL's peak carrier deviation by as much as 2dB, reducing average modulation. Many commercial STLs have this problem.

AM Transmitters and Antennas

The behavior of an FM station is more or less determined by the behavior of the exciter. Alas, this is not true in AM broadcast! The performance of an AM broadcast station is highly dependent upon the high-power sections of the transmitter and upon the behavior of the antenna system.

The extremely high average power and the pre-emphasized high-frequency component of audio processed by OPTIMOD TRIO-AM TRIO-AM put great demands upon the performance of the transmitter and antenna system. While improved results can be expected from most facilities, outstanding results can only be achieved by facilities having transmitters that can accurately reproduce OPTIMOD TRIO-AM TRIO-AM's output without changing the shape of the waveform, and having wide-band, symmetrical antenna arrays.

Any AGCs, compressors, limiters, and clippers that follow OPTIMOD TRIO-AM TRIO-AM in the circuit should be bypassed. OPTIMOD TRIO-AM TRIO-AM provides all of these functions itself.

Bypassing the Transmitter's Internal Filters and Clippers

Some AM transmitters, especially those supplied to stations outside of North or South America, contain built-in filters and clippers after their audio inputs. The filters may have various purposes: A low-pass filter is often included to ensure that the transmitter's output spectrum adheres to the occupied bandwidth specifications of the governing authority. A high-pass filter may be present to protect the transmitter from damage. Safety clippers are often present to prevent the modulator from being over-driven.

As discussed in earlier sections, accurate reproduction of OPTIMOD TRIO-AM TRIO-AM's output requires that the deviation from linear phase must be less than 10 degrees, 30-9500Hz. Frequency response must be less than 3dB down at 0.15Hz, and less than 0.1dB down at 9.5 kHz.

The highly processed output of OPTIMOD TRIO-AM TRIO-AM is carefully band-limited and peak-controlled. This output will often contain waveforms with flattops like square waves. If the transmitter has constant group delay above 30Hz, these difficult waveforms will be transmitted intact and peak modulation will be accurately controlled.

However, if low-frequency response is more than 3dB down at 0.15Hz, as would be true if a high-pass filter is present, the group delay above 30Hz will not be constant. For example, a typical 50Hz high-pass filter introduces significant non-constant group delay to 500Hz — ten times the cutoff frequency. This non-constant group delay will tilt the flattops produced by OPTIMOD TRIO-AM TRIO-AM. The tilt increases the peak level of the audio waveform, but not the average level. This will force you to decrease the average modulation to prevent the spurious peaks from overmodulating.

Similarly, a typical EBU 4.5 kHz filter will introduce significant non-constant group delay down to 1 kHz about one-fourth the cutoff frequency. This will cause overshoot in the highly processed waveforms produced by OPTIMOD TRIO-AM TRIO-AM. The overshoot increases the peak level of the audio waveform, but not the average level. This will force you to decrease average modulation even more.

Alternatively, if you do not decrease the average modulation to accommodate the spurious peaks introduced by the filters, the transmitter's safety clipper will clip the peaks. This will introduce out-of-band energy that will almost certainly violate the limits on occupied bandwidth specified by the governing authority and will greatly degrade the spectral control provided by OPTIMOD TRIO-AM TRIO-AM.

To achieve the full performance capability built into OPTIMOD TRIO-AM TRIO-AM, any filters in the transmitter must be bypassed. This is essential! OPTIMOD TRIO-AM TRIO-AM contains low-pass and high-pass filters that are fully capable of protecting the transmitter and controlling occupied bandwidth. Because of their location within OPTIMOD TRIO-AM TRIO-AM, the internal filters do not introduce spurious modulation peaks.

Any built-in peak clippers in the transmitter should be defeated. OPTIMOD TRIO-AM TRIO-AM contains a clipping system that is fully capable of controlling transmitter modulation without introducing out-of-band energy. If the drive level to the transmitter is even slightly excessive, the transmitter clipper will be driven hard enough to create excessive spurious spectrum. Defeating any clippers in the transmitter prevents this possibility.

This problem will be even worse if OPTIMOD TRIO-AM TRIO-AM's transmitter equalizer is in use. OPTIMOD TRIO-AM TRIO-AM's output level will frequently exceed 100% modulation because it is pre-distorted to complement the transmitter's pulse response. The transmitter's built-in safety clipper will surely clip this pre-distorted waveform.

Power Supplies

An AM transmitter is required to provide 150% of equivalent unmodulated carrier power when it is modulating 100%. High-voltage power supplies are subject to two major problems: sag and resonance.

Sag is a result of inadequate steady-state regulation. It causes the conventional carrier shift that is seen on a modulation monitor. Good transmitter engineering practice usually limits this shift to -5% (which corresponds to about 0.5dB — not a highly significant loudness loss).

A more serious problem is dynamic carrier shift, or bounce. This has been known to cause up to 3dB loudness loss. Resonances in the power supply's LC filter network usually cause it. Any LC network has a resonant frequency. In order to achieve reasonable efficiency, the power supply filter network must be underdamped. Therefore, high modulation excites this resonance, which can cause overmodulation on the low-voltage peaks of the resonance.

Curing bounce is not straightforward because of the requirement that the power supply filter smooth the DC sufficiently to achieve low hum. One approach that has been employed is use of a 12-phase power supply. Upon rectification, the ripple component of the DC is down about -40dB without filtering. A single-capacitor filter can thus be used, eliminating the filter inductor as a potential source of resonance with the capacitor.

Other sources of resonance include the modulation reactor and modulation transformer in conventional plate-modulated transmitters. Such transmitters will not greatly benefit from a 12-phase power supply.

The newer generations of transmitters employ switching modulation techniques to control bounce far better than do older plate-modulated designs. The latest transmitters using digital modulation techniques have even better performance and most are essentially transparent.

Asymmetry

While the physics of carrier pinch-off limit any AM modulation system to an absolute negative modulation limit of 100%, it is possible to modulate positive peaks as high as desired. In the United States, the FCC permits positive peaks of up to 125% modulation. Many countries have similar restrictions.

However, many transmitters cannot achieve such modulation without substantial distortion, if they can achieve it at all. The transmitter's power supply can sometimes be strengthened to correct this. Sometimes, RF drive capability to the final power amplifier must be increased.

Voice, by its nature, is substantially asymmetrical. Therefore, asymmetrical modulation was popular at one time in an attempt to increase the loudness of voice. Traditionally, this was achieved by preserving the natural asymmetry of the voice signal. An asymmetry detector reversed the polarity of the signal to maintain greater positive modulation. The peaks were then clipped to a level of -100%, +125%.

OPTIMOD TRIO-AM TRIO-AM takes a different approach: OPTIMOD TRIO-AM TRIO-AM's input conditioning filter contains a time dispersion circuit (phase scrambler) that makes asymmetrical input material, like voice, substantially symmetrical.

OPTIMOD TRIO-AM TRIO-AM permits symmetrical or asymmetrical operation of both the safety clipper and multiband distortion-canceling clipper. Asymmetrical clipping slightly increases loudness and brightness, and can produce dense positive peaks up to 125%. However, such asymmetrical processing by its very nature produces both odd and even-order harmonic and IM distortion. While even-order harmonic distortion may sound pleasingly bright, IM distortion of any order sounds nasty.

There is really nothing lost by not modulating asymmetrically: Listening tests easily demonstrate that modulating symmetrically, if time dispersion has been applied to the audio, produces a considerably louder and cleaner sound than does asymmetrical modulation that retains the natural asymmetry of its program material.

Some of the newer transmitters of the pulse-width modulation type have circuitry for holding the carrier shift constant with modulation. Since artificial asymmetry can introduce short-term DC components (corresponding to dynamic upward carrier shift), such carrier shift cancellation circuitry can become confused, resulting in further distortion.

System Presets and Transmitter Equalization

OPTIMOD TRIO-AM TRIO-AM's transmitter equalizer can cure linear problems caused by the transmitter or antenna system. However, the transmitter equalizer cannot cure nonlinear problems, particularly those caused by inadequate power supplies, modulation transformers, or reactors. If any of these components saturate or otherwise fail to perform under heavy power demands, no amount of small-signal equalization will solve their problems.

OPTIMOD TRIO-AM TRIO-AM was designed with the assumption that one audio processor would be devoted to no more than two transmitters, usually called main and standby (or main and alternate). Each transmitter might be required to change power at night or to drive a different antenna array. Only one transmitter is assumed to be on the air at a given time.

To drive two transmitters, OPTIMOD TRIO-AM TRIO-AM provides two mono analog outputs (called Analog Output 1 and Analog Output 2) and one AES3 digital output, which can alternatively be used to drive the main transmitter if it has a digital input.

OPTIMOD TRIO-AM TRIO-AM provides four system presets for its transmitter equalizer controls and certain other controls. Only one preset can be active at a given time; all three outputs receive the same transmitter equalization. This is consistent with the principle that only one transmitter will be on the air.

Transmitter equalizer controls in a given system preset include:

LF Gain for the LF tilt equalizer [LF GAIN]

LF Breakpoint Frequency for the LF tilt equalizer [LF FREQ]

HF Shelf tuning [HF FREQ]

HF Delay equalization [HF DELAY]

System presets also contain the following controls:

System Lowpass Filter Cutoff Frequency [LOW PASS]

System Lowpass Filter Cutoff Shape [LPF SHAPE]

System Highpass Filter Cutoff Frequency [HIGH PASS]

Positive Peak Threshold (Asymmetry) [POS PEAK]

For convenience and to describe their most common application, the four transmitter equalizer presets are labeled TX1/DAY, TX1/NIGHT, TX2/DAY, and TX2/NIGHT, although they can be applied in a completely general way to the requirements of your transmission facility.

For example, in countries observing NRSC standards you might want to transmit the full 9.5 kHz bandwidth during the day, and, in cooperation with other stations on first-adjacent channels, reduce audio bandwidth to 5 kHz at night. This will eliminate any skywave-induced monkey-chatter interference between first-adjacent channels. Alternatively, your nighttime directional antenna array might have poor VSWR performance at high modulating frequencies, so you might find that your transmitter works better and produces less distortion if you limit the audio bandwidth to those frequencies where the antenna is well behaved. Further, if you operate a talk format during certain parts of the day, you will probably find that you can operate the processing for a louder on-air sound if you restrict the transmitted bandwidth below the maximum permitted by government regulation. (Bear in mind that most AM radios have an audio bandwidth of 2.5 to 3 kHz and changing transmission bandwidth from 5 kHz to 9.5 kHz will produce virtually no audible difference on these radios.)

Antenna System

AM antenna systems, whether directional or non-directional, frequently exhibit inadequate bandwidth or asymmetrical impedance. Often, a system will exhibit both problems simultaneously.

An antenna with inadequate bandwidth couples RF energy into space with progressively less efficiency at higher sideband frequencies (corresponding to higher modulation frequencies). It reflects these higher-frequency sideband components back into the transmitter or dissipates them in the tuning networks. This not only causes dull sound on the air (and defeats OPTIMOD TRIO-AM TRIO-AM's principal advantage: its ability to create a highly pre-emphasized signal without undesirable side effects), but it also wastes energy, can cause distortion, and can shorten the life of transmitter components.

Asymmetrical impedance is the common point impedance's not being symmetrical on either side of the carrier frequency when plotted on a Smith Chart. This problem can cause transmitter misbehavior and sideband asymmetry, resulting in on-air distortion in receivers with envelope detectors.

Neither problem is easily solved. Unless the radio station engineer is a knowledgeable antenna specialist, a reputable outside antenna consultant should be employed to design correction networks for the system.

Note that many antenna systems are perfectly adequate, particularly for ordinary mono analog transmission. However, if the transmitter sounds significantly brighter and/or cleaner into a dummy load than it does into your antenna, the antenna system should be evaluated and corrected if necessary.

As noted above, if your circumstances or budget precludes correcting your antenna's bandwidth and/or symmetry, you will often get lower on-air distortion if you set OPTIMOD TRIO-AM TRIO-AM's low-pass filter to a lower frequency than the maximum permitted by the government. Because OPTIMOD TRIO-AM TRIO-AM's output bandwidth is easily adjustable in real time, it is very easy to experiment to see which bandwidth gives the best audio quality on an average AM radio, given the quality of your transmitter and antenna.

Using Lossy Data Reduction in the Studio

Many stations are now using lossy data reduction algorithms like MPEG-1 Layer 2 to increase the storage time of digital playback media. In addition, source material has often been passed 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 OPTIMOD TRIO-AM TRIO-AM's input.

All such algorithms operate by increasing the quantization noise in discrete frequency bands. If not psychoacoustically masked by the program material, this noise may be perceived as distortion, “gurgling,” or other interference. 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 into audibility. In addition, at least one other mechanism can cause the noise to become audible at the radio. OPTIMOD TRIO-AM TRIO-AM'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. In addition, 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.

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, and 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. So 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.

Transmission Levels

The transmission engineer is primarily concerned with the peak level of a program to prevent overloading or overmodulation of the transmission system. This peak overload level is defined differently, system to system.

In FM modulation, 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

The meters on the TRIO-AM show left/right input levels and both positive and negative output modulation.

Input metering is stereo because the TRIO-AM will most often be fed by a stereo source even though the TRIO-AM processing is mono.

Left and right input level is shown on a VU-type scale, 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 corresponds to the absolute maximum peak level that the TRIO-AM can accept. If you are using the AES3 digital input, the maximum digital word at the input corresponds to the 0 dB point on the TRIO-AM's input meter.

Built-in Calibrated Line-up Tones

To facilitate matching the output level of the TRIO-AM to the transmission system that it is driving, the TRIO-AM contains an adjustable test tone oscillator that produces sine waves at TRIO-AM's (analog or digital) left and right outputs. The frequency and modulation level of the line-up tones can be adjusted from the front panel (as described in ***Error! Reference source not found.*** on page 3-7-20).

You can adjust the frequency and modulation level of the built-in line-up tone. You can use the front panel, the PC Control software, 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 pass line-up tones generated earlier in the system. This mode applies a DC servo, a highpass filter, a 15 kHz lowpass filter, a safety clipper, and transmitter equalization. The negative safety clipper threshold is set to 105% modulation. The positive threshold is determined by the active transmission preset (see *System Presets and Transmitter Equalization* on page1-2-16), as are the settings of the highpass filter and transmitter equalizer.

Monitoring

Modulation Monitors and Their RF Amplifiers

Many AM modulation monitors (particularly older ones) indicate dynamic modulation inaccurately even though they may accurately measure sine-wave modulation. This occurs producing overshoot and ringing. An incorrectly designed modulation monitor may indicate that modulation is as much as 3dB higher than it actually is.

When modulation monitors are used at locations distant from the transmitter, they are driven from highly selective RF amplifiers. These sometimes suffer from similar problems. They can overshoot and ring if the passband filters are too sharp, causing the monitor to falsely indicate high modulation.

If your modulation monitor does not agree with an oscilloscope monitoring the RF envelope at the common point, do not assume that the monitor is indicating fast peaks that your eye cannot see. A probable cause of the disparity is overshoot in the modulation monitor or its RF amplifier. If you observe this problem, we recommend that you assume that what you see on the oscilloscope is correct; oscilloscopes are designed to display pulse waveforms accurately. (Make sure the oscilloscope's input is set for DC coupling.)

Note also that modulation percentages will vary depending on where in the transmission system the RF sample is taken. Depending on the location observed, actual modulation can be either lower or higher than modulation observed at the common point. What is crucial is whether the carrier is actually pinched off at the final amplifier because this carrier pinch-off is what causes splatter. On the other hand, if the carrier appears is suppressed because of a particular choice of monitoring point within the system, negative peaks will fold around zero instead of cutting off. This causes no problem with out-of-band radiation, and far-field radiation is likely to show normal AM modulation envelopes. We therefore recommend that you use an RF sample from the final amplifier.

Monitoring on Loudspeakers and Headphones

Monitor Rolloff Filter

The output of a loudspeaker fed from the modulation monitor typically sounds shrill and strident because, unlike virtually all real AM radios, the modulation monitor and loudspeaker have a flat response. Rolloff filtering can be used to supply monitors with audio that more closely resembles that heard over a typical receiver.

If a different tonal balance is desired for off-the-air monitoring, install a simple program equalizer after the Monitor Rolloff Filter and adjust the 5 kHz region to taste.

Why the North American NRSC Standard?

Over the years, as the North American air waves have become more crowded, interference from first and second adjacent stations has become more and more of a problem. Receiver manufacturers responded by producing receivers with decreased audio bandwidth so that an adjacent station's modulation extremes would not be audible as interference. This cutting of the bandwidth had the effect of reducing the receiver's high-frequency response, but it was felt that lower fidelity would be less obnoxious than interference. As long ago as 1978, Orban proposed and implemented preemphasis and low-pass filtering for AM broadcast to provide brighter sound at the receiver while minimizing interference. This approach has become widely accepted. The NRSC-formalized standard is acceptable to all industry segments, and when implemented, can result in a vast improvement in AM radio.

AM Stereo Introduces a Preemphasis Dilemma

Certain AM receivers manufactured since 1984 for sale in North America, particularly those designed for domestic AM stereo reception, have a frequency response that is substantially wider than that of the typical mono AM receiver. The frequency response was widened largely to enhance the sales potential of AM stereo by presenting a dramatic, audible improvement in fidelity in the showroom. As these new receivers became more prevalent, broadcasters had to choose whether the station's preemphasis would be optimized for the new AM stereo receivers or for the existing conventional receivers that form the vast majority of the market. If the choice was for conventional receivers (which implies a relatively extreme preemphasis), the newer receivers might sound strident or exceptionally bright. If the choice favored the newer receivers (less preemphasis and probably less processing), the majority of receivers would be deprived of much high-end energy and would sound both quieter and duller.

NRSC Standard Preemphasis and Low-pass Filtering

In response to this dilemma, the National Radio Systems Committee (NRSC) undertook the difficult task of defining a voluntary recommended preemphasis curve for AM radio that would be acceptable to broadcasters (who want the highest quality sound on the majority of their listeners' radios) and to receiver manufacturers (who are primarily concerned with interference from first- and second-adjacent stations).

After a year of deliberation, a modified 75-microsecond preemphasis/deemphasis standard was approved (See ***Error! Reference source not found.***). This provides a moderate amount of improvement for existing narrowband radios while optimizing the sound of wideband radios. Most importantly, it generates substantially less first-adjacent interference than do steeper preemphasis curves. The second part of the NRSC standard calls for a sharp upper limit of 10 kHz (at –15dB) for the audio presented to the transmitter. (See *Figure* .)

OPTIMOD TRIO-AM TRIO-AM's NRSC low-pass setting is essentially flat to 9.5 kHz and substantially exceeds the NRSC standards above that frequency. This essentially eliminates interference to second and higher adjacencies. While some have protested that this is inadequate and that 15 kHz audio should be permitted, the unfortunate fact is that interference-free 15 kHz audio could only be achieved by a complete re-allocation of the AM band.

On April 27, 1989, The FCC (U.S.A.) released a Report and Order that amended section 73.44 of the FCC Rules by requiring all U.S. AM stations to comply with the occupied bandwidth specifications of the NRSC-2 standard by June 30, 1990. The NRSC-2 standard is an RF mask that was derived from the NRSC-1 audio standard. The purpose of the NRSC-2 RF mask is to provide a transmitted RF occupied bandwidth standard that any station with a properly operating transmitter will meet if NRSC-1 audio processing is used prior to the transmitter and if the station is not overmodulating.

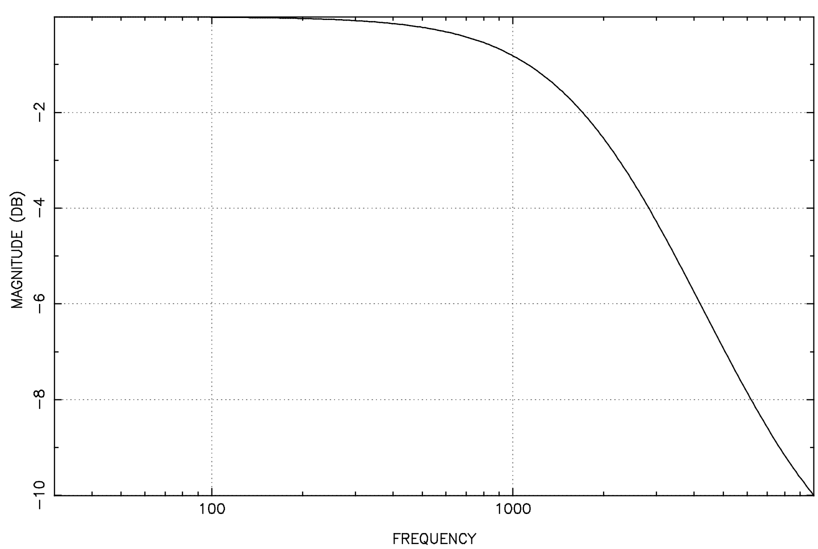


Figure 2-3: NRSC Modified 75 µs Deemphasis

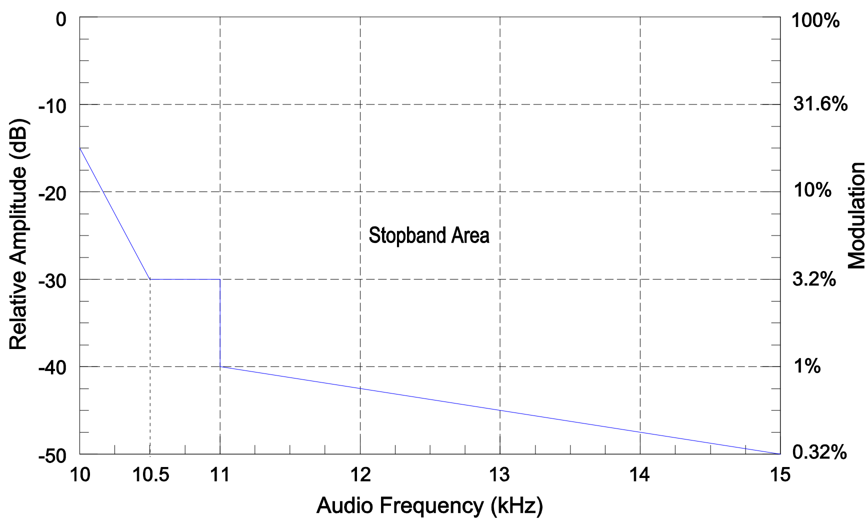


Figure 2-4: NRSC Lowpass Filter

OPTIMOD TRIO-AM TRIO-AM complies fully with the NRSC-1 standard when the 9.5 kHz NRSC low-pass filter is in use and the HF Curve control is set to NRSC.

Unfortunately, at this writing, the trend towards wider band receivers has reversed and most receivers are no wider than they were in the 1970s. For this reason, many engineers feel that using a third-order equalizer with 10 dB of ultimate boost provides a more intelligible sound on the average radio than does the NRSC curve. The TRIO's HF shelving equalizer can provide such a boost.

When a station is transmitting with 5 kHz audio bandwidth, the TRIO's 5 kHz lowpass filter can produce audible ringing that some find objectionable. Using the TRIO's bell-shaped HF parametric EQ tuned to 3 kHz can reduce the effects of this ringing by reducing the boost at 5 kHz by comparison to the TRIO's HF shelving EQ, which maintains full boost all the way to 5 kHz. Additionally, you can use the LPF Shape control to trade off brightness against ringing.

Introduction to the OPTIMOD TRIO-AM TRIO-AM and 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 OPTIMOD TRIO-AM TRIO-AM 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 TRIO-AM 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.

Receiver high frequency rolloff

A typical receiver’s severe HF rolloff reduces the headroom available at high frequencies and makes it difficult to achieve a bright sound. This is because bright sound requires considerable high frequency power to appear at the output of the receiver, thus requiring a very large amount of high frequency power to be transmitted so that a sufficient amount will survive the receiver’s rolloff.

To increase brightness and intelligibility at the receiver, the TRIO-AM’s NRSC preemphasis boosts the treble at 6dB/octave starting at 2.1 kHz. HF Curve settings from 0 to 10 produce more severe preemphasis than NRSC, boosting at 18dB/octave with 2 kHz up about 3 dB. Without very artful processing, this preemphasis will radically increase the level of the peaks and force you to decrease the average level proportionally. Orban's high frequency limiting and distortion-cancelling clipping systems greatly ease this trade-off, but cannot eliminate it. Therefore, you can only increase brightness by reducing average modulation (loudness) , unless you accept the increased distortion caused by driving the final clippers harder.

In 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 other two. Thanks to Orban's psychoacoustically-optimized designs, this is less true of Orban processors than of any others. Nevertheless, all intelligent processor designers must acknowledge and work within the laws of physics and psychoacoustics as they apply to these trade-offs.

OPTIMOD TRIO-AM TRIO-AM processing occurs in the following main stages.

- A gentle AGC that is ordinarily used to slowly ride gain, keeping long-term average drive levels into the following multiband compressor stage constant.

- A program equalizer. This starts with a three stage parametric equalizer that allows you to adjust bass, midrange, and high-frequency equalization. There are three fully parametric sections, each with non-interacting control over the amount of EQ (in dB), the bandwidth, and the center frequency. They are used to color the audio to achieve a “signature sound” for the station.

- The program equalizer also contains a high frequency shelving section we call the “receiver equalizer.” While the parametric equalizers are designed to produce program coloration as desired, the HF shelving section of the program equalizer is ordinarily used to pre-emphasize the signal to help overcome the high-frequency rolloff of typical AM radios. The shelving section can be operated as a fixed, first-order shelf to provide NRSC standard preemphasis or as a third-order semi-parametric shelf with adjustable gain and curve shape. In general, if you use a great deal of HF boost, you will have to turn down the Less-More control to avoid audible distortion.

- A five-band compressor with Orban's exclusive multiband distortion-cancelling clipper. This system embeds the clipper within the multiband crossover to permit the crossover to filter out clipping distortion products that would otherwise be audible. A feedforward sidechain provides further, highly selective cancellation of difference-frequency intermodulation distortion. The five-band compressor also incorporates a single-ended dynamic noise reduction system, which can be activated or defeated as desired.

- A safety clipper and overshoot compensator. These elements precisely control peak modulation without adding out-of-band frequencies, as a simple clipper would.

AM Processing: The Art of Compromise

Noise, interference, and narrow bandwidth inherently restrict AM audio quality. Because of this, purist goals (“the output should sound just like the input”) are irrelevant because receiver design makes them impossible to achieve. Instead, the goal of processing should be to deliver the highest subjective quality through this limited transmission channel to the listener's ear. This always requires substantial compression and limiting to ensure that the received signal will override the noise and interference over the maximum possible geographical area. It also requires high frequency boost to compensate for the high-frequency rolloff in all AM radios.

The TRIO-AM’s GEN PURPOSE MEDIUM factory preset at a LESS-MORE setting of 7 meets these requirements and provides a sound that is subjectively undistorted even on high-quality automobile radios. This is the default preset upon initial power-up of the TRIO-AM. You may continue using this preset or choose another preset as you deem appropriate.

You must also choose a setting of the system bandwidth control for the active System Preset (in System Setup). Depending on whether the bandwidth is 4.5 - 7 kHz or 7.5 - 9.5 kHz, the characteristics of any factory preset will change to complement the chosen bandwidth. For example, if the frequency is set at 7.5 kHz or above, the HF CURVE of the GEN PURPOSE MEDIUM preset will be automatically set to 10 and the HF GAIN will be set to +10dB. If the frequency is set at 7.0 kHz or below (as appropriate for areas such as Europe and Asia), the HF CURVE will be set to 0 and the HF GAIN will be set to +17dB. This matches OPTIMOD TRIO-AM TRIO-AM's HF preemphasis to the bandwidth of the radios that will most likely be in use in a given part of the world.

If the amount of transmitter power available is limited and you wish to cover the widest possible area, you may choose to process harder (by advancing the LESS-MORE control at the cost of some audible distortion and increased compression). You may also wish to reduce the amount of high frequency receiver equalization and/or decrease the audio bandwidth of the processing (by adjusting the system low-pass filter) because you will discover that you can achieve a louder sound with the same amount of distortion if you do this.

You will find out that in any setup there is *a* direct trade-off between loudness, brightness, and distortion. You can improve any single parameter, but only at the expense of one or both of the other two. This is true of any processor, not just OPTIMOD TRIO-AM TRIO-AM. Perhaps the most difficult part of adjusting a processor is determining the best trade-off for a given situation. If most of your listeners are located where your signal is strong, it is wiser to give up ultimate loudness to achieve brightness and low distortion. A listener can compensate for loudness by simply adjusting the volume control. But there is *nothing* the listener can do to make a dirty signal sound clean again, or to undo the effects of excessive high-frequency limiting.

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. On the other hand, if many listeners receive a weak signal or one that is frequently contaminated by interference, then processing harder to achieve maximum loudness, uniformity, and average modulation will let the station be heard more easily. You may therefore wish to process quite differently during the day than at night, when skywave interference is often a problem. OPTIMOD TRIO-AM TRIO-AM's programmable presets make this easy.

If women form a significant portion of the station's audience, bear in mind that women are more sensitive to distortion and listening fatigue than men are. In any format requiring long-term listening to achieve market share, great care should be taken not to alienate women by excessive stridency, harshness, or distortion.

AM radio has been losing its market share to FM in many countries because the public perceives that AM has lower sound quality. While this is inevitably true (except in the automobile, where multipath often degrades FM reception below “entertainment quality”), the damage can be minimized by processing the audio to make the best of the limitations of the AM channel and to avoid processing artifacts. OPTIMOD TRIO-AM TRIO-AM is uniquely effective in optimizing these trade-offs, and the discussion below tells you in more detail how to do this.

Shortwave/HF Processing

The goals for HF broadcasters are likely to be quite different than they would be in MW, LW, or FM broadcast. Listeners to HF broadcasts are often highly motivated and will continue to listen even when the signal is severely degraded by poor propagation conditions or by interference that would almost certainly cause the average LW, MW, or FM listener to tune to another station.

In LW and MW, the audio processor set-up controls are usually used to match the processor's "sound" to a certain type of music or talk programming. HF is different. In HF, the audio processor is usually adjusted to provide a sound at the receiver that is as esthetically satisfying as possible, *given the probable signal quality at the receiver*. The broadcasting organization usually does not have the luxury of making fine adjustments to match different types of program material, because such fine adjustments will almost certainly be masked by the variability of the propagation and interference experienced by the listener. This fact considerably simplifies the adjustment procedure.

We have tuned the TRIO-AM's "HF" presets with these compromises in mind. There is a general-purpose preset and a preset tuned to optimize voice intelligibility. We believe that further subtleties are inappropriate for the medium.

Working Together

Best results will be achieved if Engineering, Programming, and Management go out of their way to *communicate* and *cooperate* with each other. It is important that Engineering understands well the sound that Programming desires, and that Management fully understands the trade-offs involved in optimizing certain parameters (such as loudness and coverage) at the cost of others (such as brightness or distortion).

Fundamental Requirements: High-Quality Source Material and Accurate Monitoring

Very clean audio can be processed harder without producing objectionable distortion. If the source material is even slightly distorted, OPTIMOD TRIO-AM TRIO-AM can greatly exaggerate this distortion, particularly if a large amount of gain reduction is used. Potential causes for distortion are poor-quality source material, including the effects of the station's playback machines, electronics, and studio-transmitter link, as well as excessive clipping settings in the OPTIMOD TRIO-AM TRIO-AM processing. See *Maintaining Audio Quality in the Broadcast Facility* (an Orban publication downloadable from ftp.orban.com) for a discussion of how to improve source quality.

A high-quality monitor system is essential. To modify your air sound effectively, you must be able to *hear* the results of your adjustments. *Maintaining Audio Quality in the Broadcast Facility* also contains a detailed discussion of how to efficiently create an accurate monitoring environment.

Reference Radios for the Processing

However, do not rely on your monitor alone for subjectively evaluating your air sound. It is a good idea to develop a set of "reference radios" with which you are familiar and which are similar to those used by a majority of your audience. Too often, just *one* radio (typically the Program Director or General Manager's car radio) is used to evaluate air sound. Unless all of your listeners happen to have the same radio, this approach will not give an accurate indication of what your audience is hearing.

Based on their high-frequency response, AM radios can be divided into three groups:

Group 1: Wideband AM radios, typically with response that approximately follows the recommended NRSC "modified 75µs" deemphasis to 5 kHz or above. There are very few such radios available.

Group 2: Radios with a response down 3dB at approximately 2 kHz, with a gentle rolloff above that frequency. Because the rolloff is gentle, preemphasis can be used to brighten the sound.

Group 3: Radios with a response down 3dB at approximately 2 kHz, with a very steep rolloff above that frequency. The steepness of the rolloff eliminates the possibility of improving the audio through preemphasis. In our opinion, these radios must be written off as producing hopelessly bad sound. Very few people would enjoy listening to music on these radios , although they could be used for listening to talk programs, or for repelling pigeons and muggers.

The vast majority of present-day radios are in the second and third categories. In all three types of radio, bass performance is completely unpredictable from model to model. The best-sounding “Group 1” AM receiver we know of is the Sony SRF-A100 AM stereo radio (now discontinued), which can be switched between wideband and narrowband operation. Use headphones, or drive an external amplifier and speaker with the Sony's headphone output (its own tiny speakers cannot be used for reference purposes). A representative good-sounding wideband mono radio is the General Electric Superadio. In “Group 2,” we are fond of the Radio Shack MTA- series of small table radios. The last time we looked, the model number was up to MTA-17, but these numbers are updated periodically. Moreover, there is no guarantee that Radio Shack will continue to sell this series.

Be aware that many radios produce excessive distortion all by themselves, especially if they are located near the transmitter. If the station monitor (driven through OPTIMOD TRIO-AM TRIO-AM's monitor rolloff filter) sounds clean but your radio audio is distorted, don't trust the radio! If the General Manager's auto radio sounds distorted, he or she should not jump to the conclusion that there is something wrong with the station or with the engineer's ears.

Modulation Monitors

Many modulation monitors and RF amplifiers indicate higher modulation than the transmitter is actually producing. This forces the engineer to reduce transmitter modulation unnecessarily, which can cost you up to 3dB of loudness! It is *very important* to be sure that your modulation monitor is accurately calibrated and that it does not exhibit overshoot on program material. Several newer monitors are designed for accurate pulse response without overshoot. Any of these monitors will enable you to obtain the highest loudness achievable from your transmitter and antenna system. If the monitor is used remotely, be sure that the RF amp doesn't overshoot. Overshoots in RF amps have been observed to be as high as 3dB.

Monitor readings should be compared with an oscilloscope observing the modulated RF envelope. If the monitor indicates 100% negative peaks when the oscilloscope reveals no carrier pinch-off, suspect inaccuracy in the monitor.

More About Audio Processing

Psychoacoustic factors were considered carefully during the design and construction of OPTIMOD TRIO-AM TRIO-AM. The resulting audio processor is easy to use (the Less-More control greatly simplifies setup) and produces a sound that is remarkably free from unwanted processing artifacts.

Although the controls on OPTIMOD TRIO-AM TRIO-AM provide the flexibility you need to customize your station's sound, proper adjustment of these controls consists of balancing the trade-offs between loudness, density, brightness, and audible distortion. In programming the Less-More curves, we have made it easy for you to make this trade-off. As you advance the Less-More control for a given factory preset, the sound gets louder but distortion increases. However, for each setting of the Less-More control, other processing parameters are automatically adjusted to give you the lowest possible distortion for the amount of loudness you are getting.

If you want to go beyond Less-More and into the Full Control and Expert Modify adjustments, you should carefully read and understand the following section. It provides the information you need to adjust OPTIMOD TRIO-AM TRIO-AM controls to suit your format, taste, and competitive situation.

Judging Loudness

Apparent loudness in an analog AM channel will vary with the frequency response of the radio and with the accuracy with which the radio is tuned. Narrowband radios will usually get very much louder if tuned off center while a highly equalized signal is being received. Loudness is a very complex psychoacoustic phenomenon. One station cannot be judged louder than another unless it is *consistently* louder on many different receivers with many different types of program material. Because a wideband radio reproduces more of the frequency range in which the highly-equalized signal concentrates its energy (and to which the ear is most sensitive), a highly equalized signal may sound quieter than an unequalized signal on a narrowband radio, while the reverse is true on a wideband radio.

Reverberation

In the distant past, the addition of artificial reverberation was touted as an easy method of achieving greater loudness in AM broadcasting. Given the limitations of the audio processing equipment of that time, this was true: reverberation increased the signal density and average modulation without the pumping or other side effects that heavy limiting would cause if equivalent density were to be achieved by compression or limiting alone. However, because reverberation “smeared” the sound, it exacted a price of decreased definition and intelligibility in many instances.

Because OPTIMOD TRIO-AM TRIO-AM is to augment density without producing audible artifacts, reverberation is neither necessary nor desirable for achieving high loudness and density. Moreover, OPTIMOD TRIO-AM TRIO-AM actually increases definition and intelligibility.

If you still wish to use reverb to achieve a nostalgic sound in an oldies format, we recommend using it in extreme moderation and applying it to the signal before it reaches OPTIMOD TRIO-AM TRIO-AM. OPTIMOD TRIO-AM TRIO-AM will effectively increase the amount of reverb by increasing the level of the reverb decay and keeping the reverb before OPTIMOD TRIO-AM TRIO-AM will allow OPTIMOD TRIO-AM TRIO-AM to control peak modulation accurately.

Customizing the TRIO-AM’s Sound

The subjective setup controls on the TRIO-AM give you the flexibility to customize your station’s sound. Nevertheless, as with any audio processing system, proper adjustment of these controls consists of balancing the trade-offs between loudness, density, and audible distortion. The following pages provide the information you need to adjust the TRIO-AM controls to suit your format, taste, and competitive situation.

When you start with one of our Factory Presets, there are two levels of subjective adjustment available to you to let you customize the Factory Preset to your requirements: Basic Control and Full Control. A third level, Advanced Control, is accessible *only from the TRIO-AM’s PC Remote software*.

Spend some time listening critically to your on-air sound. Listen to a wide range of program material typical of your format. Listen on several types of car, table, and portable radios, not just your studio monitors.

Connecting to The OPTIMOD TRIO-AM TRIO-AM via Web Interface

The default address for the OPTIMOD TRIO-AM TRIO-AM 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 OPTIMOD TRIO-AM TRIO-AM when connecting to set up a proper network.

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

USERNAME: ADMIN (all caps)

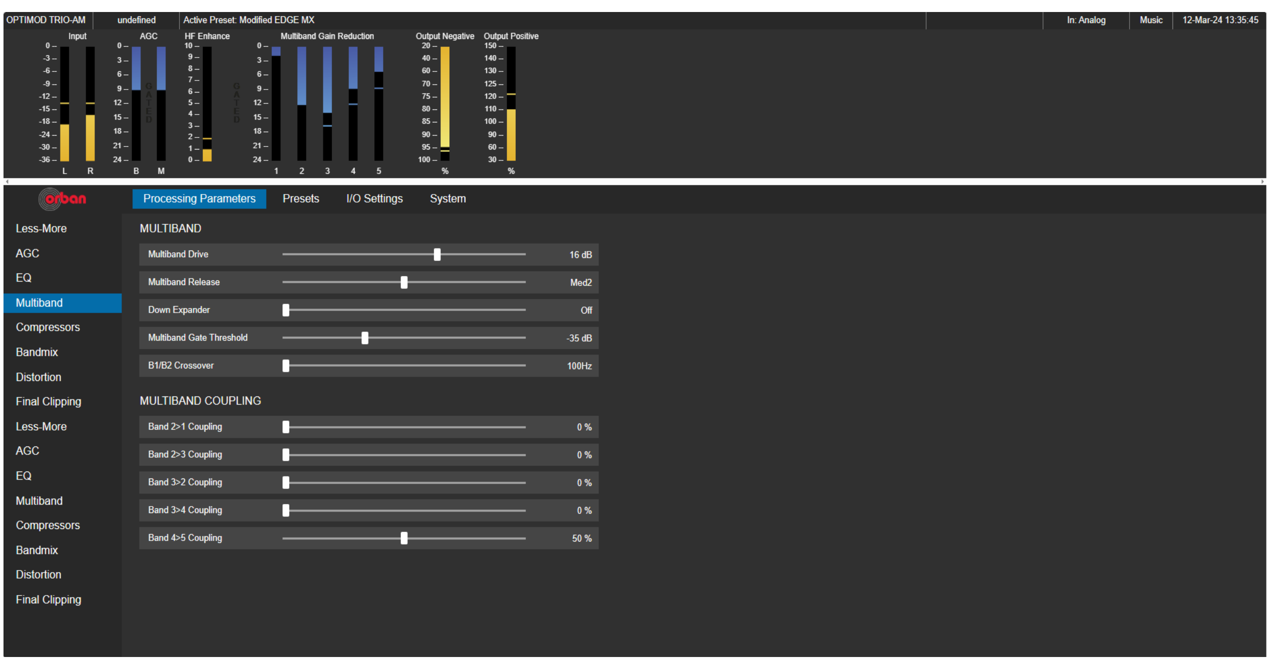
PASSWORD: 1234

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 OPTIMOD TRIO-AM TRIO-AM.
- If configured properly you will see the browser connect along with metering and control options.



Customizing the OPTIMOD TRIO-AM TRIO-AM's Sound

Gain Reduction Metering

Unlike the metering on some processors, when any the OPTIMOD TRIO-AM TRIO-AM 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 TRIO-AM TRIO_AM has been set up for a sane amount of gain reduction under ordinary program conditions.

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 OPTIMOD TRIO-AM TRIO-AM 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 OPTIMOD TRIO-AM TRIO-AM 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 OPTIMOD TRIO-AM TRIO-AM will give you an opportunity to choose a new name for the preset you are saving.

You can save user presets from the OPTIMOD TRIO-AM TRIO-AM HTML5 application. Please note that when you save presets from the HTML5 application, you save them in the OPTIMOD TRIO-AM TRIO-AM's memory (as if you had saved them from the OPTIMOD TRIO-AM TRIO-AM'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 OPTIMOD TRIO-AM TRIO-AM

In the OPTIMOD TRIO-AM TRIO-AM, 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), OPTIMOD TRIO-AM TRIO-AM'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 OPTIMOD TRIO-AM TRIO-AM realizes its processing structures as a series of high-speed mathematical computations made by Digital Signal Processing (DSP) integrated circuit chips. In the OPTIMOD TRIO-AM TRIO-AM, both structures operate simultaneously so there is no delay in switching between them, which is done with a smooth cross-fade.

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 TRIO-AM'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 TRIO-AM's programming facilities.

It is OK to use unmodified factory presets on the air. These represent the best efforts of some very experienced on-air sound designers. We are sometimes asked about unpublished "programming secrets" for OPTIMOD TRIO-AMs. In fact, there are no "secrets" that we withhold from users. Our "secrets" are revealed in this manual and the presets embody all of our craft as processing experts. The presets are editable because other sound designers may have different preferences from ours, not because the presets are somehow mediocre or improvable by those with special, arcane knowledge that we withhold from most of our customers.

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 AM radios (not just on your studio monitors). Then, if you wish, customize your sound using the information that follows.

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

Important! If you are dissatisfied with the sound available from the factory presets, please understand that each named preset is actually 19 presets that can be accessed via the LESS-MORE control. Try using this 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 choose a preset other than the one named for the type of programming on-air if you believe this other preset has a more appropriate sound. Also, if you want to fine-tune the frequency balance of the programming, feel free to use Basic Control and make small changes to the Bass, Mid EQ, and HF EQ controls. Unlike some earlier Orban's processors, the TRIO-AM lets you make changes in EQ, AGC, 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.

Description of the Factory Presets

Presets with "HF" in their names are narrowband presets intended for international shortwave transmission where 4.5 kHz audio bandwidth and difficult propagation conditions are the norm. All other presets are intended for medium wave transmission.

GENERAL PURPOSE MEDIUM is the default factory preset. It is based on the Medium-Fast multiband release time and is adjusted to sound equally good on voice and music. It is most appropriate for listeners in strong signal areas because it does not bring up low-level material as much as presets based on the Fast multiband release time.

GENERAL PURPOSE HEAVY is based on the Fast multiband release time, and is designed to sound good on voice and music. Because it processes harder than the GENERAL PURPOSE MEDIUM preset it can be louder but does not sound as punchy or dynamic. It is a good choice when many listeners are subject to noise and interference and you want the highest possible loudness.

NEWS is based on the Fast multiband release time. Because of this, the unit adapts quickly to different program material, providing excellent source-to-source consistency. This "automatic equalization" action of the multiband compressor has been adjusted to produce less bass than in the GENERAL PURPOSE presets, and the gating threshold is set considerably higher. This maximizes voice intelligibility, including low-quality sources like telephone. The high gating threshold resists noise pumping even with noisy material.

NEWS + NR is identical to the News preset except that the Dynamic Noise Reduction.

FACTORY PROGRAMMING PRESETS		
Preset Names	**Source Preset**	**Normal Less-More**
GEN PURPOSE MEDIUM	GEN PURPOSE MEDIUM	7.0
GEN PURPOSE HEAVY	GEN PURPOSE HEAVY	7.0
NEWS	NEWS	7.0
NEWS+NR	NEWS+NR	7.0
SPORTS	SPORTS	7.0
FINE ARTS	FINE ARTS	7.0
MUSIC MEDIUM	MUSIC MEDIUM	7.0
MUSIC HEAVY	MUSIC HEAVY	7.0
GREGG	GREGG	7.0
PRESENCE	PRESENCE	9.0
HF GENERAL	HF GENERAL	7.0
HF VOICE	HF VOICE	7.0

Figure 2-5: Factory Programming Presets

Function is also activated, producing even more noise reduction on moderately noisy program material. However, the Dynamic Noise Reduction function can produce audible side effects that include noise pumping on very noisy material and a subtle loss of crispness on high-quality voice. So you should listen carefully to decide if it is preferable to News for your situation.

SPORTS is based on the Fast multiband release time. Compared to NEWS, the AGC is operated with a slower release time to avoid pumping up crowd noise as much as the News preset would. Yet the Fast multiband release time still provides excellent consistency, intelligibility, and loudness.

FINE ARTS is based on the Slow multiband release time. It is designed for classical and jazz programming where an open, unprocessed sound is more desirable than the last bit of loudness. Unlike the other factory presets, the FINE ARTS LESS-MORE curves are designed to produce more compression as they are advanced, but not to significantly increase clipping distortion. So setting LESS-MORE higher will mostly increase the level of quiet passages instead of increasing the loudness of loud passages in the source material.

MUSIC MEDIUM is based on the Medium-Slow multiband release time. It is designed for various adult-oriented music formats where an easy, relaxed sound is considered more important than the highest possible loudness.

MUSIC HEAVY is based on the Fast multiband release time. However, its tuning is very different than GEN PURPOSE HEAVY. It is tuned so that the AGC operates with a fast release time, doing most of the work in compressing the program. This gives more of a "wideband compression" sound than the other factory presets. Meanwhile, the multiband compressor is operated lightly with relatively little gain reduction so it acts more like a limiter than a compressor. Music Heavy is therefore an alternative to General Purpose Fast, providing a different "flavor" of processing. Either preset could be used to achieve a highly-processed sound with music programming.

The **GREGG** preset is designed for general-purpose voice/music programming, particularly on music-oriented formats. Although not the loudest TRIO-AM preset, it has a smooth, well-balanced quality that keeps audiences listening. We tuned it to sound very similar to the legendary Gregg Laboratories 2540 AM processor (designed by Orban's Vice President of New Product Development, Greg Ogonowski, in the 1980s), using a direct A/B comparison with the Gregg processor to ensure accuracy. This required setting the B1/B2 CROSSOVER to 200 Hz and setting other controls appropriately.

The **PRESENCE** preset, as its name suggests, emphasizes the spectrum around 3 kHz. It is a very loud preset, particularly useful for talk formats. Unlike the "HF" presets, it is set for full NRSC bandwidth and tuned for a moderate amount of bass.

As always, the actual audio cutoff frequency will depend on the setting of the bandwidth control in System Setup.

MW stations seeking to increase their coverage and to cut through co-channel interference are appropriate candidates for this preset, which renders speech crisp, loud, and highly intelligible.

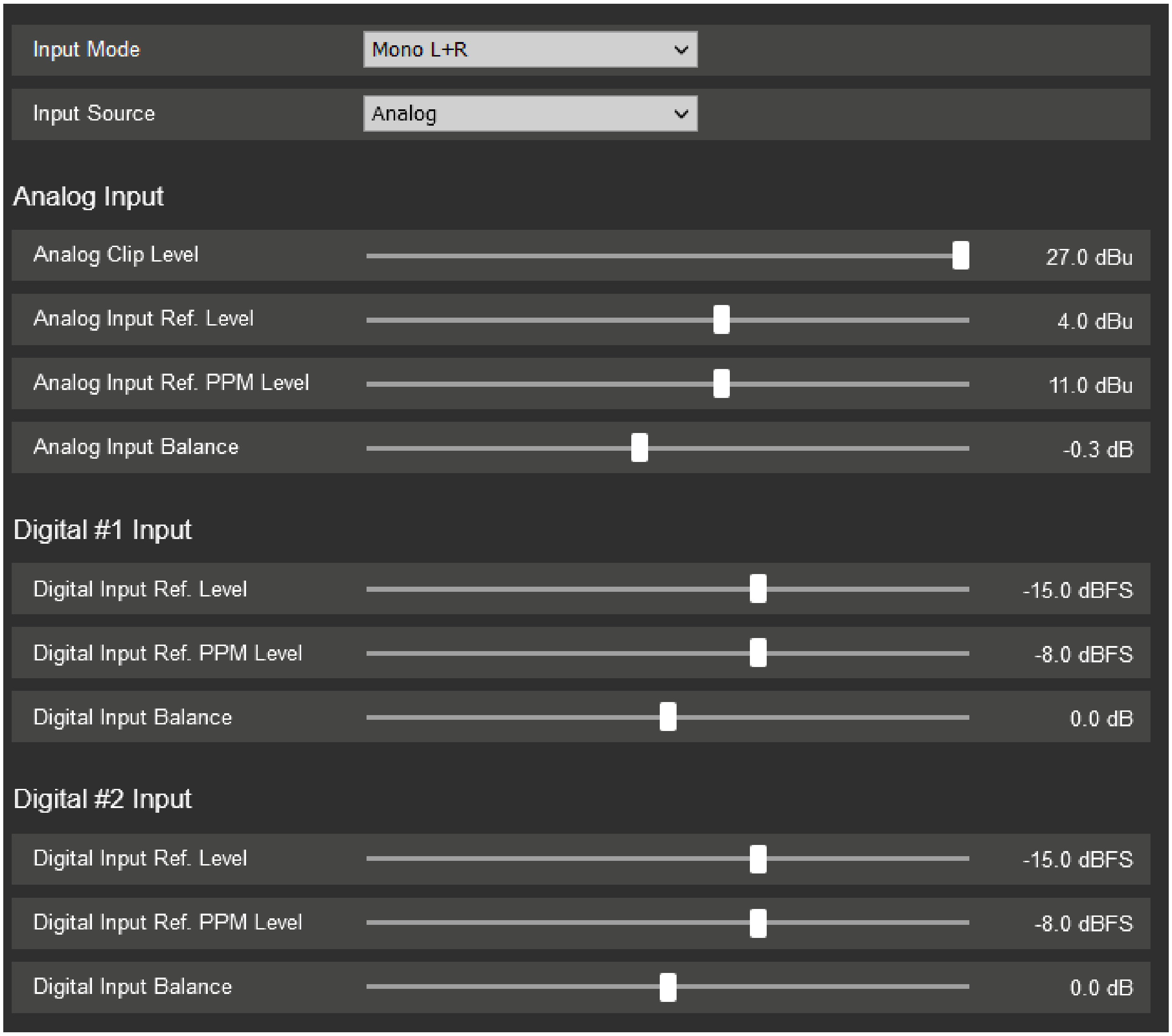
This preset is targeted for the typical narrowband MW radio and will sound shrill and unpleasant on wideband radios (of which there are very few in the market). If you feel that the preset has too much distortion, feel free to turn it down it with LESS-MORE to taste. The factory LESS-MORE setting is 9.0, so there is plenty of room to turn the preset down without seriously compromising loudness and coverage.

You can also reduce the midrange boost if you feel this is excessive. Part of the boost is implemented in the Equalization section and part is implemented by the B4 and B5 output band mix controls, which are found in Advanced Control.

HF GENERAL is a 4.5 kHz-bandwidth preset for international shortwave transmission. In recognition of the severe noise and interference problems often encountered in HF propagation, the HF GENERAL preset has been ``tuned" to emphasize loudness and intelligibility. By comparison to the medium-wave-oriented presets, HF GENERAL has a more ``forward" midrange balance and less bass. This is because bass costs modulation without contributing proportional intelligibility (it also can make intermodulation distortion worse during selective fading), and because a boosted midrange can most effectively cut through noise to provide intelligibility.

HF VOICE is a 4.5 kHz-bandwidth preset for international shortwave transmission. Compared to HF GENERAL, it emphasizes voice-range frequencies and has less bass. It is carefully crafted to maximize speech intelligibility in the presence of noise, interference, and jamming. If you need even more average modulation at the expense of distortion, turn up LESS-MORE as necessary.

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.

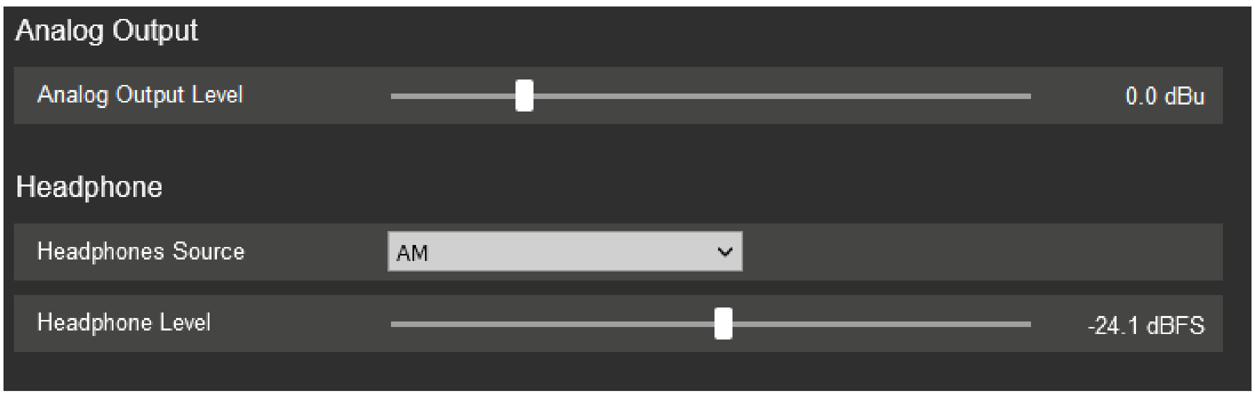
This step sets the center of the TRIO-FM's gain reduction range to the level to which your studio operators peak their program material on the studio meters. This assures that the TRIO-FM's processing presets will operate in their preferred range. You may adjust this level with a standard reference/line-up 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 MODE: Allows you to select L+R, Left Only or Right Only as your input source.

INPUT SOURCE: Allows you to select which input path will be the main input source. Dropping down the window will display options.

Analog Output



Setting the output to feed your AM transmitter requires navigation between two tabs. The ANALOG OUTPUT and the TRANSMISSION PRESET. You will need a known good modulation monitor or a transmitter that displays modulation on its front panel (such as Nautel or the BE AM-1, AM-5 series)

1) Adjust the ANALOG OUTPUT LEVEL to 0dBu.

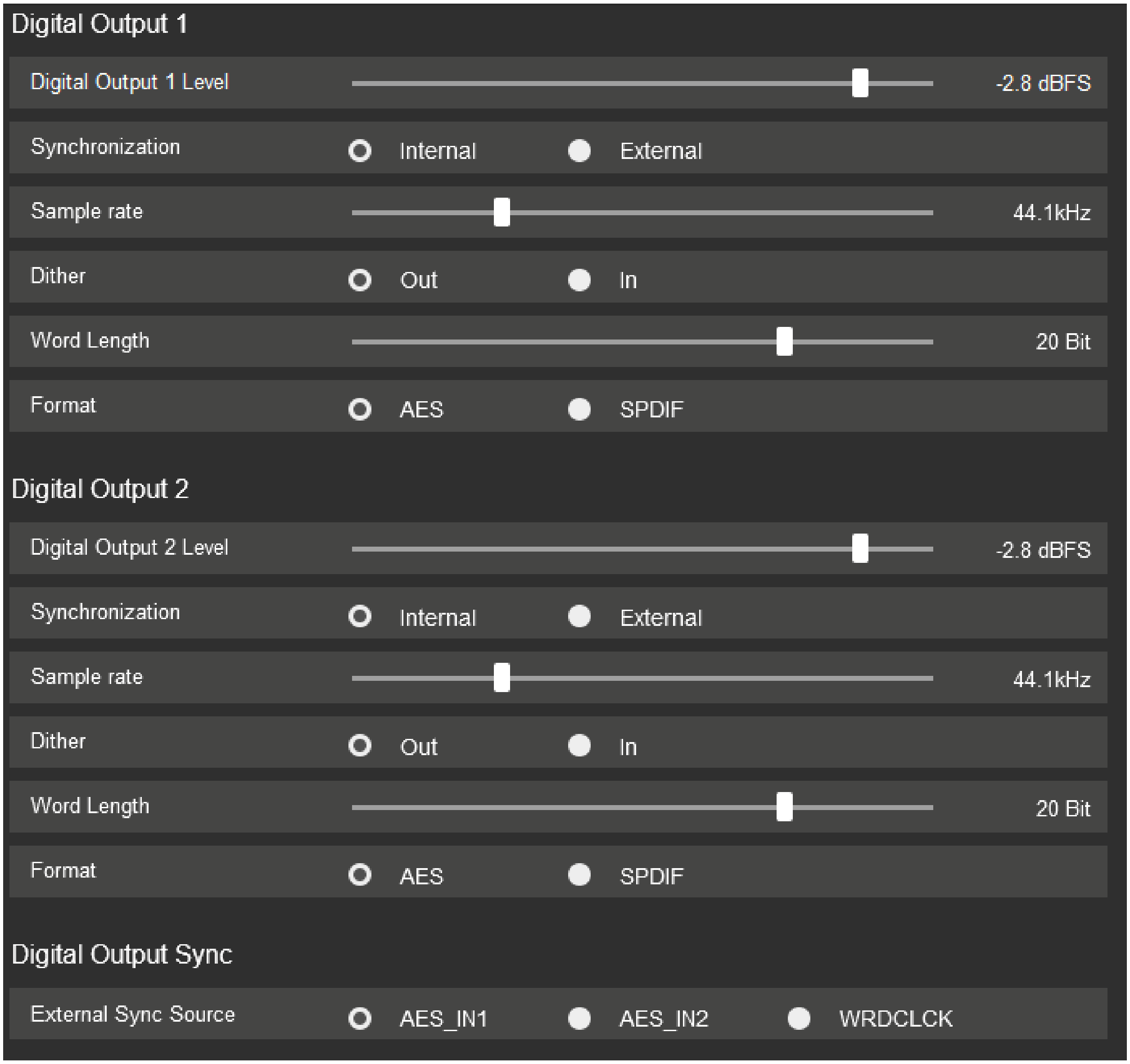
2) Navigate to TRANSMISSION PRESET (along the left column) and set the POSITIVE MODULATION to 100%.

3) Using TONE or normal programming material, adjust the ANALOG OUTPUT LEVEL so that the modulation monitor reads approximately 95% with positive and negative peaks.

4) Return to the TRANSMISSION PRESET and adjust the POSITIVE PEAK CONTROL to 120%

You should now have a satisfactory amount of loudness with your program material on the air.

Digital Outputs



Each digital output may be adjusted on its own. There are two digital outputs in the OPTIMOD TRIO-AM TRIO-AM. Each has controls for OUTPUT LEVEL, INTERNAL or EXTERNAL SYNC, SAMPLE RATE, DITHER, WORD LENGTH and FORMAT (AES or SPDIF). There is also an option to set the output sync to AES IN1, AES IN2 or WORDCLOCK.

As with the analog outputs, you will need to adjust the POSITIVE PEAK modulation in concert with the output under the TRANSMISSION PRESET tab

Transmission Presets

There are four transmitter presets for two transmitters. TX1/Day, TX1/Night, TX2/Day, TX2/Night.

- HIGH PASS FILTER: Sets the cutoff frequency for audio below the value chosen.
- LOW PASS FILTER: Sets the cutoff frequency for audio above the value chosen.
- LOW PASS FILTER SHAPE: Sets the steepness (dB/octave) of the filter applied.
- POSITIVE PEAK: Sets the positive peak level of the audio

While the physics of carrier pinch-off limit any AM modulation system to an absolute negative modulation limit of 100%, it is possible to modulate positive peaks as high as desired. In the United States, the FCC permits positive peaks of up to 125% modulation. Many countries have similar restrictions.

However, many transmitters cannot achieve such modulation without substantial distortion, if they can achieve it at all. The transmitter's power supply can sometimes be strengthened to correct this. Sometimes, RF drive capability to the final power amplifier must be increased.

Voice, by its nature, is substantially asymmetrical. Therefore, asymmetrical modulation was popular at one time in an attempt to increase the loudness of voice. Traditionally, this was achieved by preserving the natural asymmetry of the voice signal. An asymmetry detector reversed the polarity of the signal to maintain greater positive modulation. The peaks were then clipped to a level of -100%, +125%.

OPTIMOD TRIO-AM TRIO-AM takes a different approach: OPTIMOD TRIO-AM TRIO-AM's input conditioning filter contains a time dispersion circuit (phase scrambler) that makes asymmetrical input material, like voice, substantially symmetrical.

OPTIMOD TRIO-AM TRIO-AM permits symmetrical or asymmetrical operation of both the safety clipper and multiband distortion-canceling clipper. Asymmetrical clipping slightly increases loudness and brightness, and can produce dense positive peaks up to 125%. However, such asymmetrical processing by its very nature produces both odd and even-order harmonic and IM distortion. While even-order harmonic distortion may sound pleasingly bright, IM distortion of any order sounds nasty.

There is really nothing lost by not modulating asymmetrically: Listening tests easily demonstrate that modulating symmetrically, if time dispersion has been applied to the audio, produces a considerably louder and cleaner sound than does asymmetrical modulation that retains the natural asymmetry of its program material.

Some of the newer transmitters of the pulse-width modulation type have circuitry for holding the carrier shift constant with modulation. Since artificial asymmetry can introduce short-term DC components (corresponding to dynamic upward carrier shift), such carrier shift cancellation circuitry can become confused, resulting in further distortion.

System Presets and Transmitter Equalization

OPTIMOD TRIO-AM TRIO-AM's transmitter equalizer can cure linear problems caused by the transmitter or antenna system. However, the transmitter equalizer cannot cure nonlinear problems, particularly those caused by inadequate power supplies, modulation transformers, or reactors. If any of these components saturate or otherwise fail to perform under heavy power demands, no amount of small-signal equalization will solve their problems.

OPTIMOD TRIO-AM TRIO-AM was designed with the assumption that one audio processor would be devoted to no more than two transmitters, usually called main and standby (or main and alternate). Each transmitter might be required to change power at night or to drive a different antenna array. Only one transmitter is assumed to be on the air at a given time.

To drive two transmitters, OPTIMOD TRIO-AM TRIO-AM provides two mono analog outputs (called Analog Output 1 and Analog Output 2) and one AES3 digital output, which can alternatively be used to drive the main transmitter if it has a digital input.

OPTIMOD TRIO-AM TRIO-AM provides four system presets for its transmitter equalizer controls and certain other controls. Only one preset can be active at a given time; all three outputs receive the same transmitter equalization. This is consistent with the principle that only one transmitter will be on the air.

Transmitter equalizer controls in a given system preset include:

- LF Gain for the LF tilt equalizer [LF GAIN]
- LF Breakpoint Frequency for the LF tilt equalizer [LF FREQ]
- HF Shelf tuning [HF FREQ]
- HF Delay equalization [HF DELAY]
- System presets also contain the following controls:
- System Lowpass Filter Cutoff F requency [LOW PASS]
- System Lowpass Filter Cutoff Shape [LPF SHAPE]
- System Highpass Filter Cutoff Frequency [HIGH PASS]
- Positive Peak Threshold (Asymmetry) [POS PEAK]

For convenience and to describe their most common application, the four transmitter equalizer presets are labeled TX1/DAY, TX1/NIGHT, TX2/DAY, and TX2/NIGHT, although they can be applied in a completely general way to the requirements of your transmission facility.

For example, in countries observing NRSC standards you might want to transmit the full 9.5 kHz bandwidth during the day, and, in cooperation with other stations on first-adjacent channels, reduce audio bandwidth to 5 kHz at night. This will eliminate any skywave-induced monkey-chatter interference between first-adjacent channels. Alternatively, your nighttime directional antenna array might have poor VSWR performance at high modulating frequencies, so you might find that your transmitter works better and produces less distortion if you limit the audio bandwidth to those frequencies where the antenna is well behaved. Further, if you operate a talk format during certain parts of the day, you will probably find that you can operate the processing for a louder on-air sound if you restrict the transmitted bandwidth below the maximum permitted by government regulation. (Bear in mind that most AM radios have an audio bandwidth of 2.5 to 3 kHz and changing transmission bandwidth from 5 kHz to 9.5 kHz will produce virtually no audible difference on these radios.)

Antenna System

AM antenna systems, whether directional or non-directional, frequently exhibit inadequate bandwidth or asymmetrical impedance. Often, a system will exhibit both problems simultaneously.

An antenna with inadequate bandwidth couples RF energy into space with progressively less efficiency at higher sideband frequencies (corresponding to higher modulation frequencies). It reflects these higher-frequency sideband components back into the transmitter or dissipates them in the tuning networks. This not only causes dull sound on the air (and defeats OPTIMOD TRIO-AM TRIO-AM's principal advantage: its ability to create a highly pre-emphasized signal without undesirable side effects), but it also wastes energy, can cause distortion, and can shorten the life of transmitter components.

Asymmetrical impedance is the common point impedance's not being symmetrical on either side of the carrier frequency when plotted on a Smith Chart. This problem can cause transmitter misbehavior and sideband asymmetry, resulting in on-air distortion in receivers with envelope detectors.

Both of these limitations can cause severe problems in AM stereo and even worse ones in HD-AM installations.

Neither problem is easily solved. Unless the radio station engineer is a knowledgeable antenna specialist, a reputable outside antenna consultant should be employed to design correction networks for the system.

Note that many antenna systems are perfectly adequate, particularly for ordinary mono analog transmission. However, if the transmitter sounds significantly brighter and/or cleaner into a dummy load than it does into your antenna, the antenna system should be evaluated and corrected if necessary.

As noted above, if your circumstances or budget precludes correcting your antenna's bandwidth and/or symmetry, you will often get lower on-air distortion if you set OPTIMOD TRIO-AM TRIO-AM's low-pass filter to a lower frequency than the maximum permitted by the government. Because OPTIMOD TRIO-AM TRIO-AM's output bandwidth is easily adjustable in real time, it is very easy to experiment to see which bandwidth gives the best audio quality on an average AM radio, given the quality of your transmitter and antenna.

Test

The Test Modes screen allows you to switch between Operate, Bypass, and Sine, and Square. When you switch to Bypass or either tone mode (sine wave or square wave), the preset you have on air is saved and will be restored when you switch back to Operate.

The sine and square frequencies are adjustable. The triangle frequency is fixed at 100 Hz.

Error! Reference source not found. (below) shows the facilities available.

BYPASS PROTECT sets the threshold of a clipper in the bypass signal chain. It allows you to protect a transmitter in case excessive signal level is applied to the input of the TRIO-AM while in Bypass mode.

BYPASS HPF (Highpass Filter) allows you to protect your transmitter by keeping the TRIO-AM’s highpass filter to the bypass signal chain. The active System Preset determines the filter’s cutoff frequency.

Setup: Test				
Parameter Labels	Units	Default	Range (CCW to CW)	Step
MODE	---	Operate	Operate, Bypass, Sine, Square, Triangle	---
BYPASS GAIN	dB	0.0	–18 … +25	1
BYPASS PROTECT	%	100	50, 60, 70, 80, 90, 100, 105	
BYPASS HPF	---	Out	Out, In	
SINE 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, 15000	LOG
SQUARE FREQ	Hz	400	16, 20, 25, 31.5, 40, 50, 63, 80, 100, 125, 160, 200, 250, 315, 400, 500, 630, 800, 1000	LOG
SINE/TRINGL MOD	%	100	0 … 121	1
SQUARE MOD	%	30	0 … 50	1

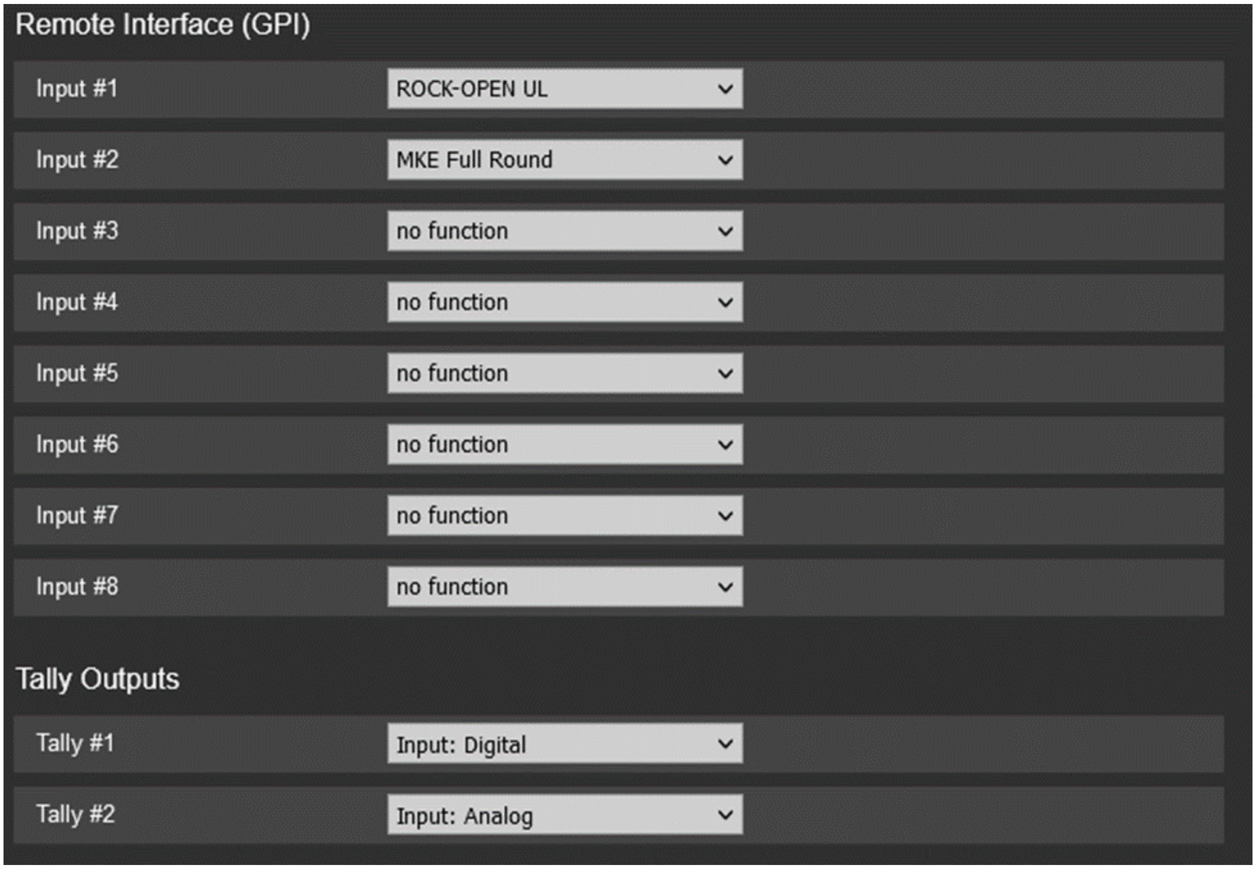
Figure 2-6: Test Modes

Utility

- **EXTERNAL AGC IN USE:** Set to YES if you are using an AGC at the studio to control STL levels.
- **STATION ID:** Sets the identification of the OPTIMOD TRIO-AM-TRIO you are connected to.

Remote Interface

The GPI/GPO closures on the DB cable on the rear of the TRIO-FM 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:** Lets you select the backup audio source if the main audio source goes silent and/or unlocks (AES).
- **SILENCE THRESHOLD:** Sets the level at which the OPTIMOD TRIO-AM-TRIO recognizes audio.
- **SILENCE DELAY:** Sets how long the OPTIMOD TRIO-AM-TRIO will wait during silence before it switches audio sources.

Equalizer Controls

Error! Reference source not found. summarizes the equalization controls available for the TRIO-AM.

“Advanced” controls are accessible only from TRIO-AM PC Remote software.

Any equalization that you set will be automatically stored in any User Preset that you create and save. For example, you can use a User Preset to combine an unmodified Factory Programming Preset with your custom equalization. Of course, you can also modify the Factory Preset (with Basic Control, Full Control, or Advanced Control) before you create your User Preset.

EQUALIZER CONTROLS			
Group	**Basic / Full Control Name**	**Advanced Name**	**Range**
Low	LF FREQ	Low Frequency	20 ... 500 Hz
	LF GAIN	Low Gain	–10.0 … +10.0 dB
	LF WIDTH	Low Width	0.8 ... 4 octaves
Mid	MID FREQ	Mid Frequency	250 ... 6000 Hz
	MID GAIN	Mid Gain	–10.0 … +10.0 dB
	MID WIDTH	Mid Width	0.8 ... 4 octaves
High	HIGH FREQ	High Frequency	1.0 … 15.0 kHz
	HIGH GAIN	High Gain	–10.0 … +10.0 dB
	HIGH WIDTH	High Width	0.8 ... 4 octaves
HF Enhancer	HF ENHANCE	High Frequency Enhancer	0 … 15
HF Gain	HF GAIN	High Frequency Shelf Gain	0 … 22 dB
HF Curve	HF CURVE	High Frequency Shelf Curve	0 … 10, NRSC
DJ Bass	DJ BASS	DJ Bass Boost	Off, On dB
	B1 DRIVE	Band 1 Input Drive	–3.0 … +3.0
	B2 DRIVE	Band 2 Input Drive	–3.0 … +3.0
	B3 DRIVE	Band 3 Input Drive	–3.0 … +3.0
	B4 DRIVE	Band 4 Input Drive	–3.0 … +3.0
	B5 DRIVE	Band 5 Input Drive	–3.0 … +3.0
System Filters	LOW PASS	Lowpass	4.5, 5.0, 5.5, 6.0, 6.5, 7.0, 7.5, 8.0, 8.5 ,9.0. 9.5(NRSC) kHz
	LPF SHAPE	LPF Shape	–0.1, –3.0, –6.0 dB
	HIGH PASS	Highpass	50 ,60, 70, 80, 90, 100 Hz

Table 2-7: Equalization Controls

In general, you should be conservative when equalizing modern, well-recorded program material. This is particularly true with general-purpose AM programming.

Low Frequency Parametric Equalizer is a specially designed parametric equalizer whose boost and cut curves closely emulate those of a classic Orban analog parametric equalizer with conventional bell-shaped curves (within ±0.15 dB worst-case). This provides warm, smooth, “analog-sounding” equalization.

LF FREQ determines the center frequency of the equalization, in Hertz. Range is 20-500Hz.

LF GAIN determines the amount of peak boost or cut (in dB) over a ±10 dB range.

LF WIDTH determines the bandwidth of the equalization, in octaves. The range is 0.8-4.0 octaves. If you are unfamiliar with using a parametric equalizer, 1.5 octaves is a good starting point. These curves are relatively broad because they are designed to provide overall tonal coloration, rather than to notch out small areas of the spectrum.

Although a certain amount of low-frequency boost must be used along with the high frequency boost in order to obtain a balanced sound on analog AM radios for MW, *do so conservatively!* Use the bassiest auto radios (all of which usually have a peaky mid-bass when you listen through the standard dashboard speaker) as a “worst case” reference. Do not boost the bass so much that your reference radio becomes muddy or boomy. With correct bass boost, your table radio will have only moderate bass and your pocket radio will sound thin and tinny.

For example, a 6dB boost corresponds to a 400% increase in power! More than 6dB of bass boost will strain many transmitters, unnecessarily increasing power supply bounce and IM distortion problems. (The bass boost is further limited dynamically in the multiband clipper , see immediately below.) Excessive bass boost will also cause many dashboard speakers to sound unacceptably muddy.

Use of a narrow bandwidth, a low boost frequency (like 65 Hz), and a relatively large boost can produce a very punchy sound in a car, or on a radio with significant bass response. It can also cost you loudness (bass frequencies take *lots* of modulation without giving you proportionate perceived loudness) and can cause thin sound on radios with only moderate bass response. A smaller amount of boost can often produce a better compromise.

In **HF broadcast**, perhaps the most difficult of all processing tradeoffs is choosing bass equalization. This is why the TRIO-AM’s bass equalizer can cut as well as boost.

When propagation conditions are good and the signal strength is high, a certain amount of bass boost (perhaps +3dB) provides the most pleasing sound. However, robust bass can easily induce intermodulation distortion in the clippers, so the amount of clipping must be reduced to provide acceptable distortion performance. In turn, this may compromise loudness by up to 3dB — the equivalent of cutting transmitter power in half!

Bass boost tends to reduce the life of power tubes in most high-powered transmitters. It will often induce intermodulation distortion in envelope detectors under selective fading, when detection becomes markedly nonlinear because of sideband asymmetry. In short, the arguments for bass cut are usually more persuasive than those for bass boost. Yet if an HF broadcasting organization seeks the highest possible subjective quality regardless of transmitter operating cost and feels that it usually delivers a strong RF signal, free from selective fading, to its listeners, then such an organization may still wish to boost bass slightly.

It is important to understand that the effect of the bass equalizer is relatively subtle, because bass balances are also affected by the action of the 150Hz and 420Hz bands of the multiband limiter and multiband distortion-canceling clipper. These bands will make bass balances more uniform (partially ``fighting'' bass-balance changes made with the bass equalizer) by increasing bass in program material that is thin-sounding, and by limiting heavy bass to a user-settable threshold below 100% modulation to prevent disturbing intermodulation between bass and higher-frequency program material. By comparison to the TRIO-AM’s preset for MW broadcasting , in the HF presets the threshold of limiting of the 150Hz band has been substantially lowered so that more gain reduction (and thus, less bass) is produced.

The multiband distortion-cancelling clipper prevents hard-clipped bass square waves from appearing at OPTIMOD TRIO-AM-AM's output. Older transmitters may respond much better to this well-controlled, benign waveform than to the hard-clipped bass square waves produced by less sophisticated processing.

The equalizer, like the classic Orban analog parametrics such as the 622B, has constant “Q” curves. This means that the cut curves are narrower than the boost curves. The width (in octaves) is calibrated with reference to 10 dB boost. As you decrease the amount of EQ gain (or start to cut), the width in octaves will decrease. However, the “Q” will stay constant.

"Q" is a mathematical parameter that relates to how fast ringing damps out. (Technically, we are referring to the "Q" of the poles of the equalizer transfer function, which does not change as you adjust the amount of boost or cut.)

The curves in the TRIO-AM's equalizer were created by a so-called "minimax" ("minimize the maximum error," or "equal-ripple") IIR digital approximation to the curves provided by the Orban 622B analog parametric equalizer. Therefore, unlike less sophisticated digital equalizers that use the "bilinear transformation" to generate EQ curves, the shapes of the TRIO-AM's curves are not distorted at high frequencies.

Midrange Parametric Equalizer is a parametric equalizer whose boost and cut curves closely emulate those of an analog parametric equalizer with conventional bell-shaped curves.

MID FREQ determines the center frequency of the equalization, in Hertz. Range is 250-6000Hz.

MID GAIN determines the amount of peak boost or cut (in dB) over a ±10 dB range.

MID WIDTH determines the bandwidth of the equalization, in octaves. The range is 0.8-4.0 octaves. If you are unfamiliar with using a parametric equalizer, 1 octave is a good starting point.

The audible effect of the midrange equalizer is closely associated with the amount of gain reduction in the midrange bands. With small amounts of gain reduction, it boosts power in the presence region. This can increase the loudness of such material substantially. As you increase the gain reduction in the midrange bands (by turning the MULTIBAND DRIVE (Multiband Drive) control up), the MID GAIN control will have progressively less audible effect. The compressor for the midrange bands will tend to reduce the effect of the MID frequency boost (in an attempt to keep the gain constant) to prevent excessive stridency in program material that already has a great deal of presence power. Therefore, with large amounts of gain reduction, the density of presence region energy will be increased more than will the level of energy in that region.

We advise using this control conservatively, because excessive midrange boost can sound strident and fatiguing. Further, bear in mind that any equalization that affects frequencies above about 3kHz will affect wideband and narrowband radios very differently; the narrowband radios will reproduce very little of the above-3kHz equalization. So, if there are wideband radios in the hands of your audience, be sure to double-check your sound on both wideband and narrowband radios before settling on a setting for the midrange equalization.

A 1 kHz boost is particularly effective in increasing the loudness of the low-quality radios that are down about 3dB at 2kHz with steep-slope rolloff thereafter (the third group of radios described in *Reference Radios for the Processing* on page 3-2-27). If you are going to do this, set the equalizer's BANDWIDTH to approximately two octaves.

We do not recommend this 1 kHz equalization if you want a sound that is competitive with FM, because it will produce the old-fashioned AM honk. We do not believe that type of sound is appropriate for the tastes of today's audiences. The 1kHz boost should be used only as a last resort - the resulting sound will be louder, but it will also be far less competitive with FM.

High Frequency Parametric Equalizer is a parametric equalizer whose boost and cut curves closely emulate those of an analog parametric equalizer with conventional bell-shaped curves.

HIGH FREQ determines the center frequency of the equalization, in Hertz. The range is 1-15 kHz.

HIGH GAIN determines the amount of peak boost or cut over a ±10 dB range.

HIGH WIDTH determines the bandwidth of the equalization, in octaves. The range is 0.8-4.0 octaves. If you are unfamiliar with using a parametric equalizer, one octave is a good starting point.

Excessive high frequency boost can exaggerate hiss and distortion in program material that is less than perfectly clean. We suggest no more than 4 dB boost as a practical maximum, unless source material is primarily from compact discs of recently recorded material. In several of our presets, we use this equalizer to boost the upper presence band (4.4 kHz) slightly, leaving broadband HF boost to the BRILLIANCE and/or HF ENHANCE controls.

HF Gain ("High Frequency Shelf Gain") determines the amount of high frequency boost provided by the TRIO-AM's receiver equalizer.

HF Curve ("High Frequency Shelf Curve") determines the shape of the high frequency shelving curves curve produced by the TRIO-AM's receiver equalizer. (See **Error! Reference source not found.**.)

The high-frequency receiver equalizer is designed to compensate for the high frequency rolloff in average AM radios. The typical AM radio is down 3dB at 2kHz and rolls off at least 18dB/octave after that. The HF equalizer provides an 18dB/octave shelving preemphasis that can substantially improve the brightness and intelligibility of sound through narrowband radios that do not have an abrupt rolloff. The HF equalizer has two controls: a gain control that determines the height of the shelving curve (dB), and a curve control, calibrated with an arbitrary number that determines how abruptly the shelving equalizer increases its gain as frequency increases. 0 provides the most abrupt curve; 10 provides the gentlest. The HF CURVE control is used to trade off harshness on wider band radios against brightness in narrowband radios.

An HF CURVE of 0 provides the same equalization that was originally supplied as standard on early OPTIMOD TRIO-AM-AM 9100 units and was later provided by the 9100's green module. Compared to higher settings of the HF Curve control, it provides much more boost in the 5 kHz region, and tends to sound strident on wideband radios. However, it can be very effective where narrowband radios remain the norm.

With an HF CURVE setting of 0, an HF GAIN control setting of 22 dB will result in a perceived bandwidth of 6 kHz on "Group 2" AM radios (see page 3-2-28); a 15 dB setting yields a 5 kHz perceived bandwidth, 10 dB yields 4 kHz, and 5 dB yields 3 kHz. Advancing the HF GAIN control will result in a brighter, higher fidelity sound, but it will also require that the listener tune the radio more carefully.

If most of your listeners have wider-band radios (as may be the case in North America), use the NRSC curve, which can be chosen with the HF CURVE control. For a somewhat brighter sound that can benefit narrowband radios more, yet is still compatible with wideband NRSC radios, use HF CURVE = 10 and HF GAIN = 10dB. HF CURVE = 10 corresponds to the RED preemphasis module in Orban's analog 9100-series OPTIMOD TRIO-AM-AM processors.

Note that the added brightness caused by using an HF CURVE of 10 (as opposed to using NRSC) may tend to increase the first-adjacent interference being generated by your station, contrary to the purpose and intentions of the NRSC.

HF CURVE settings between 0 and 10 smoothly interpolate between the two extremes, and provide more flexibility for user adjustment. An HF CURVE setting of 5 provides the curve family associated with the YELLOW preemphasis module in Orban's analog 9100-series OPTIMOD TRIO-AM-AM processors.

With the HF CURVE control at any setting other than NRSC, extreme amounts of high-frequency boost may result in a slight `lisping' quality on certain voices. This is because the high-frequency boost will increase the high-frequency content of sibilant voices, which can only be boosted to 100% modulation. Since the spectral balance of the voice is altered, this may be perceived as a lisping sound.

The receiver equalizer is of limited benefit to narrowband radios with abrupt rolloffs. We believe that these radios benefit more from a boost at 3 kHz, combined with very little HF shelving EQ. These radios have almost no response at 5 kHz and above, so boosting frequencies above 5 kHz wastes modulation. Using a bell-shaped boost at 3 kHz causes the boost to decline naturally at frequencies that the radio cannot reproduce. You can use either the midrange or HF parametric equalizer to create such a boost.

This technique can also be useful if you are limiting your transmitted audio bandwidth to 5 or 6 kHz, which is required in much of the world and is becoming more common in North America. OPTIMOD TRIO-AM-AM's sharp lowpass filters (necessary to control occupied RRF bandwidth adequately) can produce audible ringing, which many people find objectionable. By limiting the amount of boost at the cutoff frequency of the OPTIMOD TRIO-AM-AM lowpass filter, you can reduce the audibility of filter ringing.

DJ BASS ("DJ Bass Boost") control determines the amount of bass boost produced on some male voices. 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 resets 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 ON, which allows band 1 and band 2 to be gated independently.

The amount of bass boost will depend 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, so 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.

If the GATE THRESH (Gate Threshold) control is turned OFF, the DJ BASS boost setting is disabled.

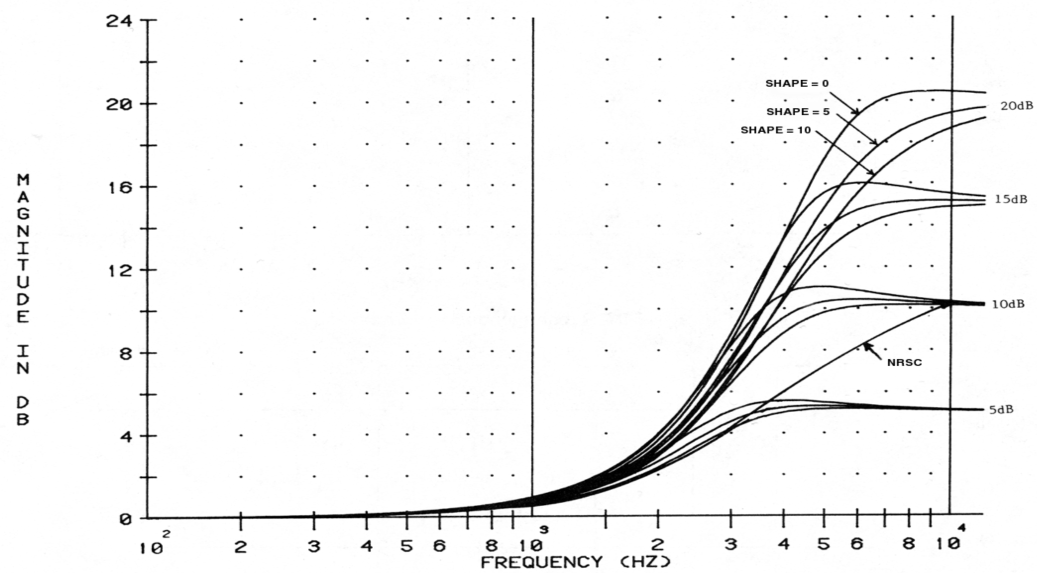


Figure 2-7: HF Receiver Equalizer Curves

HF ENHANCER ("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.

B1-B5 DRIVE ("Band 1-5 Input Drive") controls provide further equalization capability. This set of controls can be used like a graphic equalizer to get a different "flavor" of equalization from that provided by the other equalizer controls. Because the crossovers have 18dB/octave slopes, this equalization is quite frequency-selective by comparison to graphic equalizers with which you may be familiar. To prevent excessive coloration and headroom problems, the range of these controls is limited to 3dB. Thus, the multiband drive controls should be used to fine-tune equalization provided by the other equalizers.

LOWPASS ("Lowpass Filter Cutoff Frequency") allows you to decrease (but not increase) the low-pass cutoff frequency compared to its setting in active transmission preset. See step (**Error! Reference source not found.** on page 2-**Error! Bookmark not defined.**.

LPF SHAPE ("Lowpass Filter Shape") allows you to decrease (but not increase) the low-pass filter's shape compared to its setting in active transmission preset. See step (**Error! Reference source not found.** on page 2-**Error! Bookmark not defined.**.

HIGHPASS ("Highpass Filter Cutoff Frequency") allows you to increase (but not decrease) the highpass cutoff frequency compared to its setting in active transmission preset. See step (**Error! Reference source not found.** on page 2-**Error! Bookmark not defined.**.

AGC Controls

Several of the AGC controls are common to the Full Control and Advanced Control screens, with additional AGC controls available in the Advance Modify screen, as noted in the following table. (Note that "advanced" controls are accessible only from TRIO-AM PC Remote software.)

These controls are explained in detail below.

Each Factory Preset has a LESS-MORE control that adjusts on-air loudness by altering the amount of processing. LESS-MORE simultaneously adjusts all of the processing controls to optimize the trade-offs between unwanted side effects.

If you wish, you may adjust the Advanced Control parameters to your own taste. Always start with LESS-MORE to get as close to your desired sound as possible. Then edit the Advanced Control parameters using the Advanced Control screen, and save those edits to a User Preset.

AGC ("AGC Off/On") control activates or defeats the AGC.

It is usually used to defeat the AGC when you want to create a preset with minimal processing (such as a classical or fine arts preset). The AGC is also ordinarily defeated if you are using a studio level controller (like Orban's 6300). However, in this case it is better to defeat the AGC globally in System Setup.

AGC DRIVE control adjusts signal level going into the slow dual-band AGC, and therefore determines the amount of gain reduction in the AGC. 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.)

AGC Controls		
Full Control Name	Advanced Name	Range
AGC	AGC Off/On	Off/On
AGC DRIVE	AGC Drive	–10 ... 25 dB
AGC REL	AGC Master Release	0.5, 1.0, 1.5, 2 … 20 dB/S
AGC GATE	AGC Gate Threshold	Off, –44 ... –15 dB
AGC B CPL	AGC Bass Coupling	Off, 12 … 0 dB
AGC METR	AGC Meter Display	Master, Delta
---	AGC Window Size	–25 … 0 dB
---	AGC Ratio	2:1, 3:1, 4:1, infinity:1
---	AGC Window Release	0.5 … 20 dB
---	AGC Bass Threshold	–12.0 … 2.5 dB
---	AGC Idle Gain	–10 … +10 dB
---	AGC Bass Attack	1 … 10
---	AGC Master Attack	0.2 … 6
---	AGC Bass Release	1 … 10 dB/sec

Table 2-8: AGC Controls

The total amount of gain reduction in the TRIO-AM processing 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 MULTIBAND DRIVE (compressor) control.

AGC REL (“AGC Master Release”) control provides an adjustable range from 0.5 dB/second (slow) to 20 dB/second (fast). The increase in density caused by setting the AGC RELEASE control to fast settings sounds different from the increase in density caused by setting the multiband compressor’s MULTIBAND RELEASE control to FAST, and you can trade the two off to produce different effects.

Unless it is purposely speeded-up (with the AGC RELEASE control), the automatic gain control (AGC) that occurs in the AGC prior to the multiband compressor makes audio levels more consistent without significantly altering texture. Then the multiband compression and associated multiband clipper audibly change the density of the sound and dynamically re-equalize it as necessary (booming bass is tightened; weak, thin bass is brought up; highs are always present and consistent in level).

The various combinations of AGC and compression offer great flexibility:

Light AGC + light compression yields a wide sense of dynamics, with a small amount of automatic re-equalization.

Moderate AGC + light compression produces an open, natural quality with automatic re-equalization and increased consistency of frequency balance.

Moderate AGC + moderate compression gives a more dense sound, particularly as the release time of the multiband compressor is sped up.

Moderate AGC + heavy compression (particularly with a FAST multiband release time) results in a “wall of sound” effect, which may cause listener fatigue.

Adjust the AGC (with the AGC DRIVE control) to produce the desired amount of AGC action, and then fine-tune the compression and clipping with the TRIO-AM processing’s controls.

AGC GATE (“AGC Gate Threshold”) control determines the lowest input level that will be recognized as program by OPTIMOD TRIO-AM-AM; lower levels are considered to be noise or background sounds and cause the AGC or multiband compressor to gate, effectively freezing gain to prevent noise breathing.

There are two independent gating circuits in the TRIO-AM. The first affects the AGC and the second affects the multiband compressor. Each gate has its own threshold control.

The multiband compressor gate causes the gain reduction in bands 2 and 3 of the multiband compressor to move quickly to the average gain reduction occurring in those bands when the gate first turns on. This prevents obvious midrange coloration under gated conditions, because bands 2 and 3 have the same gain.

The gate also independently freezes the gain of the two highest frequency bands (forcing the gain of the highest frequency band to be identical to its lower neighbor), and independently sets the gain of the lowest frequency band according to the setting of the DJ BASS boost control (in the Equalization screen). Thus, without introducing obvious coloration, the gating smoothly preserves the average overall frequency response “tilt” of the multiband compressor, broadly maintaining the "automatic equalization" curve it generates for a given piece of program material.

If the MB GATE THR (Gate Threshold) control is turned OFF, the DJ BASS control is disabled.

AGC B CPL ("AGC Bass Coupling") control clamps the amount of dynamic bass boost (in units of dB) that the two-band AGC can provide.

The AGC processes audio in a master band for all audio above approximately 200Hz, and a bass band for audio below approximately 200Hz. The AGC B CPL control determines how closely the on-air balance of material below 200Hz matches that of the program material above 200Hz.

The AGC Master and Bass compressor sidechains operate without internal coupling. The gain reduction in the Bass audio path is either the output of the Bass compressor sidechain or the output of the Master band sidechain. The AGC BASS COUPLING control sets the switching threshold. For example, if the AGC BASS COUPLING control is set to 4 dB and the master gain reduction is 10 dB, the bass gain reduction in the audio path cannot decrease below 6 dB (10 – 4 = 6 dB) even if the gain reduction signal from the Bass compressor sidechain is lower. However, the bass gain reduction can be larger than the master gain reduction without limit. In the previous example, the bass gain reduction could be 25 dB.

The normal setting of the AGC BASS COUPLING control is 0 dB, which allows the AGC bass band to correct excessive bass as necessary but does not permit it to provide a bass boost.

AGC METR ("AGC Meter Display") determines what the AGC meter shows the gain reduction of the slow two-band AGC processing that precedes the multiband compressor. Full-scale is 25 dB gain reduction. MASTER displays the gain reduction of the Master (above-200 Hz) band. DELTA displays the difference between the gain reduction in the Master and Bass bands.

Although it is located in the Full Control screen (to make it easy for a preset developer to switch meter modes), this control is *not* part of the active preset and its setting is not saved in User Presets, unlike the other controls in the Full Control screens. The meter mode always reverts to MASTER when the user leaves Full Control.

Advanced AGC Controls

The following AGC controls are available only in the TRIO-AM PC Remote software.

AGC Window Size determines the size of the floating "slow zone" window in the master band of the AGC. (The Bass band is not windowed.)

The window works by slowing down changes in the AGC gain reduction that are smaller than the AGC WINDOW SIZE. The window has 2:1 asymmetry around the current AGC gain reduction. For example, if the AGC WINDOW SIZE is set to 4 dB, the window extends 4 dB in the release direction and 2 dB in the attack direction.

If the AGC needs to respond to a large change in its input level by making a gain change that is larger than the window, then the AGC's attack and release controls determine the AGC's response time. However, if the change in input level is smaller than the window size, the AGC WINDOW RELEASE control determines the attack and release times. This is usually much slower than the normal AGC time constants. This prevents the AGC from building up density in material whose level is already well controlled.

The previous explanation was somewhat simplified. In fact, the window has "soft edges." Instead of switching abruptly between time constants, the attack and release times morph smoothly between the setting of the WINDOW RELEASE control and the setting of the AGC master release and attack controls.

The normal setting for the AGC WINDOW SIZE is 3dB.

AGC Window Release (see AGC WINDOW SIZE above.)

AGC Bass Threshold determines the compression threshold of the bass band in the AGC. It can be used to set the target spectral balance of the AGC.

As the AGC B CPL control is moved towards "100%," the AGC BASS THRESHOLD control affects the sound less and less.

The interaction between the AGC BASS THRESHOLD control and the AGC B CPL control is a bit complex, so we recommend leaving the AGC BASS THRESHOLD control at its factory setting unless you have a good reason for readjusting it.

AGC Ratio determines the compression ratio of the AGC. 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.

AGC Idle Gain. The "idle gain" is the target gain of the AGC when the silence gate is active. Whenever the silence gate turns on, the gain of the AGC slowly moves towards the idle gain.

The idle gain is primarily determined by the AGC DRIVE setting — a setting of 10 dB will ordinarily produce an idle gain of –10 dB (i.e., 10 dB of gain reduction). However, sometimes you may not want the idle gain to be the same as the AGC DRIVE setting. The AGC IDLE GAIN control allows you to add or subtract gain from the idle gain setting determined by the AGC DRIVE setting.

You might want to do this if you make a custom preset that otherwise causes the gain to increase or decrease unnaturally when the AGC is gated.

For example, to make the idle gain track the setting of the AGC DRIVE control, set the AGC IDLE GAIN control to zero. To make the idle gain 2 dB lower than the setting of the AGC DRIVE control, set the AGC IDLE GAIN control to –2.

AGC Bass Attack sets the attack time of the AGC bass compressor (below 200Hz).

AGC Master Attack sets the attack time of the AGC master compressor (above 200Hz).

AGC Bass Release sets the release time of the AGC bass compressor.

Clipper Controls

B1-B5 Clip Thresh controls set the thresholds of four clippers in Orban's patented embedded multiband distortion-canceled clipper in units of dB with reference to the final clipper. These clippers are embedded in the multiband crossover so that any distortion created by clipping is rolled off by part of the crossover filters. The band 1+2 clipper operates on the sum of bands 1 and 2; these bands are not clipped separately.

The threshold of the Band 1+2 clipper is usually set between 2 dB and 6 dB below the threshold of the final limiter in the processing chain, depending on the setting of the LESS-MORE control in the parent preset on which you are basing your Modify adjustments. This provides headroom for contributions from the other three bands so that bass transients don't smash against the back-end clipping system, causing overt intermodulation distortion between the bass and higher frequency program material.

Some TRIO-AM users feel that the band 1+2 clipper unnecessarily reduces bass punch at its factory settings. As you raise the threshold of the clipper, you will get more bass but also more distortion and pumping. Be careful when setting this control; do not adjust it casually. Listen to program material with heavy bass combined with spectrally sparse midrange material (like a singer accompanied by a bass guitar) and listen for IM distortion induced by the bass' pushing the midrange into the clipping system. In general, unless you have a very good reason to set the control elsewhere, we recommend leaving it at the factory settings, which were determined following extensive listening tests with many types of critical program material.

This advice also holds for the thresholds of the other band clippers. These clippers prevent audible distortion in the final clipper and their settings are very critical.

MB CLIP ("Multiband Clipper") control adjusts signal level going into the multi-band clippers and therefore determines the amount of peak limiting done by clipping. Range is –4dB to +5dB.

This control and the FINAL CLIP DRIVE control govern the trade-off between loudness and distortion. Note that these ranges are relative, and do not indicate the exact amount of clipping. 0dB refers to the average setting that we have found to be the best compromise for many settings of the other controls.

If the MB RELEASE control is set to FAST or MFAST (medium-fast), perceived clipping distortion will increase as the MB DRIVE control is advanced, and the MB CLIP control may have to be turned down to compensate. To best understand how to make loudness/distortion trade-offs, perhaps the wisest thing to do is to recall a factory preset and then to adjust the LESS-MORE control to several settings throughout its range. At each setting of the LESS-MORE control, examine the settings of the MB DRIVE control, the MB CLIP control, and the FINAL CLIP drive control. This way, you can see how the factory programmers made the trade-offs between the settings of the various distortion-determining controls at various levels of processing.

HF Clipping determines the amount of protection provided by the TRIO-AM's Band 5 multiband clipper. "0" is the preferred setting. Higher values will increase both brightness and high frequency distortion. This control is for audio processing experts only; it is important to listen to a very wide range of program material before deciding to set this control above 0. When the control is set above 0, there will definitely be some program material that sounds harsh and distorted, so you must decide if the trade-off against brighter sound and more vocal presence is acceptable to your taste.

Clipper Controls		
Full Control Name	**Advanced Name**	**Range**
FINAL CLIP	Final Clip Drive	0 … +5.0 dB
MB CLIP	Multiband Clipping	–4.0 … +5.0
---	B1+B2 Clip Threshold	–16.00 … –1.25 dB, Off
---	Band 3 Clip Threshold	–16.00 … –1.25 dB, Off
---	Band 4 Clip Threshold	–16.00 … –1.25 dB, Off
---	Band 5 Clip Threshold	–16.00 … –1.25 dB, Off
---	High Frequency Clipping	0 … 6

Table 2-9: Clipper Controls

Final Clip ("Final Clip Drive") adjusts the level of the audio driving the back end clipping system that OPTIMOD TRIO-AM-AM uses to control fast peaks. This control primarily determines the loudness/distortion trade-off.

Turning up the FINAL CLIP control drives the final clipper and overshoot compensator harder, reducing the peak-to-average ratio, and increasing the loudness on the air. When the amount of clipping is increased, the audible distortion caused by clipping also increases. Although lower settings of the FINAL CLIP control reduce loudness, they make the sound cleaner.

If the RELEASE control is set to its faster settings, the distortion produced by the back-end clipping system will increase as the MULTIBAND DRIVE control is advanced. The FINAL CLIP DRIVE and/or the MULTIBAND LIMIT THRESHOLD controls may have to be turned down to compensate. To best understand how to make loudness/distortion trade-offs, perhaps the wisest thing to do is to recall a factory multiband preset, and then to adjust the LESS-MORE control to several settings throughout its range. At each setting of the LESS-MORE control, examine the settings of the MULTIBAND DRIVE and MULTIBAND LIMIT THRESHOLD controls. This way, you can see how the factory programmers made the trade-offs between the settings of the various distortion-determining controls at various levels of processing.

The TRIO-AM's multiband clipping and distortion control system works to help prevent audible distortion in the final clipper. As factory programmers, we prefer to adjust the FINAL CLIP control through a very narrow range and to determine almost all of the loudness/distortion trade-off by the setting of the Multiband Clipping control.

Multiband Controls		
Full Name	**Advanced Name**	**Range**
MB DRIVE	Multiband Drive	0 ... 25
MB GATE	Multiband Gate Threshold	Off, –44 ... –15 dB
DWNEXP TH	Downward Expander	Off, –6.0 … 12.0 dB
MB CLIP	Multiband Clipping	–4.0 … +5.0
HF CLIP	High Frequency Clip Threshold	–16.00 … 0.0, Off
LESS-MORE	LESS-MORE Index	[read-only]; 1.0 … 10.0
PARENT PRESET	Parent Preset	[read-only]
---	High Frequency Limiter	Off, –23.8 ... 0.0 dB
---	B1/B2 XOVER	100 Hz, 200 Hz

Table 2-10: Multiband Controls

Multiband Dynamics Processing

MB DRIVE ("Multiband Drive") control adjusts the signal level going into the five-band compressor, hence determining the average amount of gain reduction in the multiband compressor. Range is 25dB.

Adjust the MULTIBAND DRIVE control to your taste and format requirements. Used lightly with slower multiband release times, the multiband compressor produces an open, re-equalized sound. The multiband compressor can increase audio density when operated at faster release times because it acts increasingly like a fast limiter (not a compressor) as the release time is shortened. With faster release times, density also increases when you increase the drive level into the multiband compressor because these faster release times produce more limiting action. Increasing density can make sounds seem louder, but can also result in an unattractive busier, flatter, or denser sound. It is very 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.

The MULTIBAND DRIVE interacts with the MULTIBAND RELEASE setting. With slower release time settings, increasing the MULTIBAND DRIVE control scarcely affects density. Instead, the primary danger is that the excessive drive will cause noise to be increased excessively when the program material becomes quiet.

You can minimize this effect by carefully setting the MULTIBAND GATE THRESHOLD control to "freeze" the gain when the input gets quiet and/or by activating the single-ended noise reduction.

When the release time of the multiband compressor is set to its faster settings, the setting of the MULTIBAND DRIVE control becomes much more critical to sound quality because density increases as the control is turned up. Listen carefully as you adjust it. With these fast release times, there is a point beyond which increasing multiband compressor drive will no longer yield more loudness, and will simply degrade the punch and definition of the sound.

We recommend no more than 10 dB gain reduction as shown on the meters for band 3. More than 10dB, particularly with the FAST release time, will often create a "wall of sound" effect that many find fatiguing.

To avoid excessive density with the FAST multiband release time, we recommend using no more than 5 dB gain reduction in band 3, and compensating for any lost loudness by speeding up the MULTIBAND RELEASE instead. This is what we did in the factory LESS-MORE presets for the FAST multiband release time.

MB REL ("Multiband Release") control can be switched to any one of seven settings:

The **Slow** (SLOW and SLOW2) settings produce a very punchy, clean, open sound that is ideal for Adult Contemporary, Soft Rock, Soft Urban, New Age, and other adult-oriented formats whose success depends on attracting and holding audiences for very long periods of time.

MB Attack / Release / Threshold		
Full Name	**Advanced Name**	**Range**
MB REL	Multiband Release	Slow, Slow2, Med, Med2, MFast, MFast2, Fast
---	B1 Compression Threshold	–16.0 … –1.5, Off
---	B2 Compression Threshold	–16.0 … –1.5, Off
---	B3 Compression Threshold	–16.0 … –1.5, Off
---	B4 Compression Threshold	–16.0 … –1.5, Off
---	B5 Compression Threshold	–16.0 … –1.5, Off
---	B1 Attack	8 … 23ms, Off
---	B2 Attack	8 … 23ms, Off
---	B3 Attack	8 … 23ms, Off
---	B4 Attack	8 … 23ms, Off
---	B5 Attack	8 … 23ms, Off
---	B1 Delta Release	–6 … 6
---	B2 Delta Release	–6 … 6
---	B3 Delta Release	–6 … 6
---	B4 Delta Release	–6 … 6
---	B5 Delta Release	–6 … 6

Table 2-11: MB Attack/Release Controls

The SLOW and SLOW2 settings produce an unprocessed sound with a nice sense of dynamic range. With these settings, the TRIO-AM processing provides gentle automatic equalization to keep the frequency balance consistent from record to record (especially those recorded in different eras). And for background music formats, these settings ensure that your sound doesn't lose its highs and lows. Because it creates a more consistent frequency balance between different pieces of source material than does the Two-Band structure, SLOW is almost always preferable to the Two-Band structure for any popular music format.

The **Medium Slow** settings (MED and MED2) are appropriate for more adult-oriented formats that need a glossy show-business sound, yet whose ratings depend on maintaining a longer time spent listening than do conventional Contemporary Hit Radio (CHR) formats. With the single-ended noise reduction activated, it is also appropriate for Talk and News formats. This is the sound texture for the station that values a clean, easy-to-listen-to sound with a tasteful amount of punch, presence, and brightness added when appropriate. This is an unprocessed sound that sounds just right on music and voice when listened to on small table radios, car radios, portables, or home hi-fi systems.

The **Medium Fast** settings (MFAST and MFAST2) are ideal for a highly competitive Contemporary Hit Radio (CHR) format whose ratings depend on attracting a large number of listeners (high "cume") but which does not assume that a listener will listen to the station for hours at a time. This is the major market competitive sound, emphasizing loudness as well as clean audio. The sound from cut to cut and announcer to announcer is remarkably consistent as the texture of music is noticeably altered to a standard. Bass has an ever-present punch, there is always a sense of presence, and highs are in perfect balance to the mids, no matter what was on the original recording.

The **Fast** setting is used for the TALK and SPORTS factory programming formats. Processing for this sound keeps the levels of announcers and guests consistent, pulls low-grade telephone calls out of the mud, and keeps a proper balance between voice and commercials. Voice is the most difficult audio to process, but these settings result in a favorable trade-off between consistency, presence, and distortion.

It is possible to experiment with this sound for music-oriented programming as well. However, even with these settings, your sound is getting farther away from the balance and texture of the input. We think that this is as far as processing can go without causing unacceptable listener fatigue. However, this sound may be quite useful for stations that are ordinarily heard very softly in the background because it improves intelligibility under these quiet listening conditions. Stations that are ordinarily played louder will probably prefer one of the slower release times, where the multiband compressor takes more gain reduction and where the AGC is operated slowly for gentle gain riding only. These slower sounds are less consistent than those produced by the FAST setting. Using SLOW preserves more of the source's frequency balance, making the sound less dense and fatiguing when the radio is played loudly.

Bx THR ("Band x Compression Threshold") controls set the compression threshold in each band, in units of dB below the final clipper threshold. We recommend making only small changes around the factory settings to avoid changing the range over which the MB CLIPPING control operates. These controls will affect the spectral balance of the processing above threshold, but are also risky because they can strongly affect the amount of distortion produced by the back-end clipping system.

MB GATE ("Multiband Gate Threshold") control determines the lowest input level that will be recognized as program by OPTIMOD TRIO-AM-AM; lower levels are considered to be noise or background sounds and cause the AGC or multiband compressor to gate, effectively freezing gain to prevent noise breathing.

There are two gating circuits in the TRIO-AM. The first affects the AGC and the second affects the multiband compressor. Each has its own threshold control. The TRIO-AM's input drives the AGC gate detector; the output of the TRIO-AM's AGC drives the MB gate detector.

The multiband compressor gate causes the gain reduction in bands 2 and 3 of the applicable multiband compressor to move quickly to the average gain reduction occurring in those bands when the gate first turns on. This prevents obvious midrange coloration under gated conditions, because bands 2 and 3 have the same gain.

The gate also independently freezes the gain of the two highest frequency bands (forcing the gain of the highest frequency band to be identical to its lower neighbor), and independently sets the gain of the lowest frequency band according to the setting of the DJ BASS boost control (in the Equalization screen). Thus, without introducing obvious coloration, the gating smoothly preserves the average overall frequency response "tilt" of the multiband compressor, broadly maintaining the "automatic equalization" curve it generates for a given piece of program material.

Note: If the MB GATE THRESH (Gate Threshold) control is turned OFF, the DJ BASS control (in the Equalization screen) is disabled.

DwnExp Thr ("Downward Expander Threshold") determines the level below which the single-ended noise reduction system's downward expander begins to decrease system gain, and below which the high frequencies begin to become low-pass filtered to reduce perceived noise. Activate the single-ended dynamic noise reduction by setting the DWNEXP THR control to a setting other than OFF.

The single-ended noise reduction system combines a broadband downward expander with a program-dependent low-pass filter. These functions are achieved by introducing extra gain reduction in the multiband compressor. You can see the effect of this extra gain reduction on the gain reduction meters.

Ordinarily, the gating on the AGC and multiband limiter will prevent objectionable build-up of noise, so you will want to use the single-ended noise reduction only on unusually noisy program material. Modern commercial recordings will almost never need it. We expect that its main use will be in talk-oriented programming, including

Band Mix		
Full Name	**Advanced Name**	**Range**
B2>B1 CPL	B2>B1 Coupling	0 ... 100 %
B1 OUT	B1 Output Mix	–3.0 … +3.0
B2 OUT	B2 Output Mix	–3.0 … +3.0
B3 OUT	B3 Output Mix	–3.0 … +3.0
B4 OUT	B4 Output Mix	–3.0 … +3.0
B5 OUT	B5 Output Mix	–3.0 … +3.0
---	B1 On/Off	Band On, Band Off
---	B2 On/Off	Band On, Band Off
---	B3 On/Off	Band On, Band Off
---	B4 On/Off	Band On, Band Off
---	B5 On/Off	Band On, Band Off

sports.

Table 2-12: MB Band Mix Controls

Please note that it is impossible to design such a system to handle all program material without audible side effects. You will get best results if you set the DWNEXP THR control of the noise reduction system to complement the program material you are processing. The DWNEXP THR should be set higher when the input is noisy and lower when the input is relatively quiet. The best way to adjust the DWNEXP THR control is to start with the control set very high. Reduce the control setting while watching the gain reduction meters. Eventually, you will see the gain increase in sync with the program. Go further until you begin to hear noise modulation — a puffing or breathing sound (the input noise) in sync with the input program material. Set the DWNEXP THR control higher until you can no longer hear the noise modulation. This is the best setting.

Obviously, the correct setting will be different for a sporting event than for classical music. It may be wise to define several presets with different settings of the DWNEXP THR control, and to recall the preset that complements the program material of the moment.

Note also that it is virtually impossible to achieve undetectable dynamic noise reduction of program material that is extremely noisy to begin with, because the program never masks the noise. It is probably wiser to defeat the dynamic noise reduction with this sort of material (traffic reports from helicopters and the like) to avoid objectionable side effects. You must let your ears guide you.

B2>B1 CPL control determines the extent to which the gain of band 1 (below 100Hz or 200Hz, depending on crossover setting) is determined by and follows the gain of band 2 (centered at 400Hz). Set towards 100% (fully coupled) it reduces the amount of dynamic bass boost, preventing unnatural bass boost in light pop and talk formats. Set towards 0% (independent), it permits frequencies below 100Hz (the "slam" region) to have maximum impact in modern rock, urban, dance, rap, and other music where bass punch is crucial.

HF CLIP ("High Frequency Clipper Threshold") sets the threshold of clipping in bands 4 and 5 with reference to the overall threshold set by the MB CLIP control. The range is 0 to +6dB. We have made this control available for some major-market customers who prefer a brighter sound at the expense of audible distortion on a significant amount of program material. We recommend a setting of 0 for all program material.

This control is for audio processing experts only; it is important to listen to a very wide range of program material before deciding to set this control above 0. When the control is set above 0, there will definitely be some program material that sounds harsh and distorted and you must decide if the trade-off against brighter sound and more vocal presence is acceptable to your taste.

Advanced Multiband Controls

The following Advanced Multiband controls are available only from TRIO-AM PC Remote software.

B1-B5 Out ("Band 1-5 Output Mix") controls determine the relative balance of the bands in the multiband compressor. Because these controls mix *after* the band compressors, they do not affect the compressors' gain reductions and can be used as a graphic equalizer to fine-tune the spectral balance of the program material over a ±3 dB range.

Their range has been purposely limited because the only gain control element after these controls is the back-end clipping system (including the final clipper and overshoot compensator), which can produce considerable audible distortion if overdriven. We have carefully tuned the thresholds of the individual compressors to prevent audible distortion with almost any program material. Large changes in the frequency balance of the compressor outputs will change this tuning, leaving the TRIO-AM more vulnerable to unexpected audible distortion with certain program material. Because all these controls are located *before* the compressor, you should make large changes in EQ with the bass and parametric equalizers, the HF enhancer, and the individual band drive controls. The compressors will protect the system from unusual overloads caused the chosen equalization. Use the output mix controls only for fine-tuning and spend time with a variety of program material to make sure that your adjustments have not caused excessive clipping distortion.

You can also get a similar effect by adjusting the compression threshold of the individual bands. This is comparably risky with reference to clipper overload, but unlike the MB BAND MIX controls, does not affect the frequency response when a given band is below threshold and is thus producing no gain reduction.

B1-B5 On/Off switches allow you to listen to any band (or any combination of bands) independently. This is a feature designed for intermediate or advanced users and developers when they are creating new TRIO-AM presets.

Please note that a single band will interact with the back-end clipping system quite differently than will that band when combined with all of the other bands. Therefore, do not assume that you can tune each band independently and have it sound the same when the clipping system is processing all bands simultaneously.

B1-B5 Attack (Time) controls set the speed with which the gain reduction in each band responds to level changes at the input to a given band's compressor. These controls are risky and difficult to adjust appropriately. They affect the sound of the processor in many subtle ways. The main trade-off is "punch" (achieved with slower attack times) versus distortion (because slower attack times increase overshoots that must be eliminated in the clipping system). The results are strongly program-dependent and must be verified with listening tests to a wide variety of program material.

The ATTACK time controls are calibrated in arbitrary units. Higher numbers correspond to slower attacks.

Delta Release controls are differential controls. They allow you to vary the release time in any band of the Five-Band compressor/limiter by setting an offset between the MULTIBAND RELEASE setting and the actual release time you achieve in a given band. For example, if you set the MULTIBAND RELEASE control to medium-fast and the BAND 3 DELTA GR control to –2, then the band 3 release time will be the same as if you had set the MULTIBAND RELEASE control to medium and set the BAND 3 DELTA GR control to 0. Thus, your settings automatically track any changes you make in the MULTIBAND RELEASE control. In our example, the release time in band 3 will always be two "click stops" slower than the setting of the MULTIBAND RELEASE control.

If your setting of a given DELTA RELEASE control would otherwise create a release slower than "slow" or faster than "fast" (the two end-stops of the MULTIBAND RELEASE control), the band in question will instead set its release time at the appropriate end-stop.

B1/B2 Xover (Band 1 to Band 2 Crossover Frequency) sets the crossover frequency between bands 1 and 2 to either 150 Hz or 200 Hz. It significantly affects the bass texture, and the best way to understand the differences between the two crossover frequencies is to listen.

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.

Specifications for TRIO-AM

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. Therefore, 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

5) Except as noted, specifications apply for measurements from the analog left/right input to the analog left/right output.

6) **Frequency Response (Bypass Mode):** ±0.2 dB, 50 Hz–15 kHz, or as determined by user-settable high-pass filter in the active transmission preset.

7) **Noise:** Output noise floor will depend upon how much gain the processor is set for (Limit Drive, AGC Drive, Two-Band Drive, and/or Multiband 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.

8) **Total System Distortion** (de-emphasized, 100% modulation): <0.01% THD, 20 Hz–1 kHz, rising to <0.05% at 9.5 kHz. <0.02% SMPTE IM Distortion.

9) **Polarity**: The processing employs phase rotation to maximize loudness. Therefore, the polarity is frequency-dependent.

10) **Processing Sample Rate:** The Trio-AM is a "multirate" system, using internal rates from 32 kHz to 128 kHz as appropriate for the processing being performed. Audio clippers operate at 128 kHz.

11) **Low-Pass Filter:** 4.5, 5.0, 5.5, 6.0, 6.5, 7.0, 7.5, 8.0, 8.5, 9.0, or 9.5 (NRSC) kHz as set by user. Unit can be set up to comply easily with ITU-R and NRSC spectrum masks. Lowpass filter shape is parametric and can be set to be –0.1, –3, or –6 dB down at the cutoff frequency (see **Error! Reference source not found.** on page 2-**Error! Bookmark not defined.**).

12) **High-Pass Filter:** Constrained by user settable fifth-order "quasi-elliptical" highpass filter to 50, 60, 70, 80, 90, or 100 Hz. All filters have equal-ripple (Chebychev-like) passbands and a 25 Hz notch for transmitter protection.

13) **Channel Configuration:** Processing is monophonic.

Installation

Analog Audio Input

14) **Configuration:** Stereo (Configurable to drive the processing selection with mono from left, mono from right, or mono from sum).

15) **Impedance:** >10kΩ load impedance, electronically balanced[1].

16) **Nominal Input Level:** Software adjustable from –9.0 to +13.0 dBu (VU) / –2.0 to +20.0 dBu (PPM)

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[1] No jumper selection available for 600Ω. Through-hole pads are available on I/O module for user-installed 600Ω termination.

17) **Maximum Input Level:** +27 dBu.

18) **Connectors:** Two XLR-type, female, EMI-suppressed. Pin 1 chassis ground, Pins 2 (+) and 3 electronically balanced, floating and symmetrical.

19) **A/D Conversion:** 24 bit 128x oversampled delta sigma converter with linear-phase anti-aliasing filter. Converter outputs 64 kHz sample rate, which the Trio-AM then decimates to 32 kHz in DSP using an ultra-high-quality image-free synchronous sample rate converter.

20) **Filtering:** RFI filtered, with high-pass filter at 0.15 Hz (–3 dB).

Analog Audio Output

21) **Configuration:** Two monophonic outputs, capable to driving two transmitters. The two outputs have independent level controls.

22) **Source Impedance:** 351Ω (includes the third-order output EMI suppression network), electronically balanced and floating. The user can specify the output impedance in software to calibrate the output level accurately into a bridging or 600Ω load.

23) **Load Impedance:** 600Ω or greater, balanced or unbalanced. Termination not required or recommended.

24) **Output Level** (100% peak modulation): Adjustable from –6 dBu to +24 dBu peak, into 600Ω or greater load, software-adjustable.

25) **Polarity:** When the POLARITY control (located in the active System Preset) is set to POSITIVE, a positive-going signal at pin 2 of the XLR-type connector corresponds to positive modulation.

26) **Signal-to-Noise:** >= 90 dB unweighted (Bypass mode, de-emphasized, 20 Hz–9.5 kHz bandwidth, referenced to 100% modulation).

27) **Distortion:** <= 0.01% THD (Bypass mode, de-emphasized) 20 Hz–9.5 kHz bandwidth.

28) **Connectors:** Two XLR-type, male, EMI-suppressed. Pin 1 chassis ground, Pins 2 (+) and 3 electronically balanced, floating and symmetrical.

29) **D/A Conversion:** 24 bit 128x oversampled.

30) **Filtering:** RFI filtered.

Digital Audio Input

31) **Configuration:** Stereo per AES3 standard, 24 bit resolution, software processing selection of mono from left, mono from right or mono from sum.

32) **Sampling Rate:** 32, 44.1, 48, 88.2, or 96 kHz, automatically selected.

33) **Connector:** XLR-type, female, EMI-suppressed. Pin 1 chassis ground, pins 2 and 3 transformer balanced and floating, 110Ω impedance.

34) **Input Reference Level**: Variable within the range of –30 dBFS to –7 dBFS (VU) / –23 dBFS to 0.0 dBFS

35) **Filtering:** RFI filtered.

Digital Audio Output

36) **Configuration:** Dual-channel per AES3 standard. Each channel carries an identical monophonic signal.

37) **Sample Rate:** Internal free running at 32, 44.1, 48, 88.2 or 96 kHz, selected in software. 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.

38) **Word Length:** Software selected for 24, 20, 18, 16 or 14-bit resolution. First-order highpass noise-shaped dither can be optionally added. Dither level automatically adjusted appropriately for the word length.

39) **Connector:** XLR-type, male, EMI-suppressed. Pin 1 chassis ground, pins 2 and 3 transformer balanced and floating, 110Ω impedance.

40) **Output Level** (100% peak modulation): –20.0 to 0.0 dBFS software controlled. Level must be set lower than –2.0 dBfs to prevent clipping positive peaks at 125% modulation.

41) **Filtering:** RFI filtered.

AOIP (Audio Over IP)

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, RAVENNA supports Dante compliant
- **Number of Input Channels Supported:** Two (2) stereo pairs.
- **Number of Output Channels Supported:** Two (2) sum mono pairs
- **Sample Rate:** 44.1, 48, 88.2 and 96 kHz.
- **Networking:** One (1) RJ45 Ethernet connector 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)

OPTIONAL FEATURES

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

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.

Streaming Backup Receiver (optional)

Audio Decoders: na

Audio Bitrate: na

Audio Sample Rate: na

Audio Channels: na

Supported Streaming Servers: N/A

Emergency Player (optional)

Internal Storage: 2 GB Flash Memory

Audio Storage: 2 hours linear audio (.wav), 12 hours of (MP3 or OPUS)

Remote Computer Interface

42) HTML-5 Web Browser.

43) Ethernet Connector: Female RJ45 connector for 10 Mbps and higher networks using CAT5 cabling. Native rate is 1000 Mbps (Gigabit Ethernet).

Remote Control (GPI) Interface

44) **Configuration:** Eight (8) inputs, opto-isolated and floating.

45) **Voltage:** 6–15V AC or DC, momentary or continuous. 12 VDC provided to facilitate use with contact closure.

46) **Connector:** DB-25 male, EMI-suppressed.

47) **Control:** User-programmable for any four of user presets, factory presets, bypass, test tone, mono L mode, mono R mode, mono sum mode, analog input, digital input, clock reset, status bit pass/block, and Transmission presets.

48) **Filtering:** RFI filtered.

Tally Outputs

49) **Circuit Configuration:** Two NPN open-collector outputs.

50) **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.

51) **Current:** 30 mA maximum

52) **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 Input Silent, AES Input Error, AoIP Input Silent.

Power

53) **Voltage:** Two universal switching power supply, 80–264 VAC, 50–60 Hz, <65 VA

54) **Connector:** Two (2) IEC, EMI-suppressed. Detachable 3-wire power cord supplied.

55) **Fuse: :** T2A Quick Acting HBC, mounted on the power supply circuit board

56) **Grounding:** In order to meet EMI standards, circuit ground is hard-wired to chassis ground.

57) **Safety Standards:** ETL listed to UL standards, CE marked.

Environmental

58) **Operating Temperature:** 32° to 122° F / 0° to 50° C for all operating voltage ranges.

59) **Humidity:** 0–95% RH, non-condensing.

60) **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.

61) **Humidity:** 0–95% RH, non-condensing.

62) **RFI/EMI:** Tested according to Cenelec procedures. FCC Part 15 Class A device. KC certification; Clause 3, Article 58-2 of Radio Waves Act. (2024-03-27).

63) **Shipping Weight:** 19 lbs / 8.7 kg

Warranty

64) **Five Years, Parts and Service:** Subject to the limitations set forth in Orban's Standard Warranty Agreement.

Because engineering improvements are ongoing, specifications are subject to change without notice.

Section 3: OPTIMOD TRIO-FM

When you first energize your OPTIMOD Trio

During the boot process, you will be asked to pick which mode you want to use. For this section, choose FM. The TRIO will ask to confirm you selection. Press OK. The TRIO will not enter FM mode.

Where to locate the TRIO-FM

The best location for THE OPTIMOD-TRIO-FM 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-TRIO-FM's carefully peak-limited composite waveform—a short length of coaxial cable. If this is impossible, the next best arrangement is to feed the TRIO-FM's AES3 digital output through an all-digital, uncompressed path to the transmitter's exciter, although this will preclude using the TRIO-FM's composite limiter.

Use the TRIO-FM'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 TRIO-FM'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 TRIO-FM'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 TRIO-FM's output sample rate to 44.1 kHz or higher.

If only an audio link is available, use the TRIO-FM'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-TRIO-FM 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-TRIO-FM 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-TRIO-FM 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-TRIO-FM 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 TRIO-FM's composite limiter.

Because THE OPTIMOD-TRIO-FM controls peaks, it is irrelevant whether the audio link feeding THE OPTIMOD-TRIO-FM'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 TRIO-FM'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-TRIO-FM 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-TRIO-FM and the exciter.

If a separate stereo encoder must be used, feed the encoder directly from the TRIO-FM'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-TRIO-FM's pre-emphasis network and low-pass filters perform the same functions while retaining tight peak control. Connect the composite output of the TRIO-FM 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:

65) Pass unprocessed audio for application to the TRIO-FM's input.

66) Pass the TRIO-FM's peak controlled analog or digital left and right audio outputs.

67) Pass the TRIO-FM'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 TRIO-FM 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 TRIO-FM'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 TRIO-FM 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 TRIO-FM'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 TRIO-FM'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 TRIO-FM'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 TRIO-FM'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 TRIO-FM installation because they allow you to use the TRIO-FM'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 TRIO-FM 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 TRIO-FM 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 TRIO-FM.

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 TRIO-FM 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 TRIO-FM'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 TRIO-FM's remote control feature offers the means to reduce the peak level of the TRIO-FM's audio output as necessary.

This way, you can still use the TRIO-FM's line-up tone to adjust the steady-state deviation to 75kHz. Yet, the reduced peak level of the audio emitted from the TRIO-FM 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.

- Use the TRIO-FM's low-delay "monitor" output to drive talent headphones.

The normal delay through the TRIO-FM (from input to FM outputs) is about 18 ms when HARD or MEDIUM bass clipping is selected, as it is in 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 TRIO-FM's full look-ahead processing for low latency and about 2.5 dB loudness decrease for ultra-low latency.

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

68) 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).

69) 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 TRIO-FM. These presets differ from OPTIMOD 8200 processing in two main ways: (1) the TRIO-FM UL presets still use the TRIO-FM's stereo enhancement, equalization section, advanced-technology AGC, composite limiter, and multiplex power controller, and (2) the TRIO-FM 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 TRIO-FM'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 TRIO-FM 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 TRIO-FM can be controlled via a web browser pointed at the address assigned to it. All system controls are accessible there.

Racking the TRIO-FM

The TRIO-FM is designed to fit in a standard 19' (inch) rack. While the TRIO-FM 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 TRIO-FM requires three standard rack units (5 inches / 12.7 cm). There should be a good ground connection between the rack and the TRIO-FM 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 TRIO-FM should be energized. It cannot be stressed enough that Orban Labs recommends the TRIO-FM 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 TRIO-FM itself, it is good engineering practice to ensure all mission critical equipment is protected. The TRIO-FM input voltage is 100-240 VAC 50-60Hz, making the unit universal electrically.

Once the TRIO-FM 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 TRIO-FM.

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 TRIO-FM. 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 TRIO-FM 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 TRIO-FM'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 the FM Left/Right pre or de-emphasized 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 TRIO-FM input meters display 0dB. The reference level is adjustable using the DI REF control.
- The TRIO-FM 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 TRIO-FM'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.

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/

70) 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.

71) 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

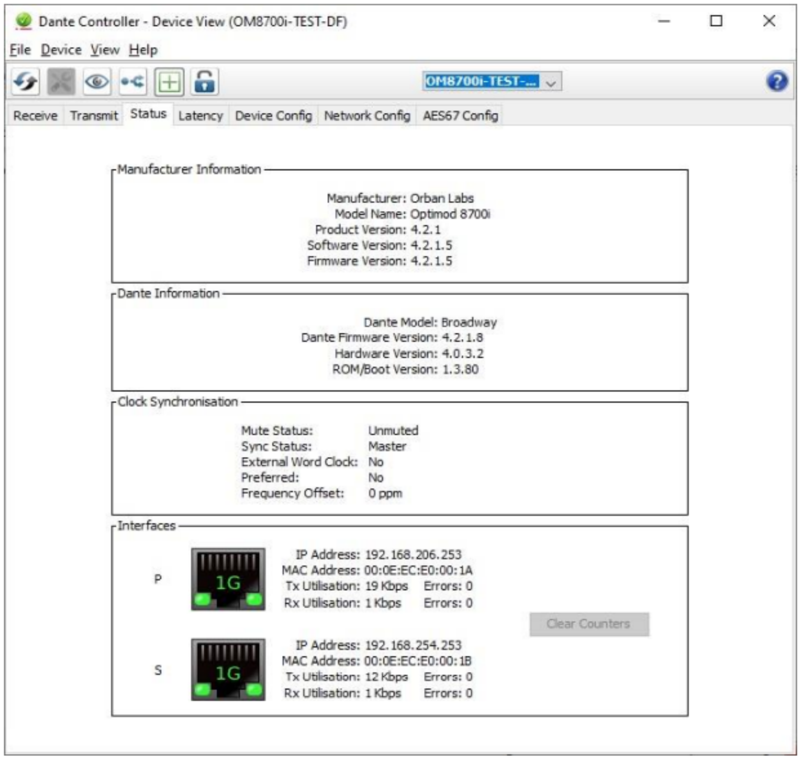
- Obtain an IP address from a DHCP server if it is present on the network

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 TRIO-FM 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 TRIO-FM.



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

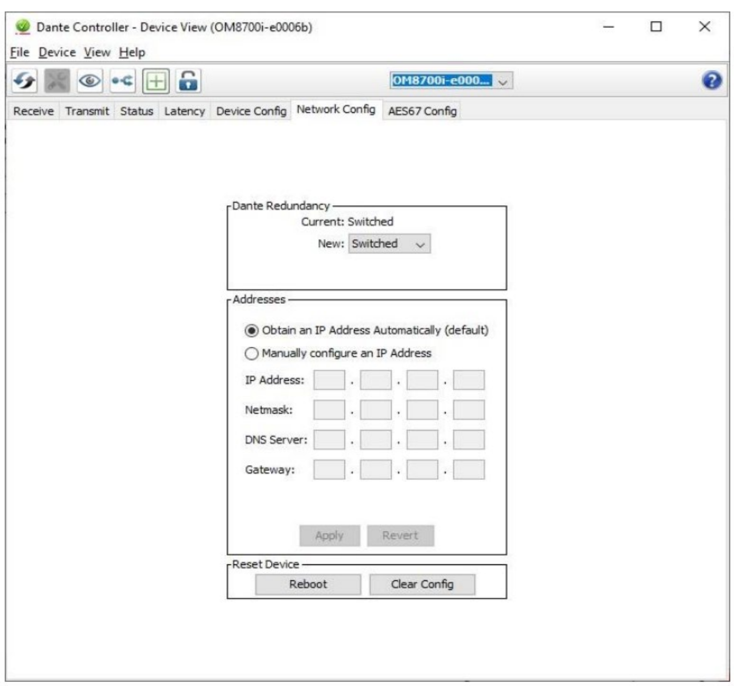


Figure 2-2: A SWITCHED setup with a DHCP-sourced IP address:

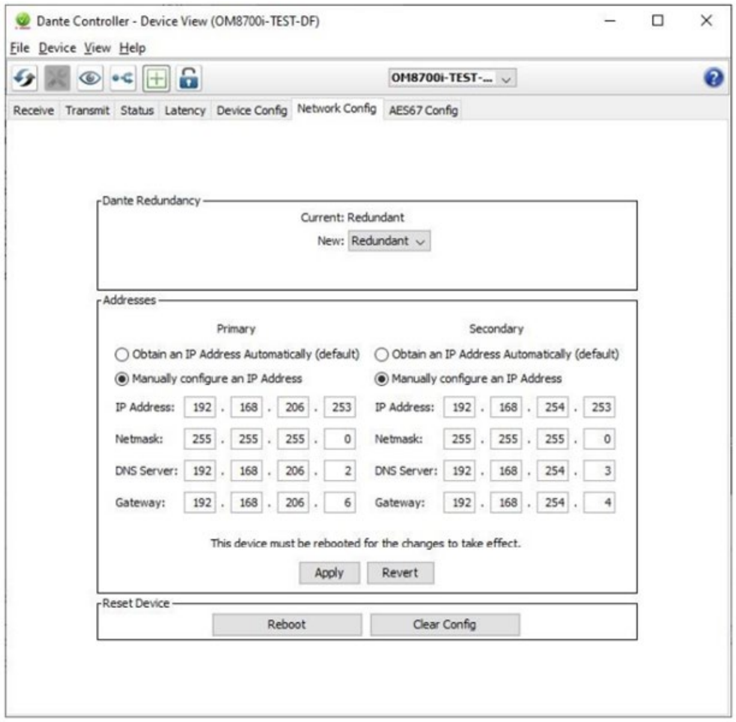
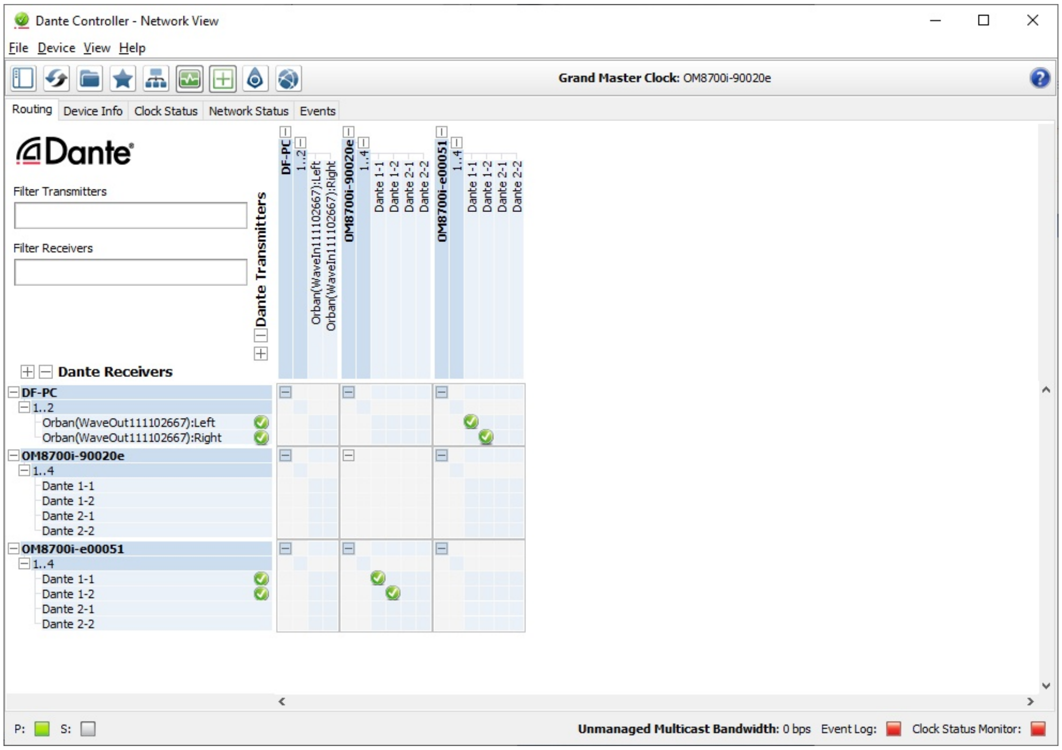


Figure 2-3: A Dante Controller reports the TRIO-FM network status when a TRIO-FM has been configured and connected for dual-redundant networks.

72) In the Dante network, route audio to and from the TRIO-FM.

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 TRIO-FM'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 TRIO-FM'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.

73) Configure the TRIO-FM'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.

c) 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.

d) Navigate to INPUT/OUTPUT > DANTE 1.

This screen is conceptually different from the other TRIO-FM 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.

e) 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.

f) Set up the DANTE 1 output controls:

- OUTPUT SOURCE
  Set to FM or MONITOR.

- 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.

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

h) 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 TRIO-FM 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 TRIO-FM 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 TRIO-FM'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 TRIO-FM'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 TRIO-FM 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 TRIO-FM'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 TRIO-FM 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 TRIO-FM's composite output in the analog domain, after D/A conversion of the TRIO-FM 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 TRIO-FM'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 TRIO-FM'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 TRIO-FM'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 TRIO-FM'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 TRIO-FM 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 TRIO-FM 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 TRIO-FM 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 TRIO-FM 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 TRIO-FM'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 TRIO-FM 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 TRIO-FM 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	

REMOTE INTERFACE

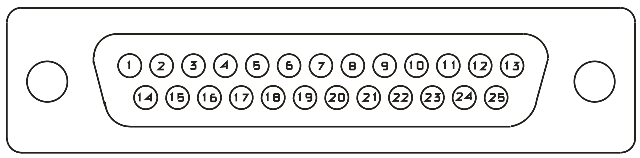
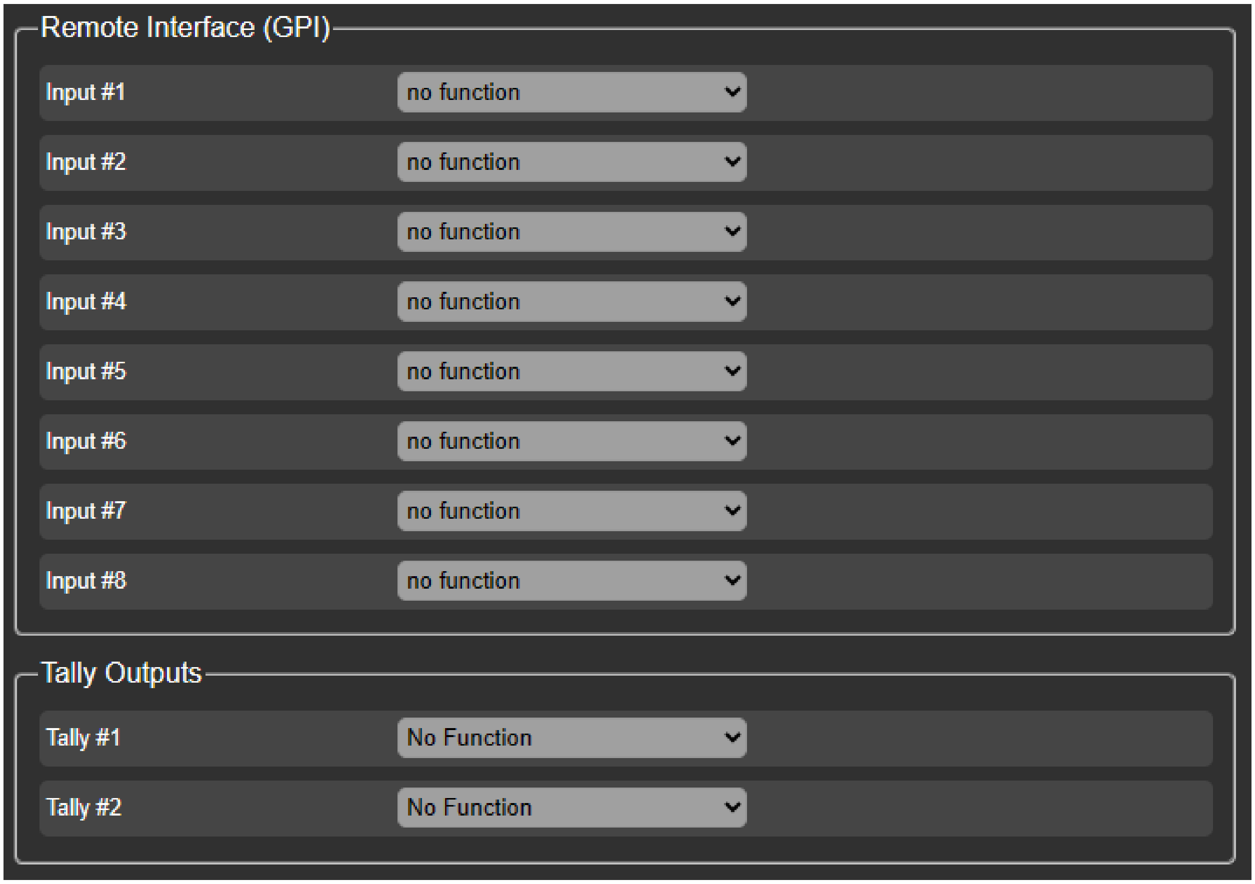


Figure 3-1: Wiring the 25-pin Remote Interface Connector



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 TRIO-FM, 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.

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

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

THE TRIO FRONT PANEL

The TRIO 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 TRIO. 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 failure. The type of failure is indicated by the cadence of the LED
- 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 TRIO 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:

  C) Monitoring current TRIO mode (AM or FM or HD)

  D) Analog In

  E) Digital In 1

F) Digital in 2

G) AoIP In 1

H) AoIP In 2

I) Stream In

Use the up and down dobby 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.

MAIN MENU

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

INFO

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

- System Version
- I/O Board Revision
- Aux PIC
- Power Pic
- Serial Number
- MAC Address
- IP, Subnet and Gateway addresses
- DNS
- IPv6 Address
- Hostname

Touching the > box will reveal the following information

- CPU Temperature
- MB Temperature
- Fan Speed
- Free Memory Flash #1

- Free Memory Flash #2
- Free Memory
- Voltages
- Power Supply Power Supply 1 and 2 voltages
- Power Supply Status

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

  J) 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..

  K) 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.

THE TRIO-FM FRONT PANEL

The TRIO-FM 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 TRIO-FM. 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 TRIO-FM 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) Analog In

E) Digital In 1

F) Digital in 2

G) AoIP In 1

H) AoIP In 2

I) 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 TRIO-FM and check on the status of the hardware.

INFO

Info is the diagnostic screens for the TRIO-FM. The following information can be found under the info tab. Use the “< and >” buttons to move through the pages of information

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 TRIO-FM.
- 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.

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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 TRIO-FM via WEB INTERFACE

The default address for the TRIO-FM 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 TRIO-FM when connecting to set up a proper network.

The TRIO-FM 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 TRIO-FM.

- If configured properly you will see the browser connect along with metering and control options.

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 TRIO-FM 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 TRIO-FM, the tradeoff between brightness and the other two parameters has been considerably improved (by 2.5 – 3 dB above 6 kHz) when an "TRIO-FM" 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 TRIO-FM—from Bach to Rock

You can adjust the OPTIMOD-TRIO-FM 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-TRIO-FM—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 TRIO-FM 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 TRIO-FM 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 TRIO-FM 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 TRIO-FM's Signal Processing Features

Dual-Mono Architecture

The TRIO-FM 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 TRIO-FM would always process stereo program material.

Signal Flow

The signal flows through the TRIO-FM 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 TRIO-FM 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 TRIO-FM’s output spectral control is immaculate, ensuring maximum stereo and RDS coverage. Because there is very little energy above 16 kHz, the TRIO-FM’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 TRIO-FM 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.

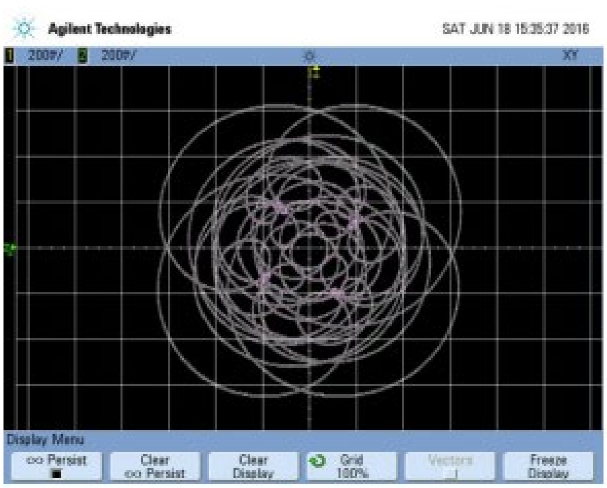
The second stereo enhancement algorithm is based on the well-known “Max” technique. This passes the L–R signal through a delay line and adds this decorrelated signal to the unenhanced L–R signal. Gating circuitry similar to that used in the “222- style” algorithm prevents over-enhancement and undesired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

“Multipath Mitigator” Left/Right Phase Skew Correction

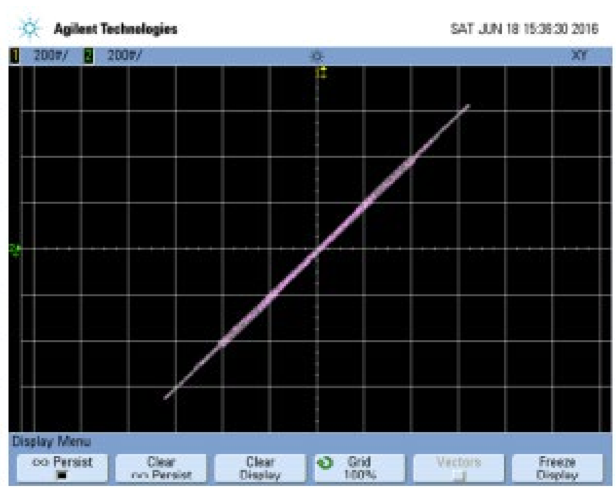
The phase skew corrector maximizes the quality of a mono mixdown or blend that might occur in a receiver. At higher frequencies (where audible comb filtering of the mono sum is most likely to occur), the corrector removes phase differences between the left and right channels, converting the HF signal 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 TRIO-FM 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 TRIO-FM’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 TRIO-FM 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 TRIO-FM controls high frequencies with distortioncanceled clipping and, in all but 5-band TRIO-FM 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 TRIO-FM's clipping system to do this.

Except in TRIO-FM 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, TRIO-FM presets use a sophisticated bass pre-limiter located immediately before the system's main distortion-controlled clipper. In non-TRIO-FM 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 TRIO-FM's speech/music detector can activate it automatically.

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

"Intelligent" Clipping: In non–TRIO-FM presets, the TRIO-FM 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.

TRIO-FM 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–TRIO-FM presets.

For more information on the TRIO-FM clipper technology, see Appendix D.

Speech Mode: You can set many of the processing parameters separately for speech signals, as detected by the TRIO-FM'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 TRIO-FM 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 TRIO-FM's stereo encoder is derived from algorithms first developed for the high-performance Orban 8218 stand-alone encoder. The TRIO-FM'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 TRIO-FM has two independent composite outputs, whose levels are both software settable. For convenience, two SCA inputs sum into the TRIO-FM'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 TRIO-FM does not digitize SCAs.

SSB Stereo Encoder Operation: The TRIO-FM 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 TRIO-FM'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 TRIO-FM's composite output signal extends to 38,150 Hz when the TRIO-FM'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 TRIO-FM'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 TRIO-FM'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 TRIO-FM'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 TRIO-FM's composite output, the TRIO-FM 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 TRIO-FM'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 TRIO-FM 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 TRIO-FM'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 TRIO-FM's main clipping system to do the vast majority of the work because of its sophisticated distortion-controlling mechanisms. This means that the TRIO-FM 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 TRIO-FM's processing—the TRIO-FM gets competitively loud without composite clipping. However, it is also possible to reduce the drive level to the TRIO-FM'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 TRIO-FM 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 TRIO-FM 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 TRIO-FM, 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 TRIO-FM 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 TRIO-FM'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 TRIO-FM'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.

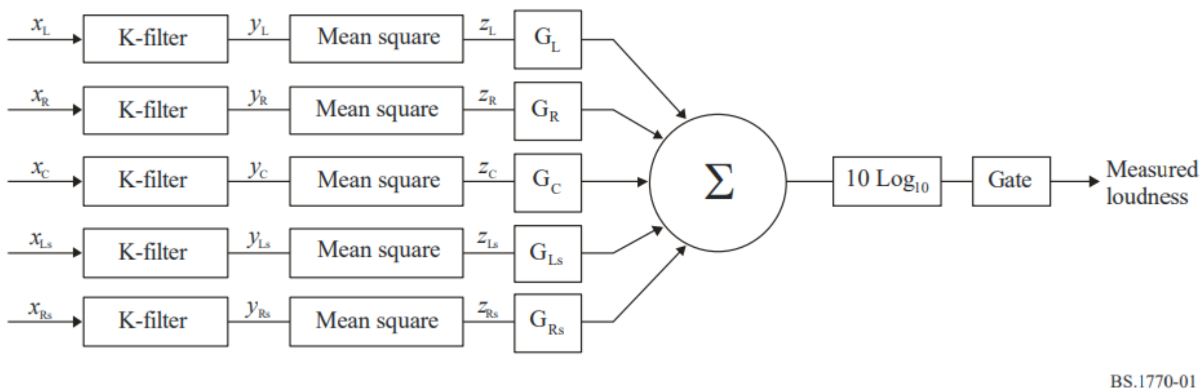
With the FM->HD CONTROL COUPLING set to INDEPENDENT, there are several approaches to minimizing brightness and conditioning the signal to work well at low bitrates.

- 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

The TRIO-FM 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.

BS.1770 Loudness Level



BS.1770 Loudness Meter Block Diagram (from ITU-R document)

The subjective loudness meters, labeled LOUDNESS in the Trio'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 Trio's digital radio processing chain is adjusted to make the dominant program material indicate "0 dB" on the Trio'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.

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 Trio's MPX POWER meter is therefore calibrated so that it indicates 0 dB when the composite output of the Trio 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 Trio'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 Trio. Such overshoots necessitate turning down the Trio's output level control after it has been calibrated with tone using an FM modulation meter and the Trio'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 Trio's output level control was turned down.

See ***Optimal Control of Peak Modulation Levels*** starting on page 1-2-10 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 Trio does not digitize subcarriers applied to its subcarrier inputs, the Trio'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 Trio 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 Trio 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 Trio. Setting the control at "0" will correctly control the multiplex power when there is no overshoot after the Trio. 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 Trio. For example, if RF peak deviation exceeds the peak deviation produced by the Trio'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 Trio 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 Trio'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 Trio'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 Trio'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.

Input/Output Delay

The sophisticated look-ahead algorithms in the TRIO-FM 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 TRIO-FM 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, "TRIO-FM" presets introduce delays of approximately 265 ms, which makes real-time monitoring through headphones impossible for talent. The TRIO-FM 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-TRIO-FM 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 TRIO-FM's Sound

Gain Reduction Metering

Unlike the metering on some processors, when any THE OPTIMOD-TRIO-FM 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-TRIO-FM 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 TRIO-FM 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 TRIO-FM 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 TRIO-FM will give you an opportunity to choose a new name for the preset you are saving.

You can save user presets from the TRIO-FM HTML5 application. Please note that when you save presets from the HTML5 application, you save them in the TRIO-FM's memory (as if you had saved them from the TRIO-FM'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 TRIO-FM

In the TRIO-FM, 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), TRIO-FM'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 TRIO-FM realizes its processing structures as a series of high-speed mathematical computations made by Digital Signal Processing (DSP) integrated circuit chips. In the TRIO-FM, 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 TRIO-FM 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 TRIO-FM 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 TRIO-FM’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 TRIO-FM’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, 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. 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.

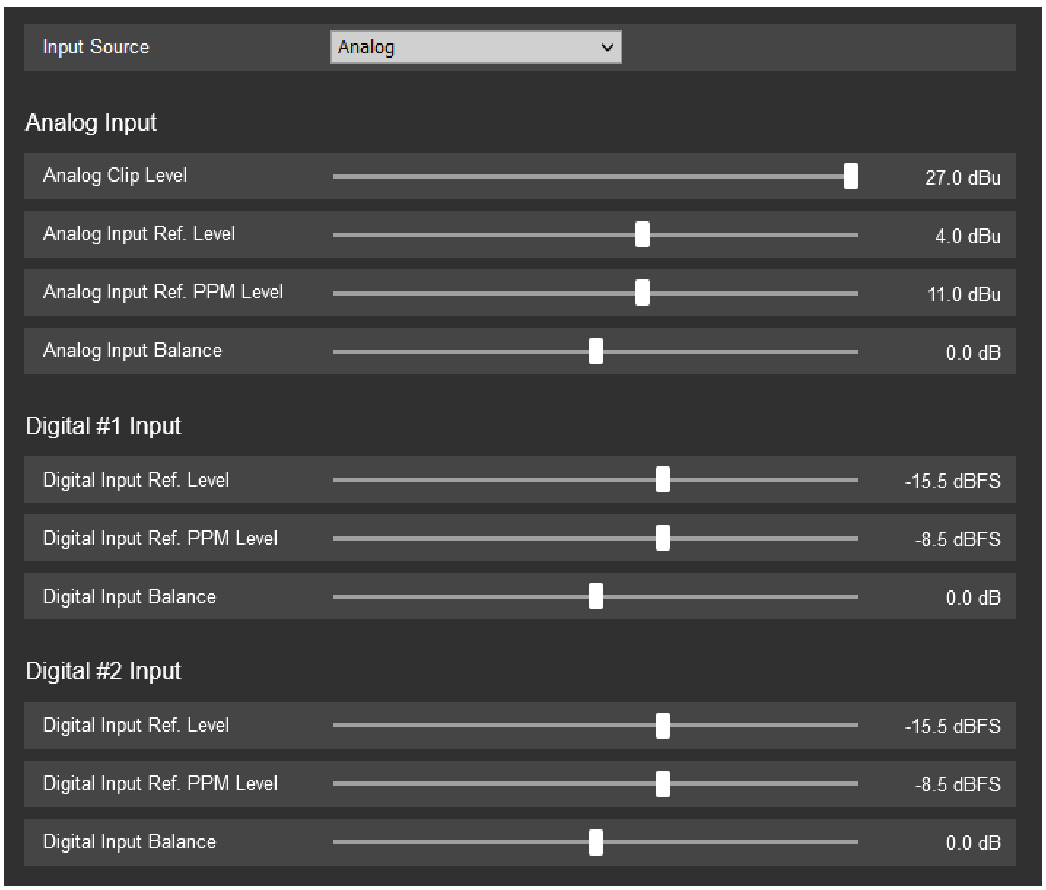
URBAN-LIGHT: This is an 8500-style preset that has been retained mainly for historical reasons.

Adjusting and using the browser based PC Remote

The TRIO-FM HTML5 application allows you to adjust every parameter in the TRIO-FM 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 TRIO-FM's gain reduction range to the level to which your studio operators peak their program material on the studio meters. This assures that the TRIO-FM'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 TRIO-FM 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 TRIO-FM's processing.

If you are using a non linear audio path, consider moving the TRIO-FM to the transmitter site if possible. This will restore the TRIO-FM'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
- Monitor (A low latency signal to feed talent headphones for an "off air" experience)

HEADPHONE

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

- FM
- Monitor (A low latency signal to feed talent headphones for an "off air" experience)
- 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 TRIO-FM 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 TRIO-FM's processing.

If you are using a non linear audio path, consider moving the TRIO-FM to the transmitter site if possible. This will restore the TRIO-FM'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 TRIO-FM's HD audio path)

Synchronization determines if the sample rate appearing at the digital-channel output is synced to the TRIO-FM's internal clock, to an AES3 signal appearing at the TRIO-FM's digital input, or to an AES11 signal appearing at the TRIO-FM'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 TRIO-FM'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 TRIO-FM'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 TRIO-FM Input or Sync In, set SR Sync to Internal and select the desired output sample rate.

The TRIO-FM 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 TRIO-FM is 24 bits. The TRIO-FM can also truncate its output word length to 20, 18, 16, or 14 bits. The TRIO-FM 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 TRIO-FM 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 TRIO-FM 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).

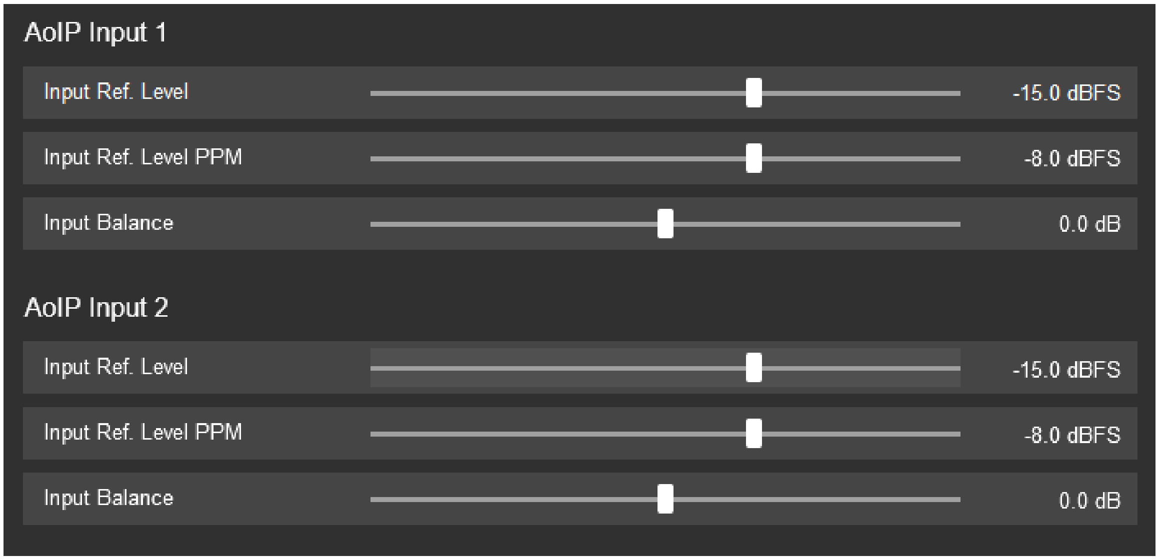
SCA 1: Sets the level of the digital SCA 1 input.

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 TRIO-FM's gain reduction range to the level to which your studio operators peak their program material on the studio meters. This assures that the TRIO-FM'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 TRIO-FM. 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 TRIO-FM's processing.

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

- FM
- Monitor (A low latency signal to feed talent headphones for an "off air" experience)

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

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

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

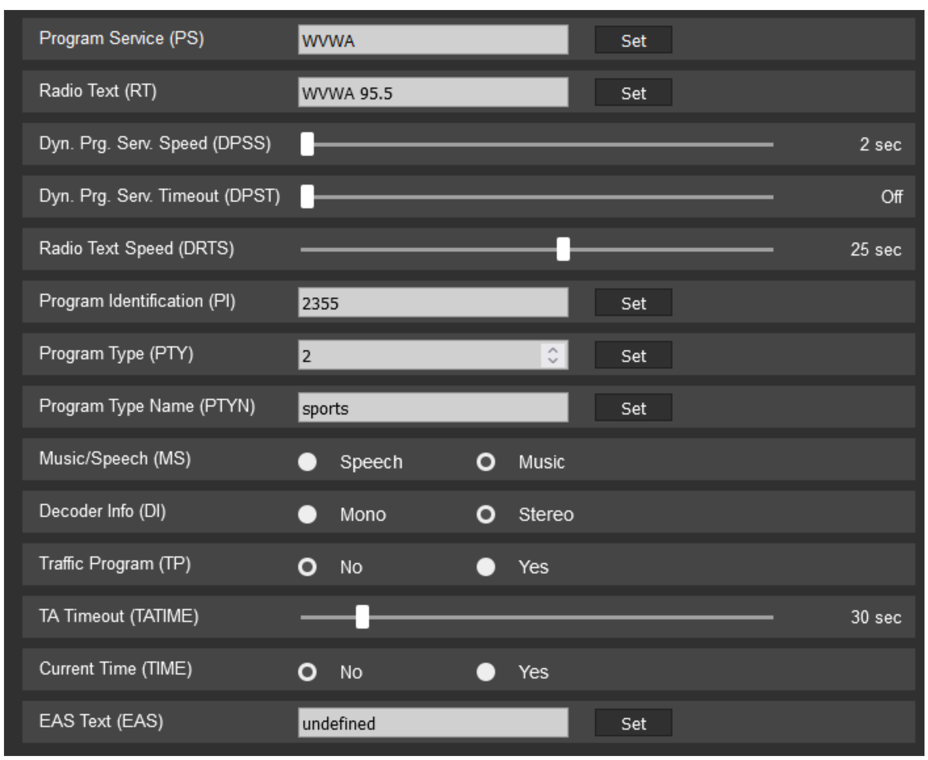
The amount of dither automatically tracks the setting of the Word Length control. This first-order noise shaped dither adds considerably less noise in the midrange than does white PDF dither. However, unlike extreme noise shaping, first-order noise shaped dither adds a maximum of 3 dB of excess total 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 TRIO-FM by turning this control on.

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

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 TRIO-FM 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)

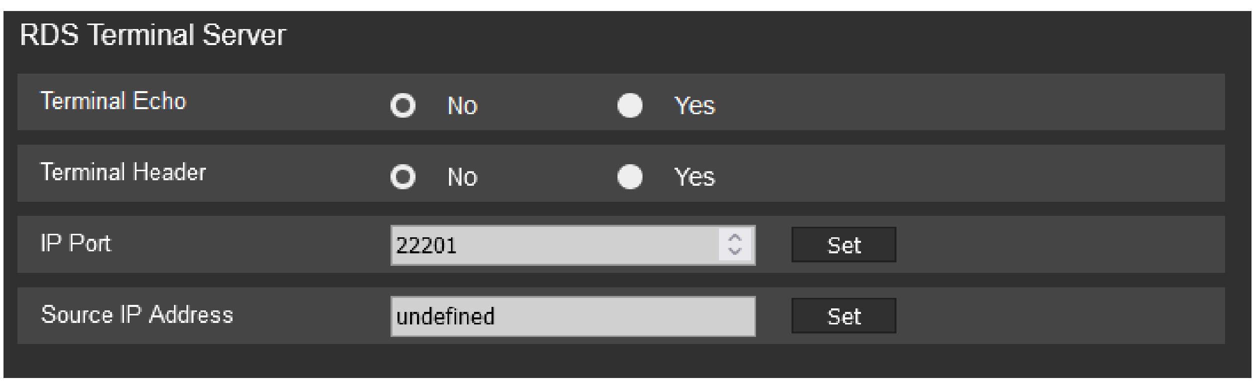
The RDS encoder supports network-based UECP connectivity with either the TCP/IP or UDP protocol. Use the PHTML5 application to set these client settings from the RDS tab in the I/O Setup dialog: Site Address, Encoder Address, TCP/IP Port number, UDP, and UECP Active status. Use the values suggested by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

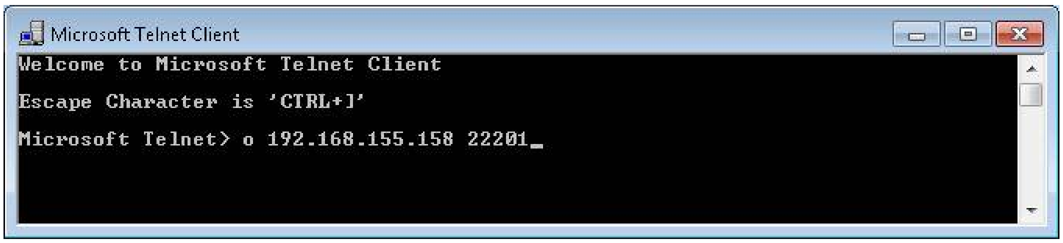


You can control RDS from Windows directly via the Windows Telnet command line utility or the free utility PuTTY.

L) 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."

M) 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.



N) 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 TRIO-FM 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↲

COMMAND	PARAMETER	INFORMATION	EXAMPLE
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**[2] 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.↵
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	TIMEOUT=15↵ (current connection

[2] 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
		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.*	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

MHz	CHAN	MHz	CHAN	MHz	CHAN	MHz	CHAN
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
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
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.

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

µMPX

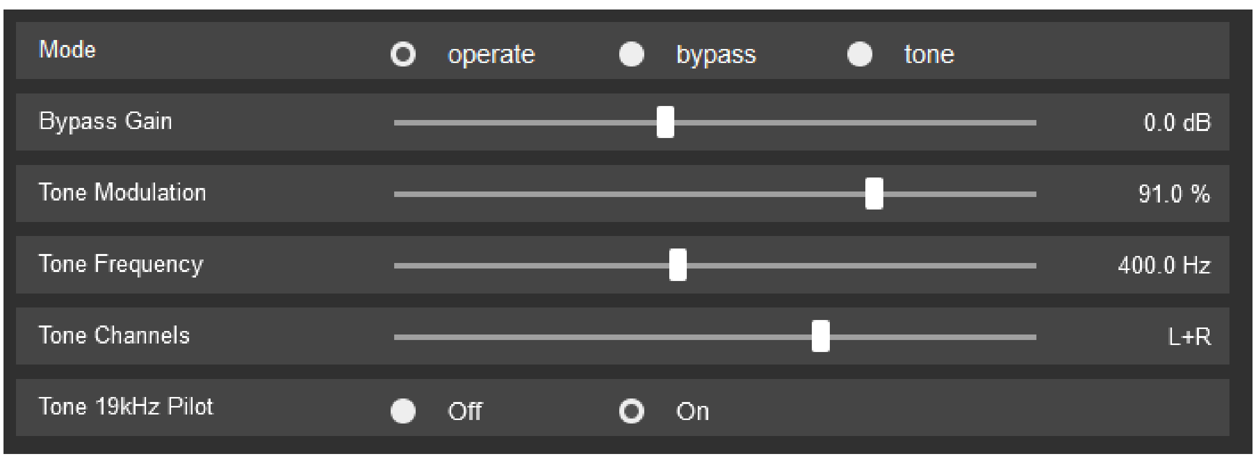
µMPX in its simplest form is AES-EBU digital audio sampled at 192 kHz. AES sampling is done at 44.1 or 48 kHz in most all broadcast plants. µMPX allows the FM multiplex signal including baseband audio, pilot, stereo sub, RDS and 67 kHz subcarriers to be processed and transmitted either directly at the 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 TRIO-FM 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 3-6: Digital MPX

Facilities are self-explanatory, except for the following:

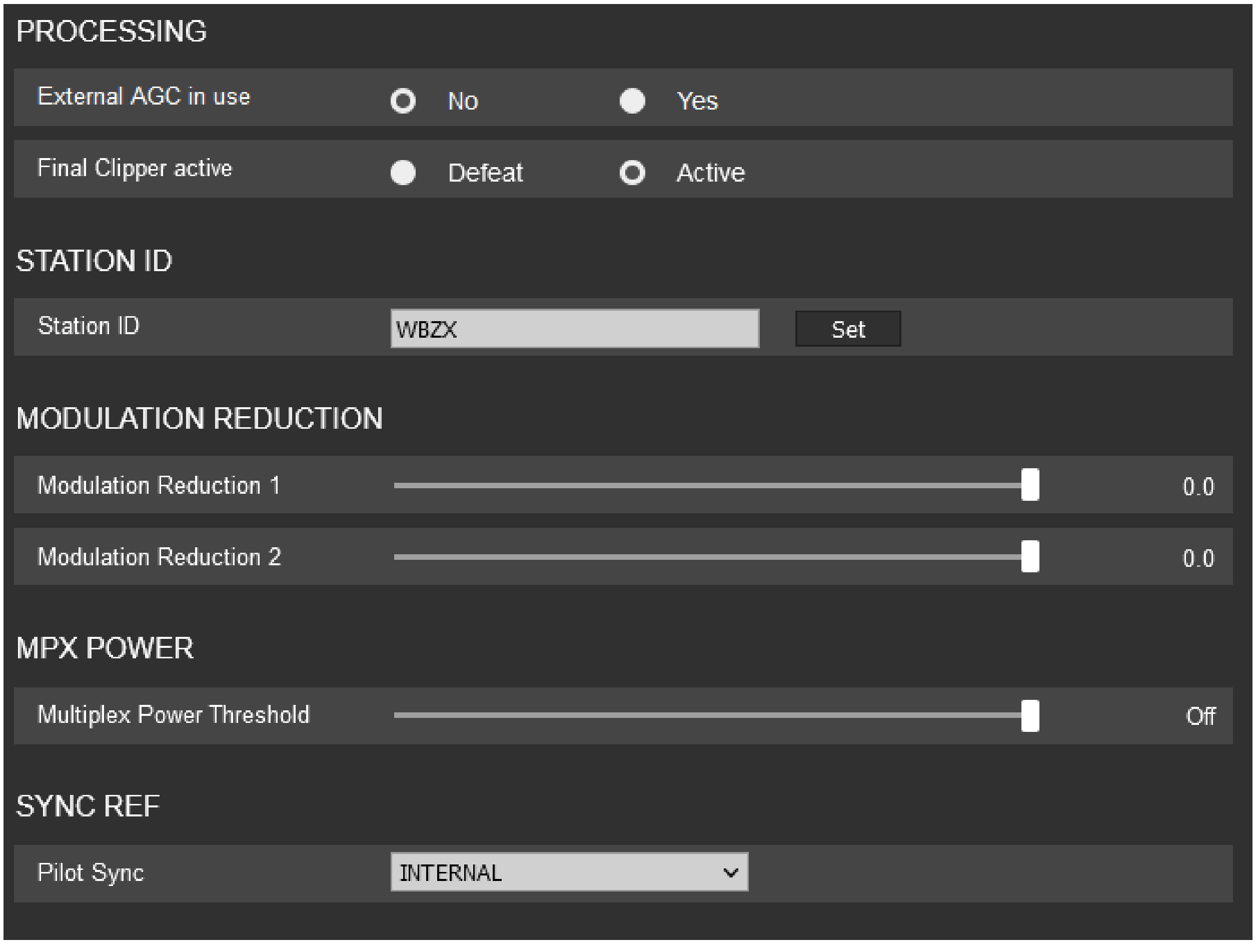
The TONE LVL control is calibrated under the assumption that the stereo pilot tone contributes 9% to the total modulation. Hence, a TONE LVL setting of 91% produces 100% modulation (91% + 9%).

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 TRIO-FM 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 TRIO-FM 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 TRIO-FM 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 TRIO-FM 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 TRIO-FM 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.

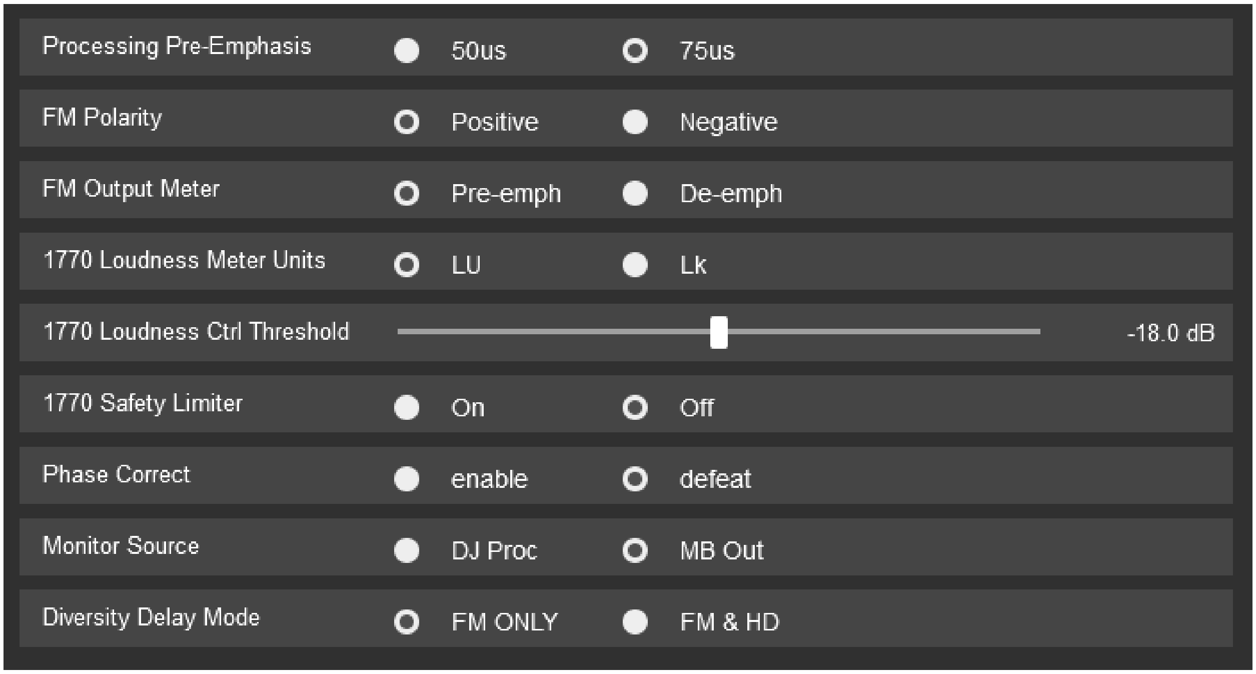
The analog-chain BS.1770 safety limiter and meter are calibrated with reference to 100% modulation with 50µs preemphasis (75 kHz deviation) per section 5.9 of EBU Tech 3344 ("Practical guidelines for distribution systems in accordance with EBU R 128").

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.



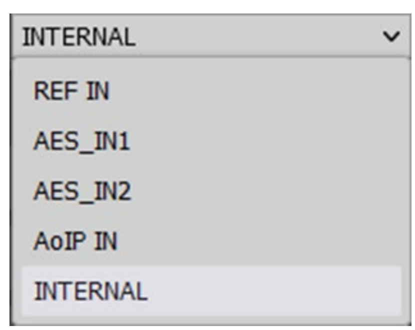
PROCESSING PRE-EMPHASIS: Sets the pre-emphasis that is required for your location. Most of the Americas use 75µs, much of Europe uses 50µs. If you are unsure, consult your local regulatory body.

FM POLARITY: Makes positive the output polarity of the TRIO-FM's FM analog channel processing. In HD Radio installations, this command is useful when switching the TRIO-FM 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.

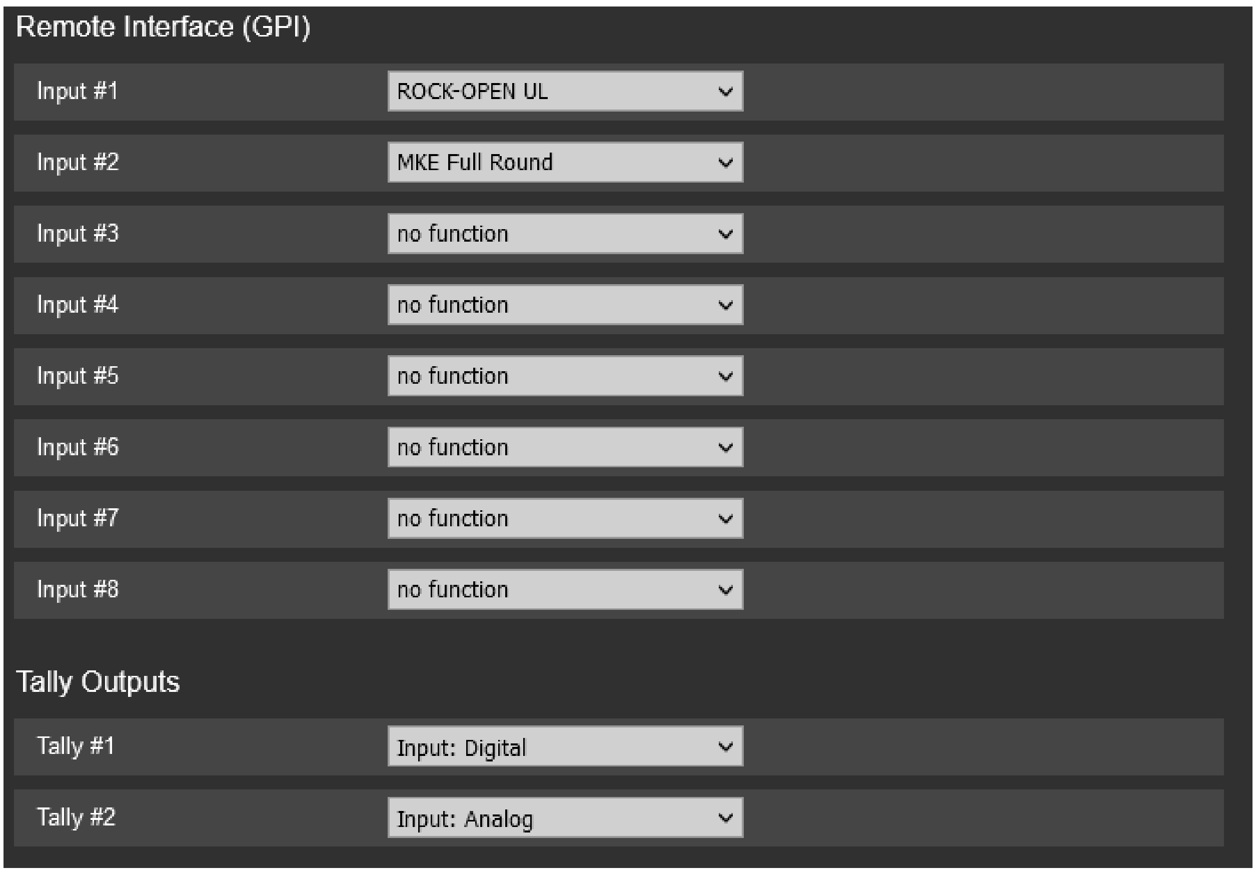
PHASE CORRECT: The phase correct option will "repair" asymmetrical voice audio so that it is more symmetrical and easier to process. Orban's recommended setting is on.

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 TRIO-FM will switch to an active analog input with audio present.

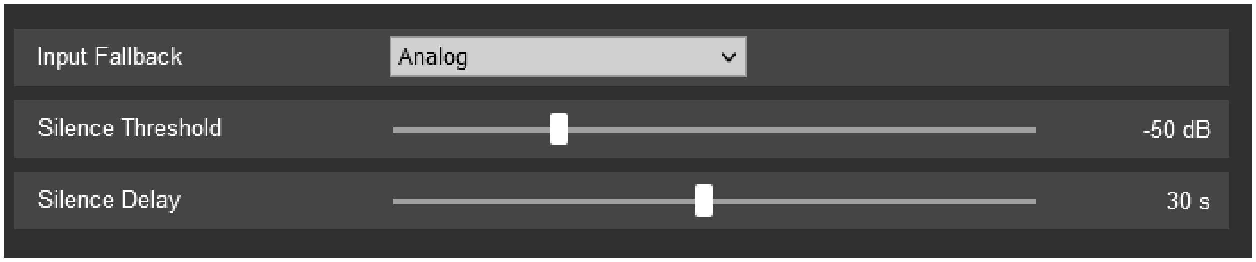
DIVERSITY DELAY MODE: Selects if the processor had a delay path in the FM or FM+HD audio path.

REMOTE INTERFACE



The GPI/GPO closures on the DB cable on the rear of the TRIO-FM 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 TRIO-FM 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 TRIO-FM 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 TRIO-FM 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

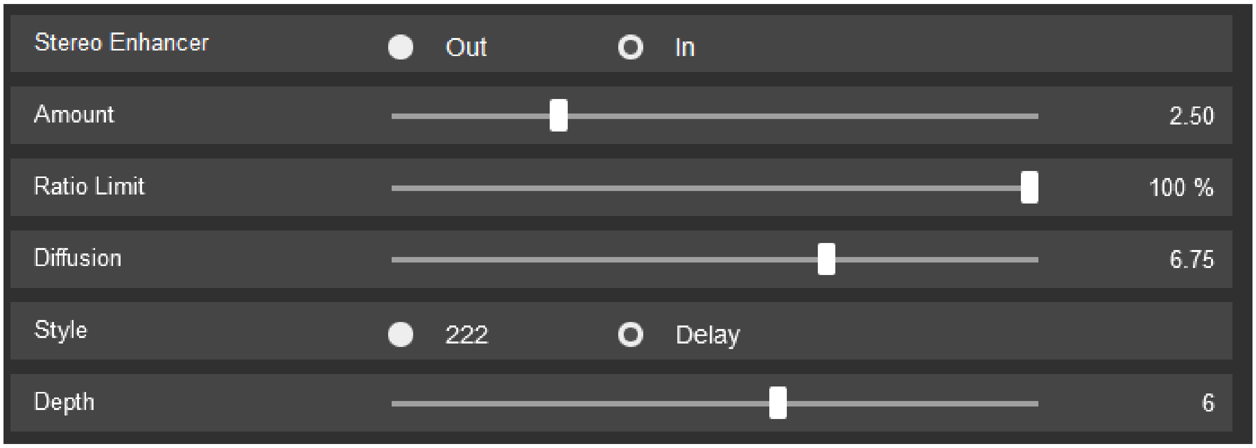
The processing parameters tab opens up a sidebar which allows the user the ability to fine tune each stage of the audio. The following section of this manual will walk you through the steps and help you understand what the various controls mean.

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 TRIO-FM 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 TRIO-FM, 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 TRIO-FM 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 TRIO-FM 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 TRIO-FM 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 TRIO-FM'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.

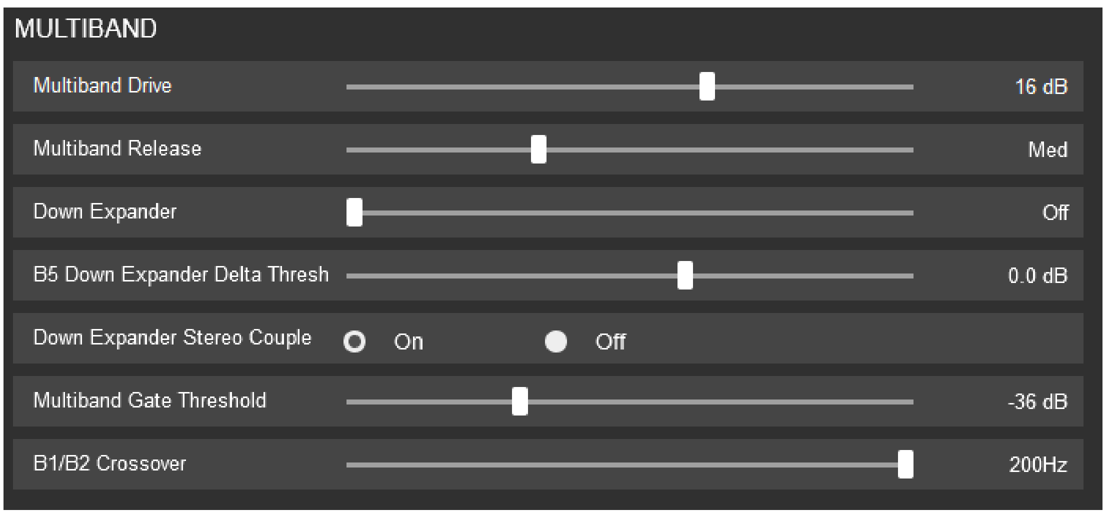
The TRIO-FM HF Enhancer is a program-controlled HF shelving equalizer.

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 TRIO-FM 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 TRIO-FM'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 TRIO-FM'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.

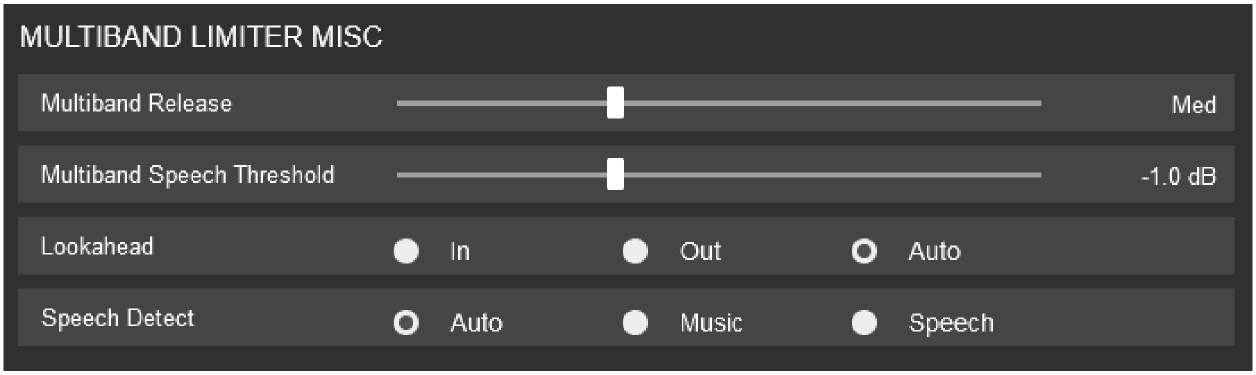
- **MULTIBAND ATTACK** – Sets the rate of speed with which the band you are setting will decrease gain at its input. Each Band (B1-B5) has it's own control. Longer attack times for bands 3 and 4 will be most transparent for horns and sustained vocals. Shorter attack times in Bands 1 and 5 will be more 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 TRIO-FM'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 TRIO-FM into Music mode, overriding the Speech/Music detector.

This control is contained in the processing preset.



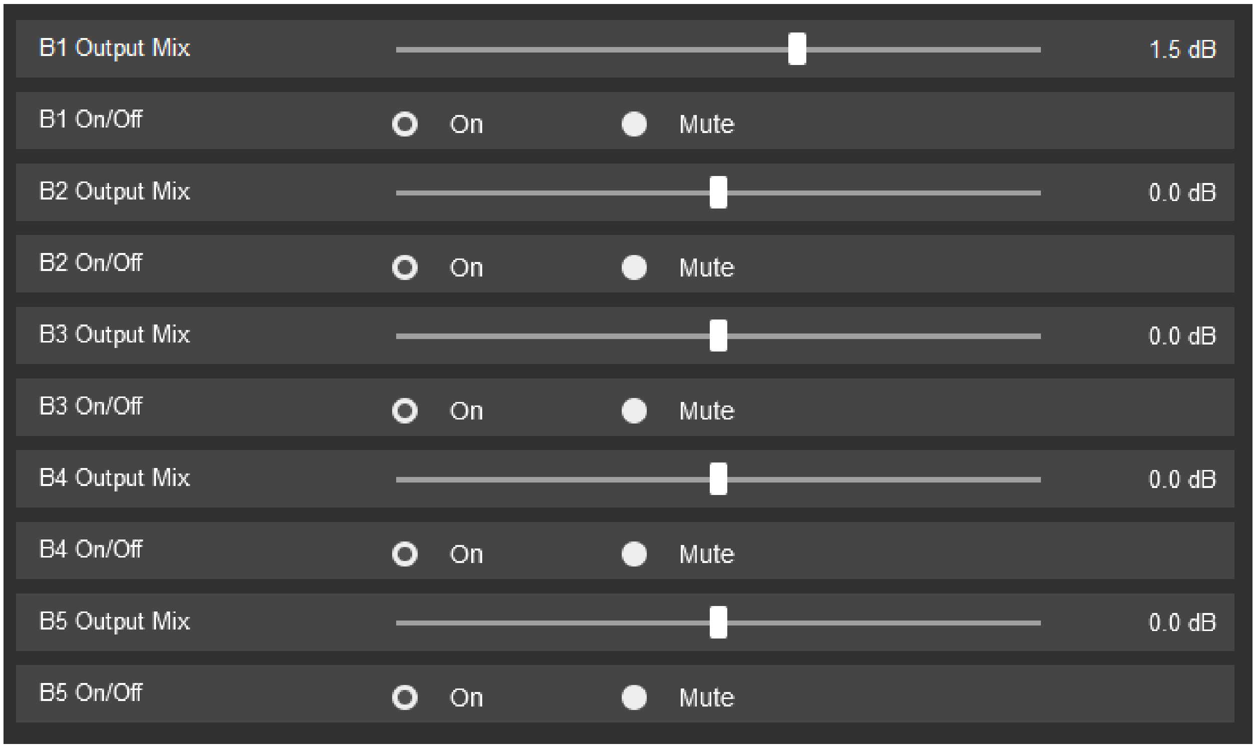
- **MULTIBAND ATTACK** – During speech, Sets the rate of speed with which the band you are setting will decrease gain at its input. Each Band (B1-B5) has it's own control.

- **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 TRIO-FM into MUSIC only, SPEECH only or AUTO detect mode.

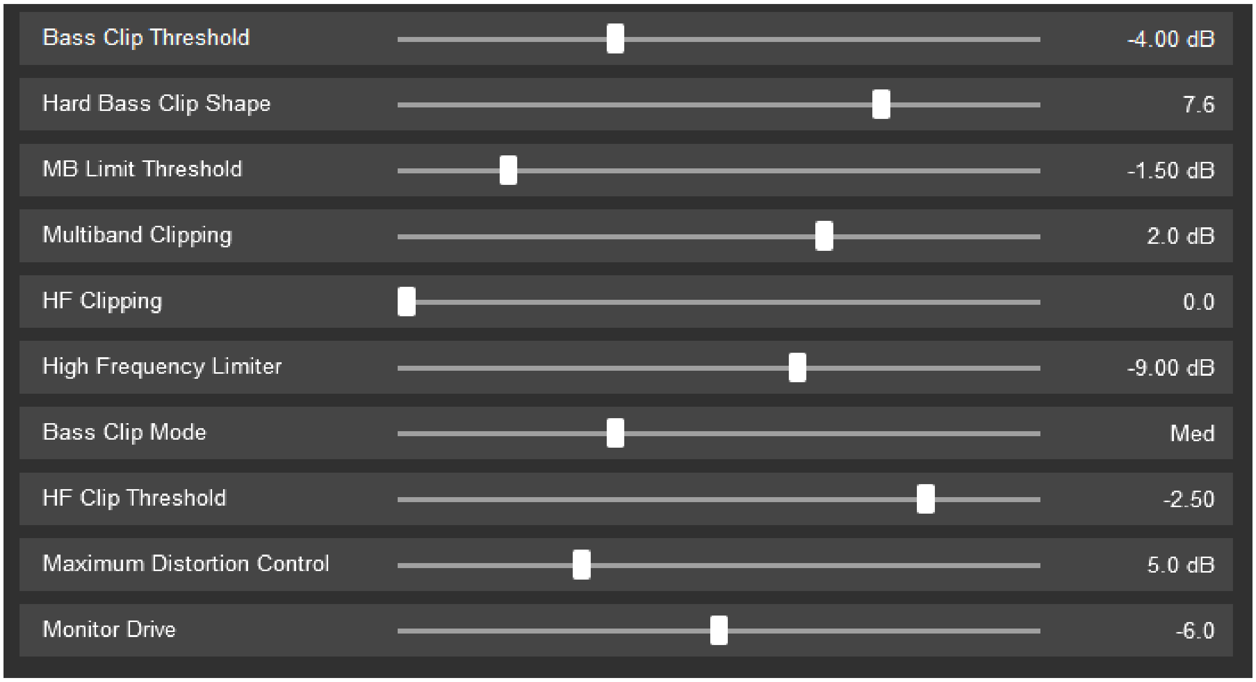
BANDMIX



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.

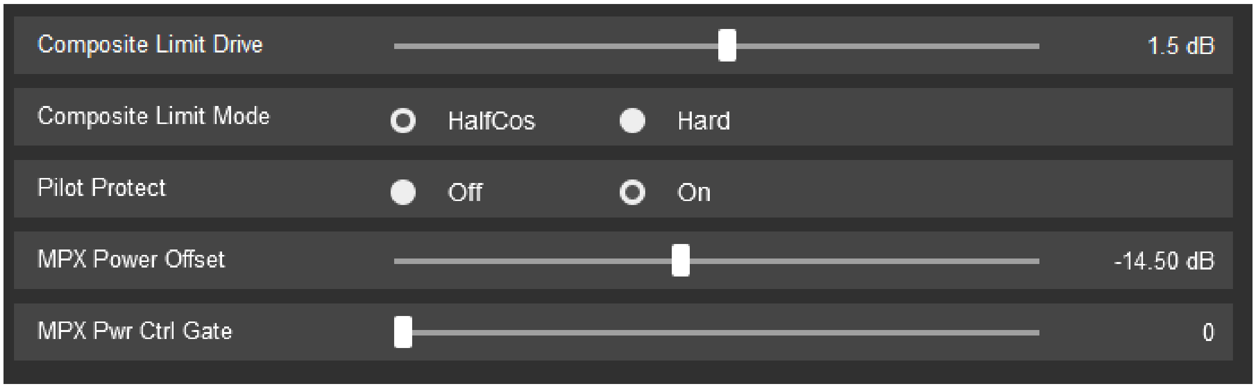
The ON/OFF control turns on and off the output of each band. Useful in some aspects, turning off a band of processing will unmask distortion from neighboring bands, especially with aggressive presets. Orban's intelligent clipper design prevents that type of distortion from being audible when all bands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



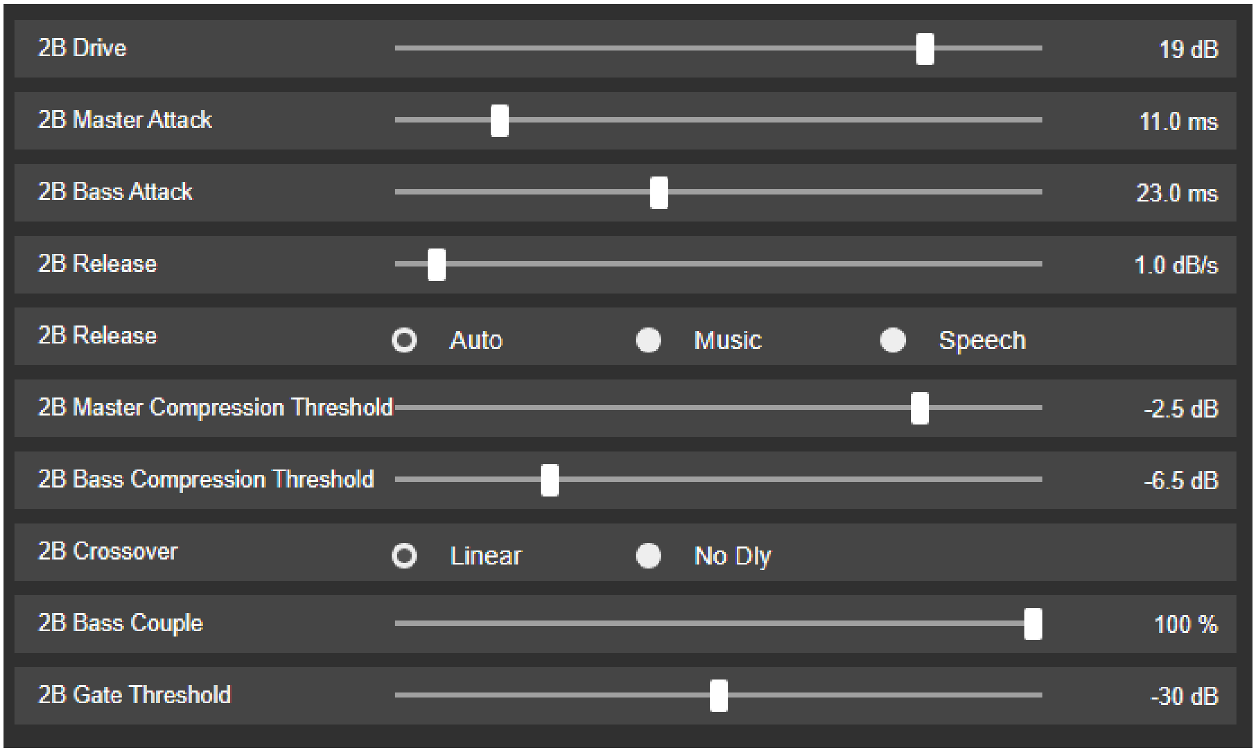
- **COMPOSITE LIMIT DRIVE –** Sets the amount of peak limiting in the composite limiter.
- **COMPOSITE LIMIT MODE –** Half cosine provides better separation and maintains stereo imaging. Hard mode provides a brighter sound as it creates waveforms closer to square waves.
- **PILOT PROTECT –** Protects 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.

2 BAND PROCESSING

In addition to five-band processing, suitable for pop music and talk formats, the TRIO-FM 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 TRIO-FM'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 TRIO-FM 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.

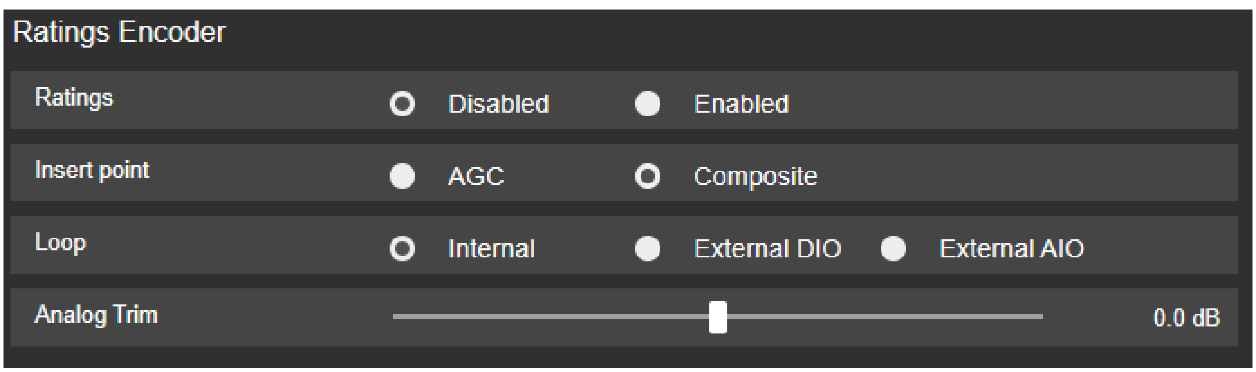
- **2B (Two Band) 6-15 kHz HF LIMITER –** Sets the amount of extra gain reduction in the top band of the multiband high frequency limiter.

- **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 TRIO-FM 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 Trio. 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.

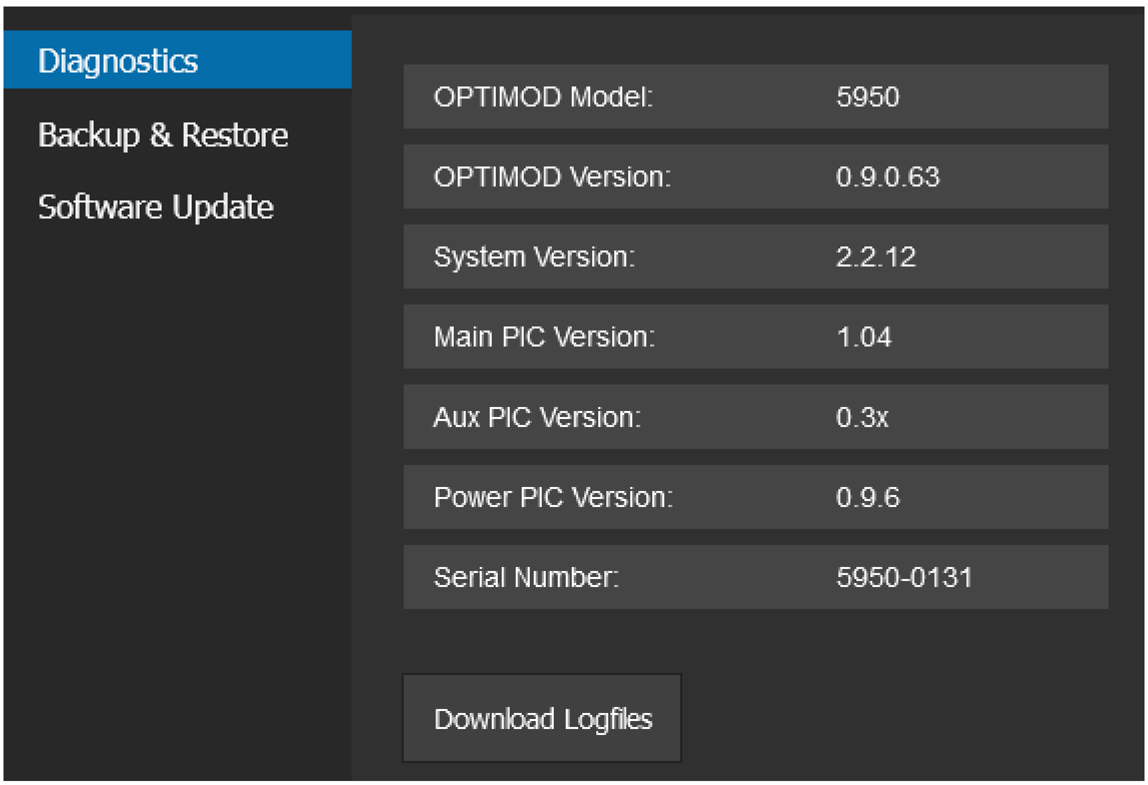
To insert an analog ratings encoder, you will need to connect the analog L/R output to your encoder. Next, take the encoder's output and insert it into the ANALOG IN on the rear of the Trio. Once complete, you will set ratings to ENABLE, choose where you want the encoding inserted (either after the AGC 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.

SYSTEM TAB

The system tab opens up three options

- Diagnostics
- Backup and Restore
- Software Update

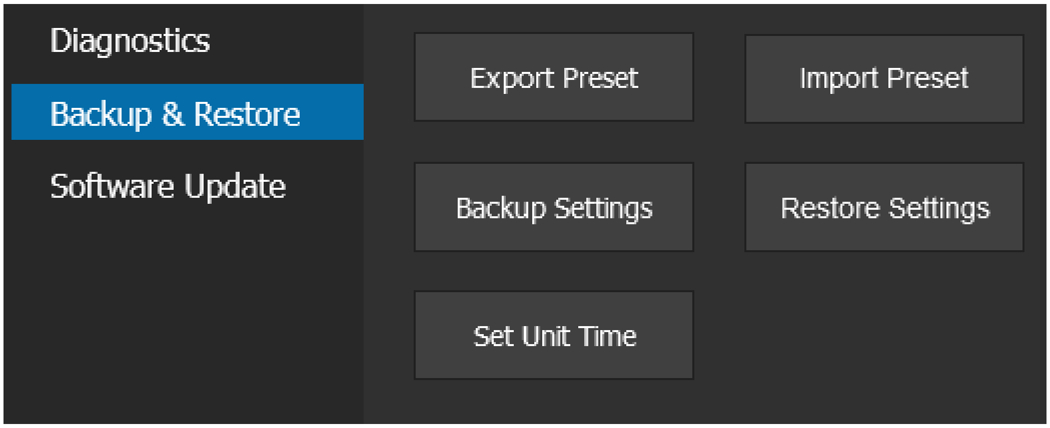
DIAGNOSTICS



The DIAGNOSTICS screen is useful in that it shows the current version running on the Trio, as well as other information that will be useful should you need to contact technical support with an issue.

You can also download the logfiles from this screen. This is yet another tool that is useful to technical support should you have an issue.

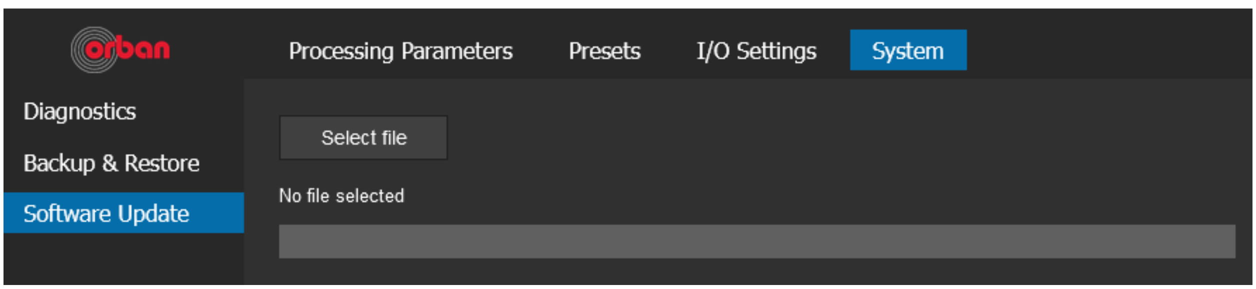
BACKUP AND RESTORE



Backup and restore allows you to inport and export presets from/to another Trio. It also allows you to back up and restore your system settings.

Another option allows you to set the system time. As of this version of the manual, that option is not available and system time must be set from the front panel.

UPDATING THE TRIO-FM



Under the SYSTEM tab, you will find an option to update the software on your TRIO-FM (SOFTWARE UPDATE). Choosing SELECT FILE will open a dialogue box where you can navigate to the downloaded update.

Once the file is selected, the status bar will show the process of the update being uploaded to the TRIO-FM.

Specifications

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 Trio 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

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, or the low delay monitor signal.

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.

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.

Section 4: TRIO DM

Trio DM is a mode of the OPTIMOD Trio that is in development at this time. It is scheduled for the 4th quarter of 2024.

Section 5: Maintenance

Routine Maintenance

The Trio 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 Trio's output.

A good ear will pick up many faults. Familiarize yourself with the “sound” of the Trio 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 Trio is at fault. The troubleshooting information in Section 5 will help you determine if the problem is with THE OPTIMOD-Trio 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 Trio 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 6: Introduction

Problems and Potential Solutions

Always verify that the problem is not the source material being fed to the Trio, 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 Trio 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 Trio 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 Trio may be in stand-alone stereo encoder mode. The active on-air preset determines this.

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

In AM mode, the Trio may be processing symmetrical audio.

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 Trio normally controls peak modulation to an accuracy of ±2%. This accuracy will be destroyed if the signal path following the Trio 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 Trio'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 Trio'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 Trio has higher dynamic range than older designs. Therefore, without compromising the Trio’s noise level, we could eliminate a control that was frequently misadjusted.

If you are using the Trio’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 Trio’s AGC. In this case, reduce the setting of the AGC’s MAXDELTAGR controls.

If you are using an external processor ahead of the Trio, 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 Trio 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 Trio'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 Trio to digitize the input. (This ratio is better than 108 dB.) It is important to drive the Trio with professional levels (more than 0 dBu reference level) to achieve adequately low noise. (Clipping occurs at +27 dBu.)

The Trio'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 Trio'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 Trio, the STL's noise level can severely limit the overall noise performance of the system because compression in the Trio 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 Trio 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 Trio and the rest of the transmission system are operating correctly, subcarriers should experience no interference.

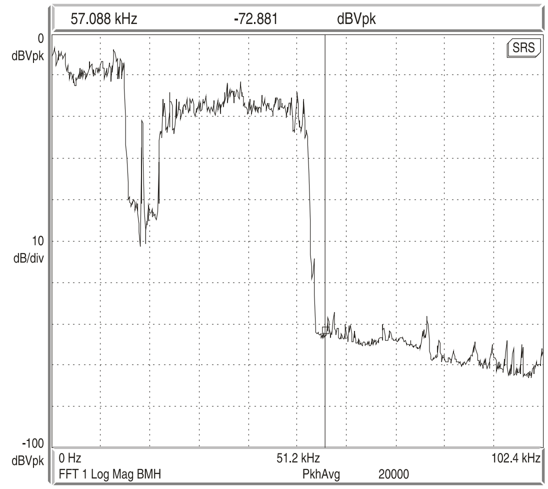


Figure 5-1: Typical Trio FM baseband spectrum with heavy processing, 0-100 kHz.

Interference from the stereo into a subcarrier is best diagnosed with a spectrum analyzer. First examine the spectrum of the Trio'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 Trio'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 Trio'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 Trio 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 Trio'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 Trio can generate test tones itself. The Trio 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 Trio, or the Trio should be temporarily switched to BYPASS to pass the tones.

System Receiving Trio's Digital Output Will Not Lock

Be sure that the Trio's output sample rate is set match the sample rate that the driven system expects. Be sure that the Trio'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 Trio'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.

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.

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.

General Dissatisfaction with Subjective Sound Quality

The Trio 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 Trio 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 Trio, 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 Trio. 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 Trio that we may post from time to time. Manuals (in .pdf form) and Trio 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 7: Introduction

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-Trio'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-Trio'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.

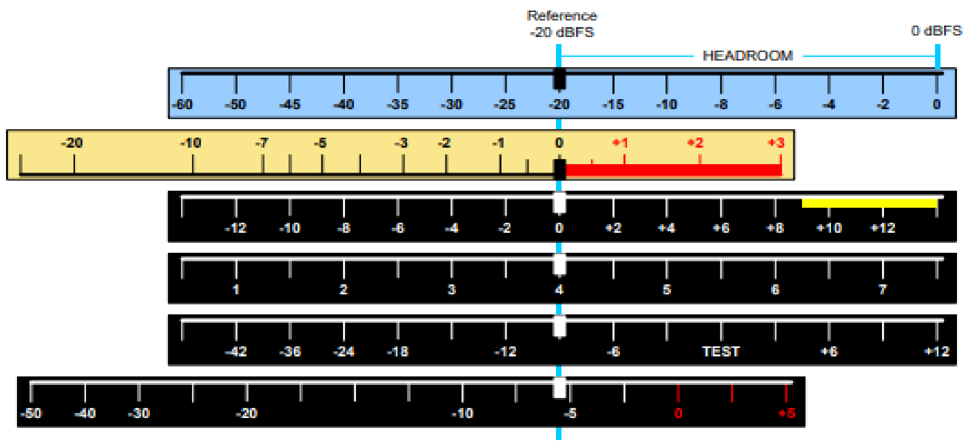
Studio Line-up Levels and Headroom

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.

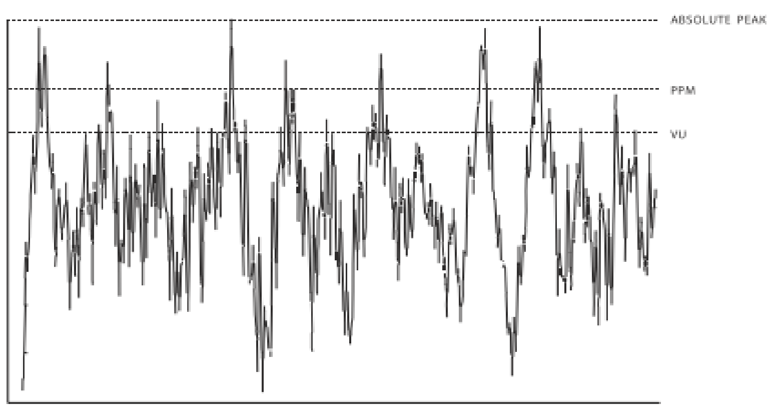
For line-up, the studio engineer uses a calibration tone at a studio standard level, commonly called line-up level, reference level, or operating level. Metering at the studio is by a VU meter or PPM (Peak Program Meter). As discussed above, the VU or PPM indication under-indicates the true peak level.

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 Trio 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 Trio can accept. If you are using the AES3 digital input, a full-scale digital word corresponds to the 0 dB point on the Trio's input meter.

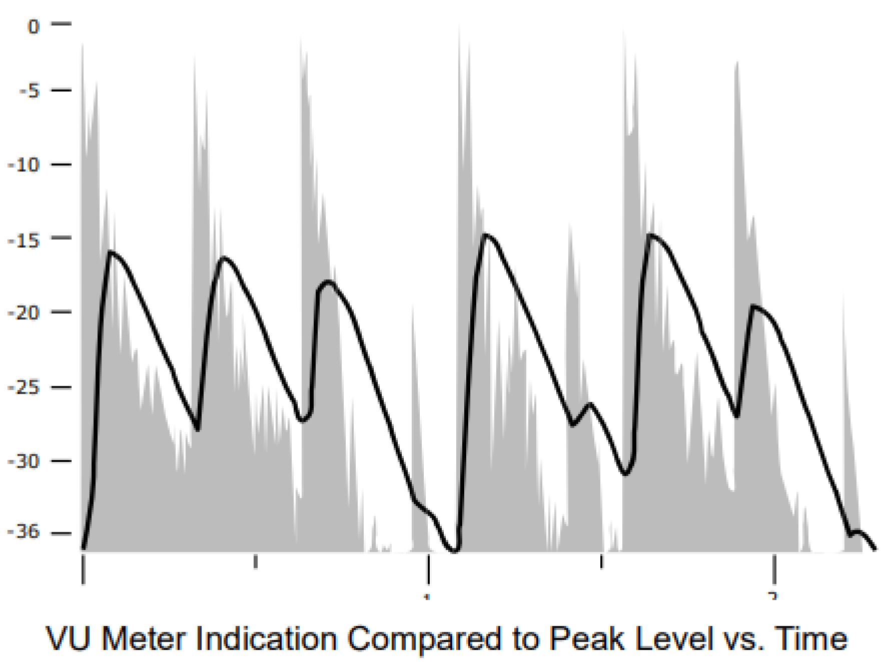


Figure 6-3: Metering of Levels and Subjective Loudness

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 Trio 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 Trio'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

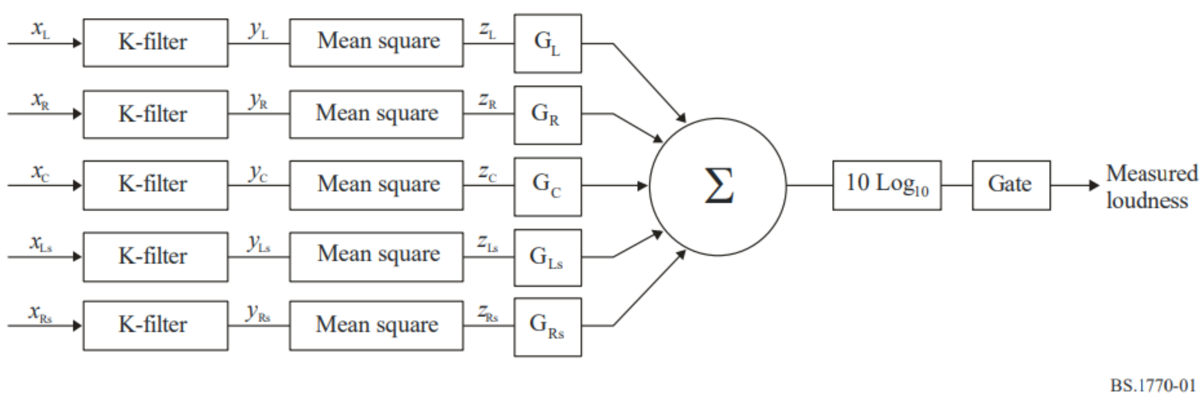


Figure 6-4: BS.1770 Loudness Meter Block Diagram (from ITU-R document)

The subjective loudness meters, labeled LOUDNESS in the Trio'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 Trio's digital radio processing chain is adjusted to make the dominant program material indicate "0 dB" on the Trio'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 Trio to the transmission system that it is driving, the Trio contains an adjustable test tone oscillator that produces sine waves at Trio'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 Trio's left/right analog output is switched to FLAT, a de-emphasis filter is inserted between output of the Trio's audio processing and its line output. Thus, as the frequency of the Test Tone is changed, the level at the Trio'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 Trio's internal de-emphasis, so the Trio'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 Trio'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 Trio's MPX POWER meter is therefore calibrated so that it indicates 0 dB when the composite output of the Trio 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 Trio'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 Trio. Such overshoots necessitate turning down the Trio's output level control after it has been calibrated with tone using an FM modulation meter and the Trio'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 Trio's output level control was turned down.

See ***Error! Reference source not found.*** starting on page 1-2-10 for a discussion of overshoots and how they force the average modulation to be reduced to prevent peak overmodulation of the FM carrier.

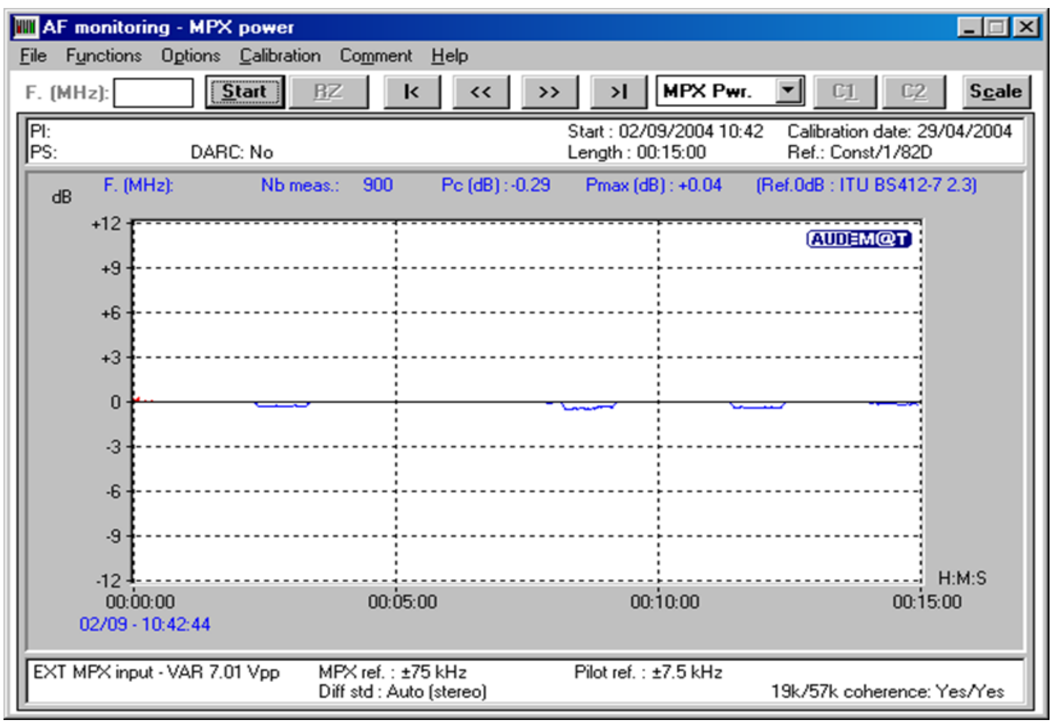
Because the Trio does not digitize subcarriers applied to its subcarrier inputs, the Trio's MPX POWER meter (which operates in the DSP domain) cannot indicate the power added by such subcarriers. These usually have constant power, so it is easy to compensate for them. For example, if an FM subcarrier 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 Trio 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 Trio 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.

If your country enforces the standard, you should set the control to complement the amount of peak overshoot in the transmission system following the Trio. Setting the control at "0" will correctly control the multiplex power when there is no overshoot after the Trio. 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 Trio. For example, if RF peak deviation exceeds the peak deviation produced by the Trio'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 Trio 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 Trio'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 Trio'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 Trio'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.

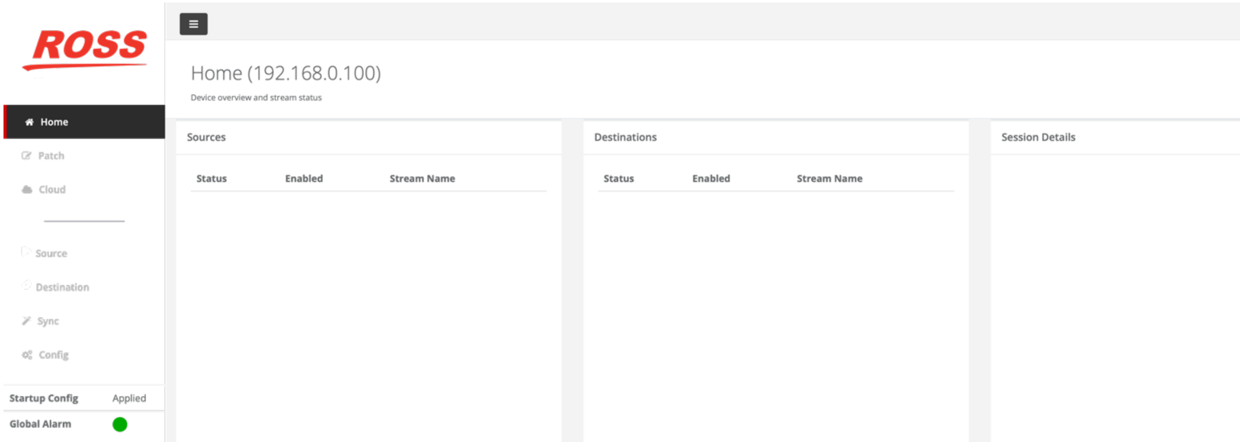
Optimod Trio AoIP Ross Configuration Guide

Connecting to the AoIP management interface

The 5950's AoIP features can be managed via a web interface. The default IP address of this web interface is 192.168.0.100. To access this interface, connect either of the AoIP ethernet ports on the back of the 5950 to a network switch. Now on the client computer change the network configuraGon to be on the same subnetwork, for example:

```
IP Address      192.168.0.50
Subnet Mask     255.255.255.0
Gateway         192.168.0.1
```

Using your web browser go to hLp://192.168.0.100 and the web interface should be displayed.



Navigating the AoIP web interface

This web interface can take a couple of seconds to navigate between tabs. Please allow 3-5 seconds for each tab to fully populate before interacGng with the controls.

Configuring the network interfaces

The first thing the user may want to do is to customize the network interface for easier access. This can be achieved in the Config tab:

75) Scroll to the boLom and select the PRIMARY interface.

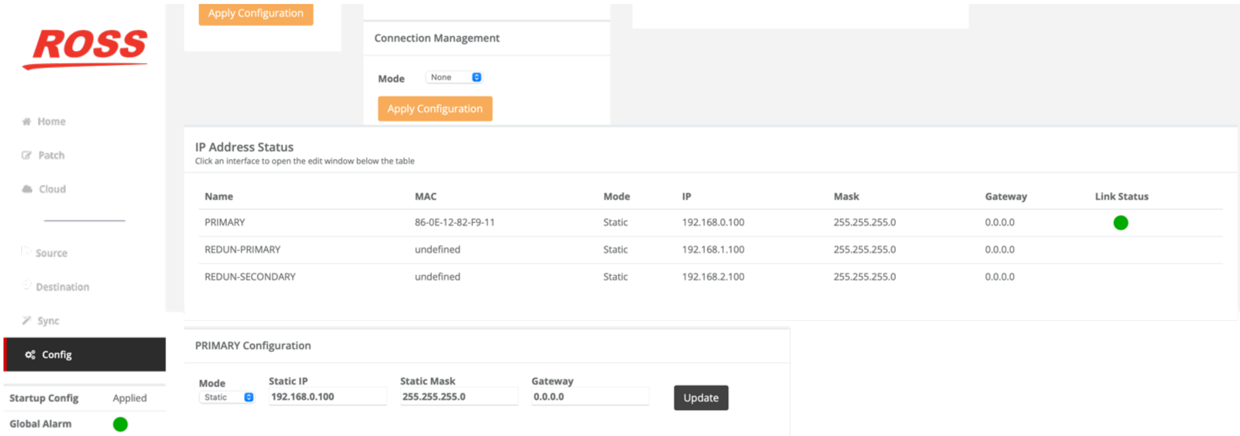
76) AUer selecGng this interface, you may need to scroll a bit more to view the current configuraGon.

77) Type your desired staGc IP or select dynamic on the dropdown menu to use DHCP.

78) Important: Take note of the new IP address as for the Gme being the 5950 does not report this address via its web or front panel interfaces.

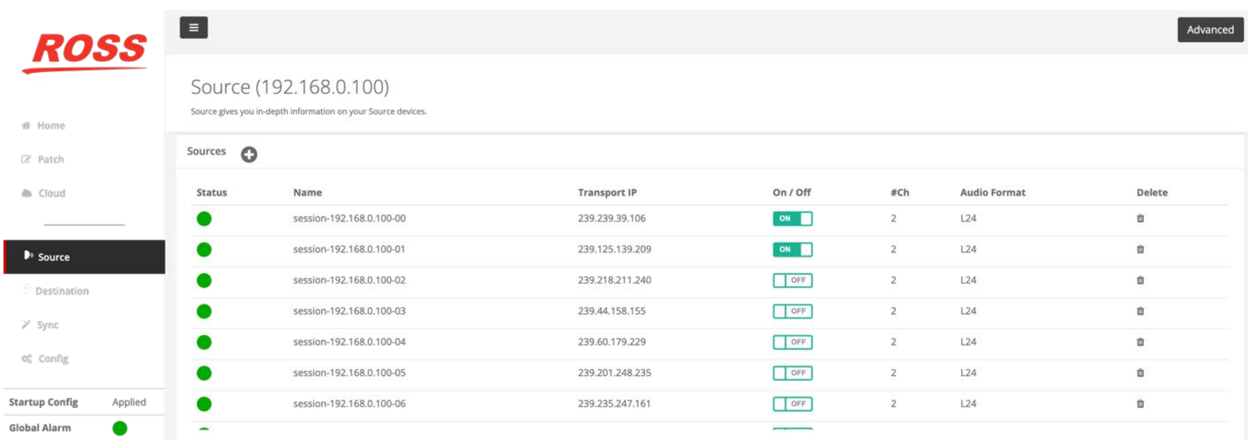
79) Select Update

80) AUer this you will be asked to reboot the unit for the changes to take effect.

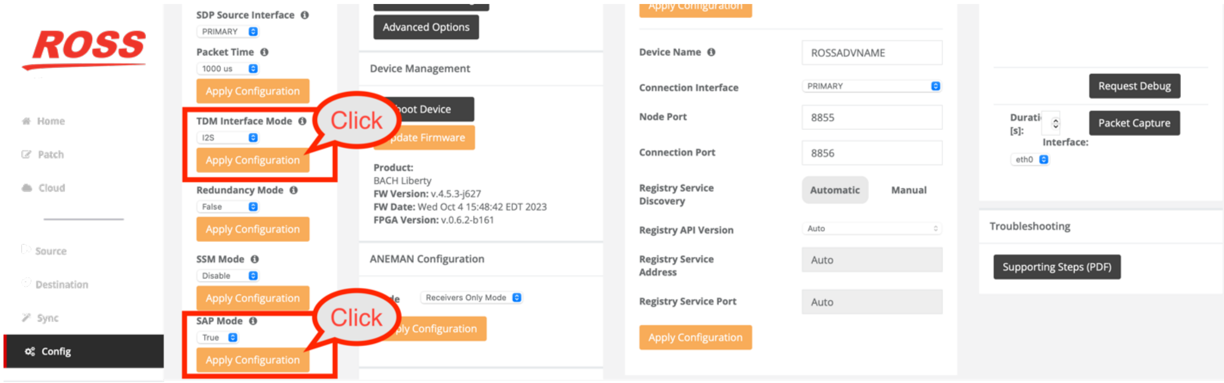


Configuring the AoIP outputs

The 5950 currently supports 2 AoIP outputs (4 will be supported in a future update). By default, these outputs are turned off. To acGvate them go to the Source tab and turn on the first 2 sources.

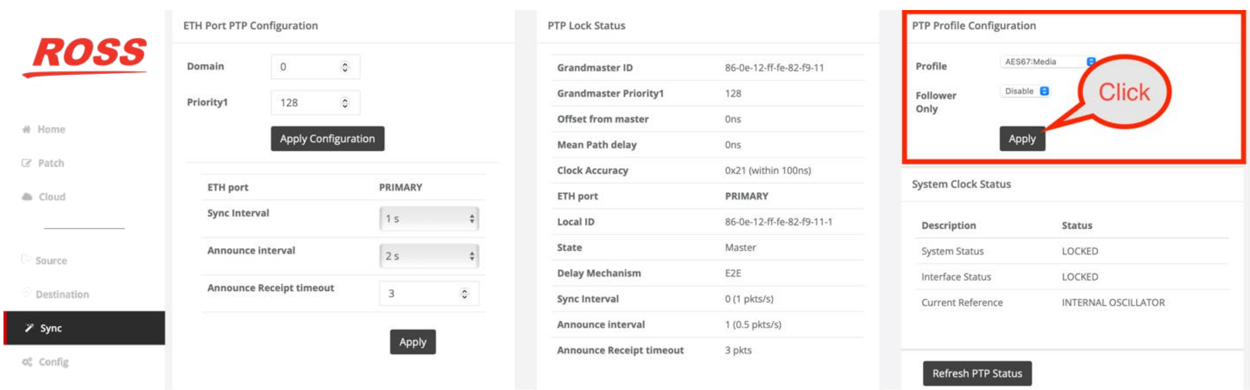


Now navigate to the *Config* tab and set *TDM interface mode* to *I2S* and *SAP* to *True*. Hit apply configuraGon on EACH se_ng aUer updaGng it.



Set up the PTP profile

Go to the *Sync* tab and change the *PTP Profile configura=on* to *AES67:Media* and click Apply.



Configuring the AoIP Inputs

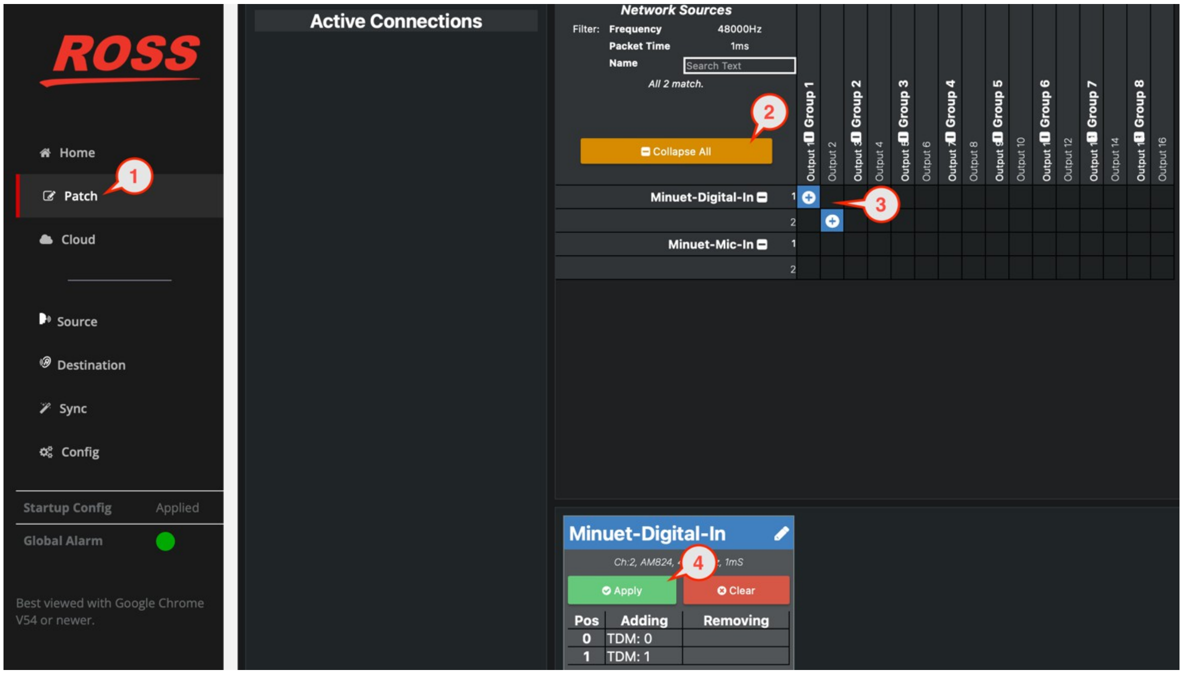
81) Go to the Patch tab and you should see a list of compaGble devices in your network.

82) Expand all Network Sources

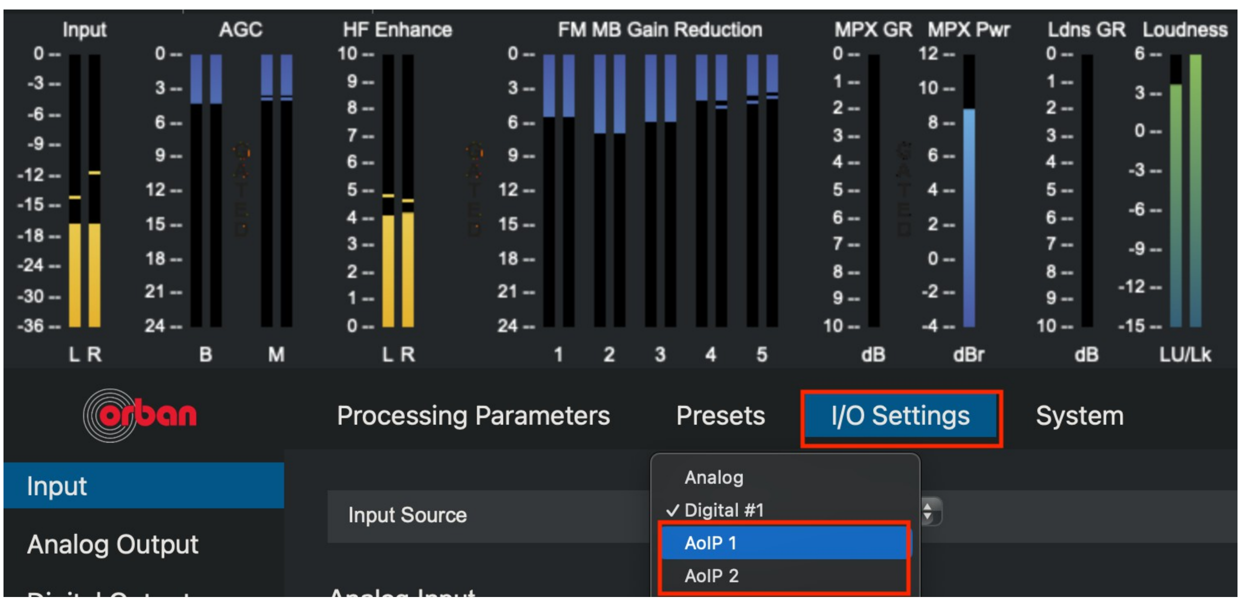
83) Connect your source to either Group 1 (AoIP Input 1) or Group 2 (AoIP Input 2)

84) Select Apply.

85) AUer this you should see your device in the AcGve connecGons secGon.



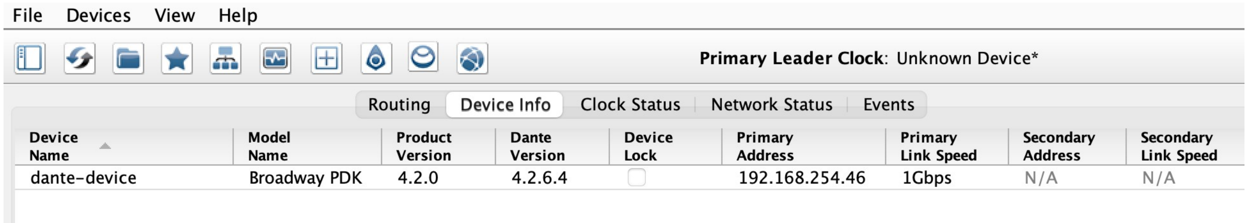
Finally on the 5950 web GUI change your input to AoIP 1 or 2 under I/O se_ngs.



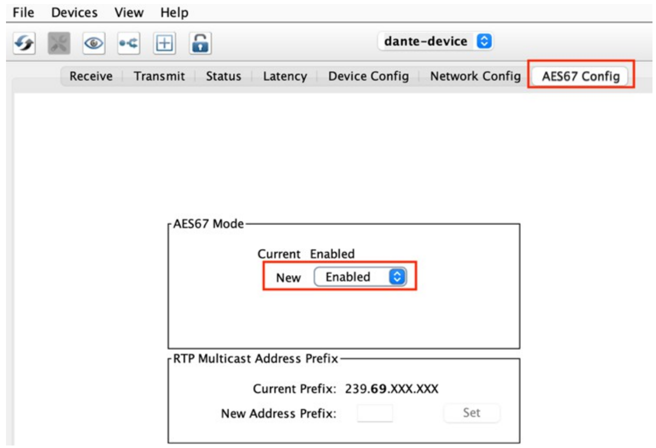
(Optional) Configuring a Dante Source

Some, but not all Dante devices support AES67. In the following example we'll be se_ng a up Dante Broadway device. For this step we'll be using Dante Controller (hLps://my.audinate.com/support/downloads/dante-controller).

First make sure your Dante device has AES67 support enabled. Open Dante controller and find your device listed on the Device Info tab.



Double click on your device and navigate to the AES67 Config tab and make sure AES67 Mode is enabled. Reboot the device if prompted.



Next we'll set up the source for our input:

86) Go to the Transmit tab

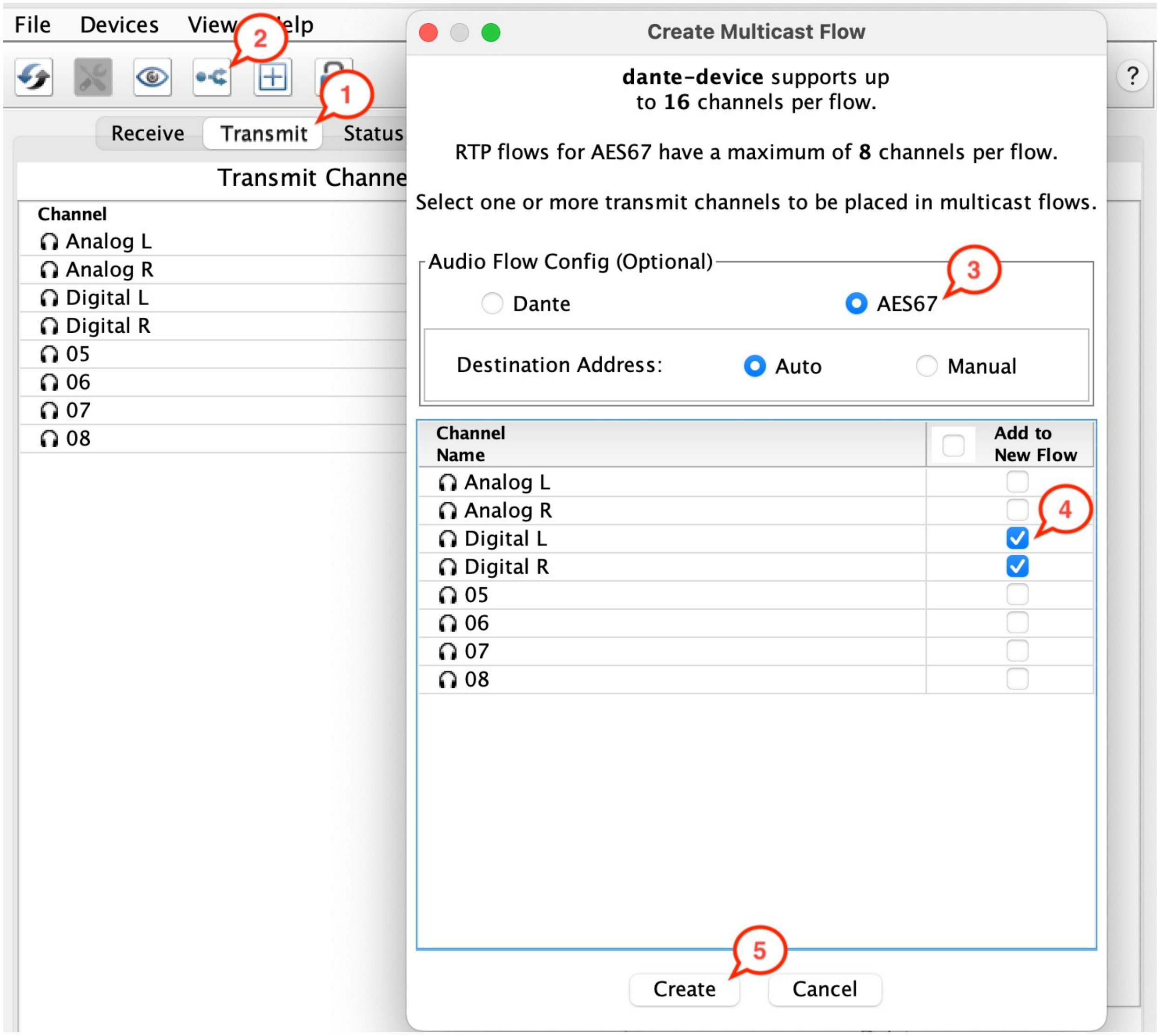
87) Click on the MulGcast Flow buLon.

88) Select AES67 ConfiguraGon

89) Select the channels you'd like in your source

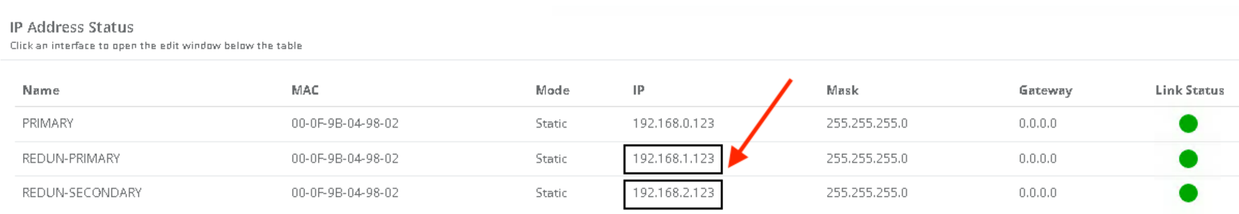
90) Click Create

AUer this step, the Dante device should be listed in the Patch tab of the 5950's AoIP web interface and you may connect it as an input source.



(Optional) Configuring the redundant AoIP network

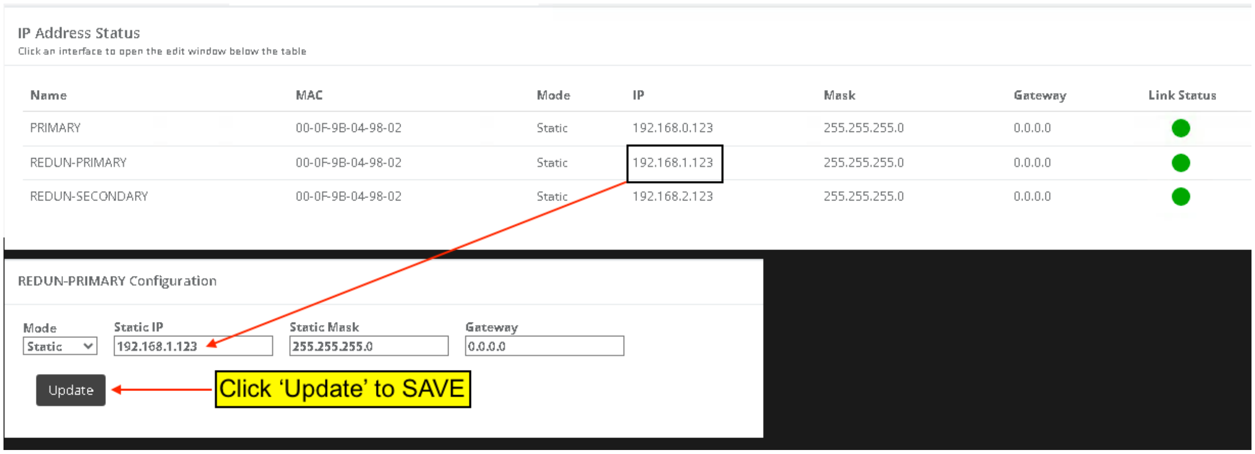
Configure the redundant IP subnets for each primary and secondary LAN. It is recommended that each LAN run independently on discrete managed switches. Our example listed below with the redundant primary LAN 192.168.1.x and redundant secondary 192.168.2.x


IP Address Status

Click an interface to open the edit window below the table

Name	MAC	Mode	IP	Mask	Gateway	Link Status
PRIMARY	00-0F-9B-04-98-02	Static	192.168.0.123	255.255.255.0	0.0.0.0	
REDUN-PRIMARY	00-0F-9B-04-98-02	Static	192.168.1.123	255.255.255.0	0.0.0.0	
REDUN-SECONDARY	00-0F-9B-04-98-02	Static	192.168.2.123	255.255.255.0	0.0.0.0	

To edit the StaGc IP address for each redundant LAN click on either IP listed above to open up the edit box below.

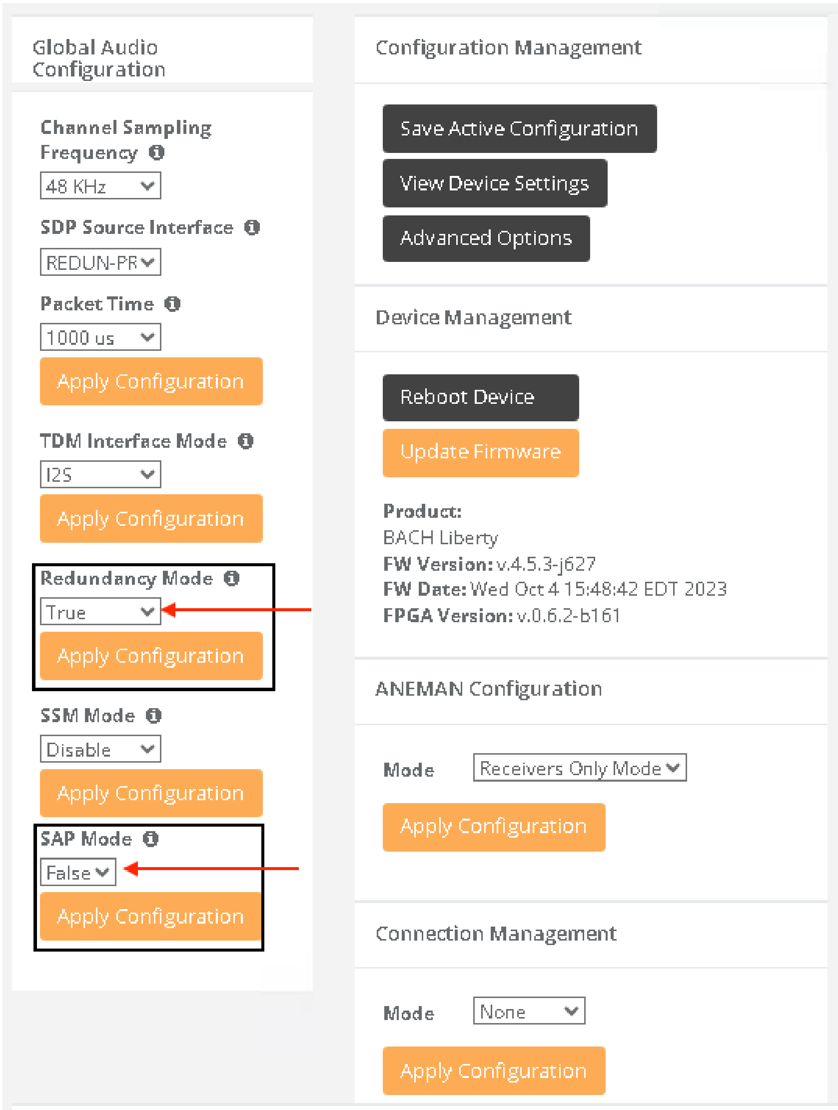


Enter the staGc IP here or set the mode to Dynamic for DHCP networks. Click UPDATE once completed to save.

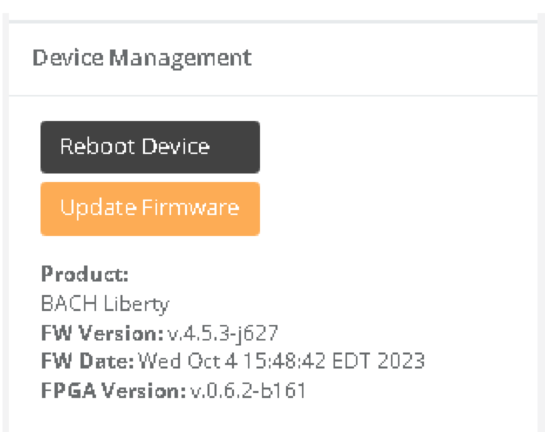
Next begin se_ng up the Global Audio ConfiguraGon found within the Device ConfiguraGon tab.

91) Redundancy Mode – set this to TRUE and click Apply ConfiguraGon. **It should be noted that enabling this mode will disable the 'Primary' LAN and now use the 'Redundant Primary' LAN as the primary network.** Moving the IP subnet from Primary to Redundant Primary will result in the webserver to lose communicaGon. Enter the new IP address for the Redundant Primary LAN in the URL to gain control of the webserver again.

92) SAP Mode – set this to False and click Apply ConfiguraGon.



It is necessary to reboot the Bach Liberty module aUer this step aUer to load in the new network se_ngs.

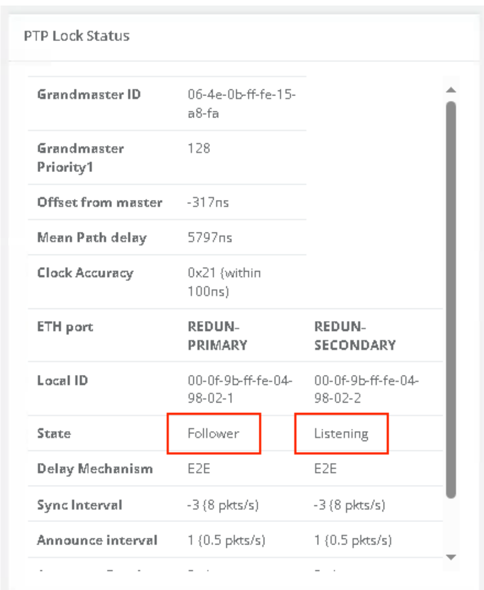


Once rebooted the Link Status for each LAN (Primary, Redundant Primary, and Redundant Secondary) should have a GREEN light. **Note: reboo?ng the Bach Liberty can take up to 2 minutes.**

Web UI will now be available through either the Primary or Secondary redundant IP addresses but not the primary non-redundant IP address.

Make physical connecGons to the rear of the OpGmod – RJ45 ports (AoIP 1 & AoIP 2) from discrete subnets arriving from separate switches. The redundant LANs must have PTP lock.

Loss of audio on either LAN will automaGcally switch to the other. In our example, we can see within the PTP Lock Status secGon of the Sync tab that the state of our redundant primary LAN is FOLLOWING and the redundant secondary LAN is LISTENING. Loss of audio on the redundant primary LAN will seamlessly switch over to the redundant secondary LAN. The PTP Lock Status should now display the redundant secondary as FOLLOWING and redundant primary as LISTENING.



PTP Lock Status

Grandmaster ID	06-4e-0b-ff-fe-15-a8-fa	
Grandmaster Priority1	128	
Offset from master	-317ns	
Mean Path delay	5797ns	
Clock Accuracy	0x21 (within 100ns)	
ETH port	REDUN-PRIMARY	REDUN-SECONDARY
Local ID	00-0f-9b-ff-fe-04-98-02-1	00-0f-9b-ff-fe-04-98-02-2
State	Follower	Listening
Delay Mechanism	E2E	E2E
Sync Interval	-3 (8 pkts/s)	-3 (8 pkts/s)
Announce interval	1 (0.5 pkts/s)	1 (0.5 pkts/s)

Note: *When performing network maintenance, it is recommended to disable redundancy and run from Primary LAN.*

Changing Modes with OPTIMOD Trio

At first power on of your new OPTIMOD Trio, you will be asked to choose which processing mode i.e., AM (MW) or FM, you want your Trio to operate as, followed by a confirmation question. This selected processing mode will remain intact during operation and any subsequent reboot.

1) From the front panel, tap MAIN MENU on the lower right corner.
2) Go to SETUP
3) Choose FACTORY DEFAULTS
4) You will be asked to keep user presets or not. Choose which option you want.
5) Next it will ask you if you want to change modes during this reset.
6) The Trio will ask you to confirm.

7) The Trio screen will go white as it reconfigures.
8) You will be prompted as to which mode you want the Trio to boot to. Choose the new mode you wish.
9) Double check your network settings, they will default to factory settings. Treat the new mode as if this is the first time you are using the Trio.