Operating Manual

OPTIMOD-FM

8600

Digital Audio Processor

Version 4.3 Software

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IMPORTANT NOTE: Refer to the unit's rear panel for your Model Number.

Model Number:	Description:
8600 8600J 8600X 8600XJ	OPTIMOD 8600, Stereo Encoder, Digital I/O, Protection Structure, Two-Band Structure, Multi-Band Structure, HD Radio™ / Digital Radio / Netcast Processing,115V (for 100-130V operation) or 230V (for 200-250V operation), switchable to 50µs or 75µs. 8600J and 8600FMJ are for 90-117V operation. 8600X has a blank front panel and must be controlled via 8600 PC Remote software.
8600FM 8600FMJ	As above, except HD Radio™ / Digital Radio / Netcast Processing omitted. Upgradeable to 8600. 8600FMJ for 90-117V operation

CAUTION

RISK OF ELECTRIC SHOCK
DO NOT OPEN

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

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

Safety Instructions (German)

Gerät nur an der am Leistungsschild vermerkten Spannung und Stromart betreiben.
Sicherungen nur durch solche, gleicher Stromstärke und gleichen Abschaltverhaltens ersetzen. Sicherungen nie überbrücken.
Jedwede Beschädigung des Netzkabels vermeiden. Netzkabel nicht knicken oder quetschen. Beim Abziehen des Netzkabels den Stecker und nicht das Kabel enfassen. Beschädigte Netzkabel sofort auswechseln.
Gerät und Netzkabel keinen übertriebenen mechanischen Beaspruchungen aussetzen.
Um Berührung gefährlicher elektrischer Spannungen zu vermeiden, darf das Gerät nicht geöffnet werden. Im Fall von Betriebsstörungen darf das Gerät nur Von befugten Servicestellen instandgesetzt werden. Im Gerät befinden sich keine, durch den Benutzer reparierbare Teile.
Zur Vermeidung von elektrischen Schlägen und Feuer ist das Gerät vor Nässe zu schützen. Eindringen von Feuchtigkeit und Flüssigkeiten in das Gerät vermeiden.
Bei Betriebsstörungen bzw. nach Eindringen von Flüssigkeiten oder anderen Gegenständen, das Gerät sofort vom Netz trennen und eine qualifizierte Servicestelle kontaktieren.

Safety Instructions (French)

On s'assurera toujours que la tension et la nature du courant utilisé correspondent bien à ceux indiqués sur la plaque de l'appareil.
N'utiliser que des fusibles de même intensité et du même principe de mise hors circuit que les fusibles d'origine. Ne jamais shunter les fusibles.
Eviter tout ce qui risque d'endommager le câble seceur. On ne devra ni le plier, ni l'aplatir. Lorsqu'on débranche l'appareil, tirer la fiche et non le câble. Si un câble est endommagé, le remplacer immédiatement.
Ne jamais exposer l'appareil ou le câble ä une contrainte mécanique excessive.
Pour éviter tout contact averc une tension électrique dangereuse, on n'oouvrira jamais l'appareil. En cas de dysfonctionnement, l'appareil ne peut être réparé que dans un atelier autorisé. Aucun élément de cet appareil ne peut être réparé par l'utilisateur.
Pour éviter les risques de décharge électrique et d'incendie, protéger l'appareil de l'humidité. Eviter toute pénétration d'humidité ou fr liquide dans l'appareil.
En cas de dysfonctionnement ou si un liquide ou tout autre objet a pénétré dans l'appareil couper aussitôt l'appareil de son alimentation et s'adresser à un point de service aprésvente autorisé.

Safety Instructions (Spanish)

Hacer funcionar el aparato sólo con la tensión y clase de corriente señaladas en la placa indicadora de características.
Reemplazar los fusibles sólo por otros de la misma intensidad de corriente y sistema de desconexión. No poner nunca los fusibles en puente.
Proteger el cable de alimentación contra toda clase de daños. No doblar o apretar el cable. Al desenchufar, asir el enchufe y no el cable. Sustituir inmediatamente cables dañados.
No someter el aparato y el cable de alimentación a esfuerzo mecánico excesivo.
Para evitar el contacto con tensiones eléctricas peligrosas, el aparato no debe abrirse. En caso de producirse fallos de funcionamiento, debe ser reparado sólo por talleres de servicio autorizados. En el aparato no se encuentra ninguna pieza que pudiera ser reparada por el usuario.
Para evitar descargas eléctricas e incendios, el aparato debe protegerse contra la humedad, impidiendo que penetren ésta o líquidos en el mismo.
En caso de producirse fallas de funcionamiento como consecuencia de la penetración de líquidos u otros objetos en el aparato, hay que desconectarlo inmediatamente de la red y ponerse en contacto con un taller de servicio autorizado.

Safety Instructions (Italian)

Far funzionare l'apparecchio solo con la tensione e il tipo di corrente indicati sulla targa riportante i dati sulle prestazioni.
Sostituire i dispositivi di protezione (valvole, fusibili ecc.) solo con dispositivi aventi lo stesso amperaggio e lo stesso comportamento di interruzione. Non cavallottare mai i dispositivi di protezione.
Evitare qualsiasi danno al cavo di collegamento alla rete. Non piegare o schiacciare il cavo. Per staccare il cavo, tirare la presa e mai il cavo. Sostituire subito i cavi danneggiati.
Non esporre l'apparecchio e il cavo ad esagerate sollecitazioni meccaniche.
Per evitare il contatto con le tensioni elettriche pericolose, l'apparecchio non deve venir aperto. In caso di anomalie di funzionamento l'apparecchio deve venir riparato solo da centri di servizio autorizzati. Nell'apparecchio non si trovano parti che possano essere riparate dall'utente.
Per evitare scosse elettriche o incendi, l'apparecchio va protetto dall'umidità. Evitare che umidità o liquidi entrino nell'apparecchio.
In caso di anomalie di funzionamento rispettivamente dopo la penetrazione di liquidi o oggetti nell'apparecchio, staccare immediatamente l'apparecchio dalla rete e contattare un centro di servizio qualificato.

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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 $35.00).

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

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

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

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

Packing

When you pack the unit for shipping:

(1) Tighten all screws on any barrier strip(s) so the screws do not fall out from vibration.
(2) Wrap the unit in its original plastic bag to avoid abrading the paint.
(3) Seal the inner and outer cartons with tape.

If you are returning the unit permanently (for credit), be sure to enclose:

- The Manual(s)
- The Registration / Warranty Card
- The Line Cord
- All Miscellaneous Hardware (including the Rack Screws and Keys)
- The Extender Card (if applicable)
- The Monitor Rolloff Filter(s) (OPTIMOD-AM only)
- The COAX Connecting Cable (OPTIMOD-FM and OPTIMOD-TV only)

Your dealer may charge you for any missing items.

If you are returning a unit for repair, do not enclose any of the above items.

Further advice on proper packing and shipping is included in the Manual (see Table of Contents).

Trouble

If you have problems with installation or operation:

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

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

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

WARNING

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

WARNING

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

IMPORTANT

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

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

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

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This manual was published May 2018.

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Operating Manual

OPTIMOD-FM

8600

Digital Audio Processor

Version 4.3 Software

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Function	Description	Drawing	Page
Chassis	Circuit Board Locator and basic	Top view	6-37

Function	Description	Drawing	Page
	interconnections	(not to scale)	
Base Board	Glue logic; supports CPU board and RS-232 daughterboard. ***Contains:***	Parts Locator Drawing	6-38
	System Connections	Schematic 1 of 4	6-39
	CPU board interface	Schematic 2 of 4	6-40
	Power Supply Monitor	Schematic 3 of 4	6-41
	CPLD, General Purpose Interface, and Remotes	Schematic 4 of 4	6-42
CPU board	Control microprocessor. Services front panel, serial port, Ethernet, DSP board, and control board. Resides on base board. ***Contains:***	Parts Locator Drawing	6-43
	Ethernet	Schematic 1 of 5	6-44
	General Purpose Bus	Schematic 2 of 5	6-45
	Memory	Schematic 3 of 5	6-46
	Miscellaneous Functions	Schematic 4 of 5	6-47
	Power and Ground Distribution	Schematic 5 of 5	6-48
RS-232 Board	Supports Serial Port	Parts Locator Drawing	6-49
		Schematic 1 of 1	6-50
Power Supply	±15V analog supply; ±5V analog supply; +5V digital supply	Parts Locator Drawing	6-51
		Schematic 1 of 1	6-52
I/O Board	Analog Input/output AES3 Input/output Composite Output SCA Input. ***Contains:***	Parts Locator Drawing	6-53
	L and R Analog Inputs	Schematic 1 of 6	6-54
	L and R Analog Outputs	Schematic 2 of 6	6-55
	Composite / SCA	Schematic 3 of 6	6-56
	Digital I/O	Schematic 4 of 6	6-57
	Control and Miscellaneous	Schematic 5 of 6	6-58
	Interface and Power Distribution	Schematic 6 of 6	6-59
DSP Board	DSP Chips; Local +3.3V regulator. ***Contains:***	Parts Locator Drawing	6-60
	Interconnects	Schematic 1 of 9	6-61
	Enhanced Serial Audio Interface (ESAI)	Schematic 2 of 9	6-62
	Control Interface	Schematic 3 of 9	6-63
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	86xx 8-Bit Control Interface	Schematic 6 of 9	6-66
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	External Memory Controller	Schematic 9 of 9	6-69

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	LCD Carrier	Schematic 1 of 3	6-71
	Headphone and Encoder Board	Parts Locator Drawings	6-72
	Headphone Board	Schematic 2 of 3	6-73
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Front-Panel Interface Board		Parts Locator Drawing	6-75
		Schematic 1 of 2	6-76
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Digital MPX Board	Retrofit board for pre-MPX 8600s	Parts Locator Drawing	6-78
	For 8600MPX ***Contains:***	Parts Locator Drawing	6-79
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Section 1 Introduction

About this Manual

The Adobe pdf form of this manual contains numerous hyperlinks and bookmarks. A reference to a numbered step or a page number (except in the Index) is a live hyperlink; click on it to go immediately to that reference.

> If the bookmarks are not visible, click on the "Bookmarks" tab on the left side of the Acrobat Reader window.

This manual has a table of contents and index. To search for a specific word or phrase, you can also use the Adobe Acrobat Reader's text search function.

The OPTIMOD-FM 8600 Digital Audio Processor

Orban's all-digital 8600 OPTIMOD-FM Audio Processor can help you achieve the highest audio quality in FM stereo broadcasting. Because all processing is performed by high-speed mathematical calculations within Freescale 24-bit digital signal processing chips, the processing has cleanliness, quality, and stability over time and temperature that is unmatched by analog processors.

OPTIMOD-FM 8600 is descended from the industry-standard OPTIMOD-FM audio processors. Thousands of these processors are on the air all over the world. They have proven that the "OPTIMOD sound" attracts and keeps an audience even in the most competitive commercial environment.

Because OPTIMOD-FM incorporates several audio processing innovations exclusive to Orban products, you should not assume that it can be operated in the same way as less sophisticated processors. If you do, you may get disappointing results.

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

The rest of Section 1 explains how OPTIMOD-FM fits into the FM broadcast facility. Section 2 explains how to install it. Section 3 tells how to operate OPTIMOD-FM. Section 4 through Section 6 provides reference information.

OPTIMOD-FM was designed to deliver a high quality sound while simultaneously increasing the average modulation of the channel substantially beyond that achievable by "recording studio"-style compressors and limiters. Because such processing

can exaggerate flaws in the source material, it is very important that the *source audio be as clean as possible.*

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

If you wish to place level protection prior to your studio / transmitter link (STL), use the Orban Optimod 6300 or Optimod-PC 1101 or 1101e. These processors can be adjusted so that they substitute for the AGC circuitry in OPTIMOD-FM, which is then defeated.

The 8600 simultaneously processes for analog FM and digital channels like the iBiquity™ HD Radio™ system, DAB, DRM, or netcasts. The 8600's HD output provides look-ahead peak limiting that is optimized to make the most of limited bit-rate codecs used in many digital radio systems. By eschewing any clipping, the HD output prevents the codec from wasting precious bits encoding clipping distortion products, allowing the codec to use its entire bit budget to encode the desired program material.

Thanks to a base sample rate of 64 kHz throughout the 8600's processing, the HD output can be set for audio bandwidths between 15 and 20 kHz. Many codecs operate better when fed 15 kHz audio because this enables them to use their available bit bandwidth most efficiently by concentrating on the part of the audio spectrum that is critical to perceived audio quality. This is particularly true for low rates, like 32 kbps. However, at higher sample rates, full 20 kHz bandwidth provides the same bandwidth as typical source material, so you may prefer to use it for rates of 96 kbps and above.

OPTIMOD-FM 8600FM is the same as the 8600 except that it does not provide digital radio processing. The 8600FM can be upgraded to an 8600 in the field by installing the plug-in CPU board contained in the 8600UPG/HD upgrade kit, which can be purchased from your Orban dealer.

User-Friendly Interface

- A large 5.7" (quarter-VGA) **color liquid crystal display** (LCD) makes setup, adjustment and programming of the 8600 easy. Navigation is by a miniature joystick, two dedicated buttons, and a large rotary knob. The LCD shows all metering functions of the processing structure in use.

- Use the Locate **joystick** to navigate through a menu that lets you recall a preset, modify processing (at three levels of expertise), or to access the system's setup controls.

Absolute Control of Peak Modulation

- The 8600 provides **universal transmitter protection and audio processing** for FM broadcast. It can be configured to interface ideally with any commonly found transmission system in the world, analog or digital.

- The 8600 provides **preemphasis limiting** for the internationally used preemphasis curves of 50μs and 75μs. Compared to its predecessor, its new clipping/preemphasis-control technology preserves 2.5 to 3 dB more high frequency energy while significantly reducing audible clipping distortion at all frequencies. This produces a clean, open sound whose subjective brightness more closely matches the original program.

- The 8600 achieves extremely tight **peak control** at all its outputs—analog, AES3 (for both the analog FM and HD channels), and composite baseband.

- The stereo encoder has **two outputs** with independent level controls, each capable of driving 75Ω in parallel with 47,000pF, (100ft / 30m of coaxial cable).

- By integrating the **stereo encoder** with the audio processing, the 8600 eliminates the overshoot problems that waste valuable modulation in traditional external encoders.

- The 8600 prevents aliasing distortion in subsequent stereo encoders or transmission links by providing **bandwidth limiting and overshoot compensated** 15 kHz low-pass filters ahead of the 8600's audio outputs and stereo encoder.

- The 8600 has an **internal, DSP-based stereo encoder** (with a patented **"half-cosine interpolation" composite limiter** operating at 512 kHz sample rate) to generate the pilot tone stereo baseband signal and control its peak level. The composite limiter is a unique, "you can only do this in DSP" process that beats composite clippers by **preserving stereo imaging** while **fully protecting the stereo pilot tone, RDS/RBDS, and subcarriers**.

- The stereo encoder's stereo subchannel modulator can operate in normal double sideband mode and in an experimental **compatible single sideband mode** that is offered to enable users to compare and assess the two modes. See *SSB Stereo Encoder Operation* on page 3-11.

- The Digital Radio and analog radio processing chains offer **ITU-R BS.1770-3+ Loudness Meters** and **Safety Limiters** for use in countries that enforce a BS.1770 loudness limit on digital and/or analog radio broadcasts.

- Starting with V1.4 software, the 8600HD implements **"true peak" control** in the **HD processing chain** by oversampling the HD peak limiter's sidechain at 256 kHz. This allows the 8600 **to prevent clipping in a playback device's analog signal path** by predicting and controlling the analog peak level following the playback device's reconstruction filter to an accuracy of better than 0.5 dB. For typical program material, accuracy is 0.2 dB

  Without true peak control, analog clipping can occur even if all peak values of the digital samples are below 0 dBFS. This phenomenon has also been termed "**0 dBFS+**."

Thanks to true peak control, **sample rate conversion**, unless it removes high frequency program energy or introduces group delay distortion, **cannot cause sample peaks to increase more than 0.5 dB**. For example, sample rate conversion from 48 kHz to 44.1 kHz is highly unlikely to cause sample peak clipping in the 44.1 kHz audio data.

Flexible Configuration

- OPTIMOD-FM 8600 is supplied with **analog and AES3 digital inputs and outputs**. The digital input and the two digital outputs are equipped with **sample-rate converters** and can operate at 32 kHz, 44.1 kHz, 48 kHz, 88.1 kHz, and 96 kHz sample rates. The **preemphasis status and output levels are separately adjustable** for the analog and digital outputs. Each output can emit the analog FM processed signal, the analog FM processed signal with diversity delay applied, the digital radio processed signal, or the low-delay monitor signal.

- A **loop-through connection** is provided to enable a ratings encoder (such as **Nielsen Audio®**) to be embedded in the audio processing. (See *Connecting a Ratings Encoder* on page 2-8.)

- The 8600 has **two independent equalizers, multiband compressors, and peak limiters for the FM analog and digital media processing channels**, allowing you to separately optimize the processing for each.

- An AES11 **sync input** allows you to synchronize the output sample rate of either (or both) AES3 outputs to this input. You can also **synchronize the outputs to the AES3 digital input or to 8600's internal clock**. The sync source of each AES3 output is independently selectable.

- A defeatable **diversity delay** can delay the FM analog processing output up to **13 seconds**. Delay can be trimmed in intervals of one sample of 64 kHz to **match the analog and digital paths** in the HD Radio system, eliminating the need to use the delay built into the HD Radio exciter and **permitting the 8600's internal stereo encoder and composite limiter to drive the analog FM exciter**. Both the 8600 and 8600FM offer this feature, making it convenient to use the 8600FM in dual-processor HD installations where the digital channel receives independent processing from a processor like Orban's Optimod-DAB or Optimod-PC. Each output (Analog, Digital 1, Digital 2, Composite) can be **independently configured to emit the delayed or undelayed signal**.

- Built-in **RDS/RBDS generator.**

- The analog inputs are **transformerless, balanced 10kΩ instrumentation-amplifier circuits**. The analog outputs are transformerless balanced and floating (with 50Ω impedance) to ensure highest transparency and accurate pulse response.

- The 8600 has **two independent composite baseband outputs** with digitally programmable output levels. Robust line drivers enable them to drive 100 feet of RG-59 coaxial cable without audible performance degradation.

- The 8600's two **subcarrier inputs** are mixed with the output of the 8600's stereo encoder before application to the composite output connectors. One input can be re-jumpered to provide a 19 kHz pilot reference output. Both inputs have internal level trims to accommodate subcarrier generators with output levels as low as 220 mV.

- The 8600 precisely **controls the audio bandwidth** of its **analog FM processing** to 16.5 kHz. This prevents significant overshoots in uncompressed digital links operating at a 44.1 kHz-sample rate or higher and prevents interference to the pilot tone and RDS (or RBDS) subcarrier. The bandwidth of the 8600's **digital radio output** is adjustable in 1 kHz increments between 15 kHz and 20 kHz.

- The 8600 has a defeatable **multiplex power limiter** that controls the multiplex power to ITU-R BS412 standards. An adjustable threshold allows a station to achieve maximum legal multiplex power even if the downstream transmission system introduces peak overshoots into the 8600-processed signal. Because this limiter closes a feedback loop around the audio processing, it allows the user to **adjust the processor's subjective setup controls freely** without violating BS412 limits, regardless of program material. The multiplex power limiter acts on all outputs (not just the composite output). In its most common configuration, it reduces clipper drive when it reduces power, simultaneously **reducing clipping distortion**. However, to accommodate customers who **wish to use heavier clipping to achieve a certain sound**, the MPX power controller can be configured to reduce gain after the clippers.

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

- The 8600 is designed and certified to **meet all applicable international safety and emissions standards**.

Adaptability through Multiple Audio Processing Structures

- 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, although all are active simultaneously to permit mute-free switching between them. The 8600 realizes its processing structures as a series of high-speed mathematical computations made by Digital Signal Processing (DSP) chips.

- The 8600 features six processing structures. Four of these are the same as Optimod-FM 8500: **Five-Band** (or "Multiband") for a consistent, "processed" sound with 17 ms delay (typical), free from undesirable side effects, **Low-Latency Five-Band** (12 ms delay), **Ultra-Low-Latency Five-Band** (3.7 ms delay), and **Two-Band**

(17 or 22 ms delay) for a transparent sound that preserves the frequency balance of the original program material. A special Two-Band preset creates a no-compromise "Protect" function that is functionally similar to the "Protect" structures in earlier Orban digital processors.

The 8600's two new **"MX" structures** (five-band and two-band) are similar to their 8500 counterparts except that they use the 8600's new peak limiting technology to **decrease distortion** while achieving substantial improvements in **transient punch** and **high frequency clarity**. This advanced technology requires more input-to-output delay than the older structures, so it is impractical for talent to monitor these structures off-air with headphones. The 8600 offers a special **low-delay monitor output** for this purpose. This uses a separate instance of the Ultra-Low-Latency structure to allow the monitor to provide a "FM-processed" sound.

- The 8600 **can increase the density and loudness** of the program material by multiband compression, limiting, and clipping—improving the consistency of the station's sound and increasing loudness and definition remarkably, without producing unpleasant side effects.

- The 8600 **rides gain** over an adjustable range of up to 25 dB, compressing dynamic range and compensating for both operator gain-riding errors and gain inconsistencies in automated systems.

- The 8600's Two-Band processing structures can be made **phase-linear** to maximize audible transparency.

- **Automatic Multipath Mitigator / phase skew corrector**, when activated, removes phase shifts between all elements in the input audio, converting the audio into "intensity stereo." By removing phase shifts between the left and right channels, it minimizes the amount of energy in the stereo subchannel, which consequently **minimizes multipath distortion without compromising stereo separation**.

  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.

  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%.

Controllable

- The 8600 **can be remote-controlled** by 5-12V pulses applied to eight programmable, optically isolated GPI (general-purpose interface) ports.

- The 8600 is equipped with a **serial port** to interface to an IBM-compatible computer running Orban's PC Remote software. The connection can be either direct or through an external modem.

- The 8600 has a **second serial port** that allows the user to set up **security and communications parameters** through a simple ASCII terminal program running on any PC.

- The 8600 can be connected through its built-in 100 Mbps **Ethernet port** to a **TCP/IP network**.

- The 8600 includes a **Telnet client** that allows presets to be recalled via batch files using the free PuTTY and Plink applications. The commands are **simple ASCII strings**, facilitating interface to automation systems that can emit such strings through an Ethernet or RS232 serial connection.

- A Bypass Test Mode can be invoked locally or by remote control to permit broadcast system **test and alignment** or "proof of performance" tests.

- The 8600's **software can be upgraded** remotely or locally through the 8600's serial or Ethernet port.

- **8600 PC Remote software** is a graphical application that runs under Windows XP and higher. It communicates with a given 8600 **via TCP/IP** over **modem, direct serial,** and **Ethernet** connections. You can configure PC Remote to switch between many 8600s via a convenient organizer that supports giving any 8600 an alias name and grouping multiple 8600s into folders. Clicking an 8600's icon causes PC Remote to connect to that 8600 through an Ethernet network, or initiates a Windows Dial-Up or Direct Cable Connection if appropriate. The PC Remote software allows the user to access all 8600 features and allows the user to **archive and restore presets, automation lists, and system setups** (containing I/O levels, digital word lengths, GPI functional assignments, etc.).

- The 8600 contains a built-in **line-up tone generator**, facilitating quick and accurate level setting in any system.

- The 8600 contains a versatile **real-time clock**, which allows **automation** of various events (including recalling presets) at pre-programmed times. To maintain accuracy, this clock can be automatically synchronized via the Internet to a reference time source.

- The **SNMP** (Simple Network Management Protocol) features allow you to monitor your Optimod's status and to send Alarm notifications via your Optimod's Ethernet connection to your network.

8600MPX Features

- A **192 kHz AES3 digital composite output** that is **compatible and interoperable with the 192 kHz standard being implemented by several transmitter manufacturers**. These implantations use the left audio channel of the AES3 connection to carry the baseband, but their 96 kHz Nyquist frequency makes them unable to convey 92 kHz SCAs without aliasing. Orban's implementation extends the standard by using the right audio channel of the link to provide bandwidth extension that accommodates the full 0-99 kHz baseband when used with hardware designed to use this extra information.
- **Two additional SCA inputs** that are **digitized** and can be **summed into the digital composite outputs**.

  These inputs are not mixed into the analog composite outputs. If you need to use both analog and digital composite outputs, you must use a Y connection to split the output of your subcarrier generators so that each generator output drives one digitized SCA input and one non-digitized SCA input.
- **Wordclock/10 MHz Sync Reference Input (**called REF IN and appearing on the digital composite breakout cable) can be used to lock the stereo pilot tone and the digital composite sample rate to a high precision external reference like a GPS-based frequency standard. This facilitates using the 8600 in single-frequency network applications.

Presets in OPTIMOD-FM

There are two kinds of presets in the 8600: 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 20 separate presets, selected by entering BASIC MODIFY and using the LESS-MORE control to adjust OPTIMOD-FM for amount of dynamics processing required.

Factory Presets are stored in OPTIMOD-FM'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, to which you are free to name as you wish (as long as that name does not duplicate another preset name). 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 as many User Presets (at least 100) as you have available memory. You may enter in any name you wish, up to 18 characters. The only exception is that you cannot name a User Preset the same as a Factory Preset, regardless of upper or lower case. (For example, if a Factory Preset is called "Jazz," you cannot have a User Preset called "jazz" or "JAZZ.")

User Presets cannot be created from scratch. You must always start by recalling a Factory Preset. You can then immediately store this in a User Preset, name it as you wish (within the constraints described above), and then make changes to the settings. Alternatively, you can recall a Factory Preset, make the changes, and then store this in a User Preset.

Either way, the Factory Preset remains for you to return to if you wish.

User Presets are stored in non-volatile memory that does not require battery backup.

Input/output Configuration

OPTIMOD-FM 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, or low delay monitor.
- Digital AES11 sync reference input.
- Analog left/right inputs.
- Analog left/right outputs, which can be switched independently to carry the following signals: FM analog processed without diversity delay, FM analog processed with diversity delay, digital radio processed, or low delay monitor.
- Composite stereo outputs.
- Subcarrier (SCA and RDS/RBDS) input.

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.

OPTIMOD-FM simultaneously accommodates digital and analog inputs and outputs. You can switch any of the 8600's outputs between the analog-channel processing, a low-delay monitor signal, and the HD-channel processing.

You select whether OPTIMOD-FM 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 8600's two AES3 outputs to this sync even if the digital input is asynchronous to house sync.

In Ratings Encoder Loop-Through mode, AES3 output #2 drives the ratings encoder and the Optimod's Sync Input is repurposed to receive the output of the ratings encoder. (See *Connecting a Ratings Encoder* on page 2-8.)

Analog Left/right Input/output

The left and right analog inputs are on XLR-type female connectors on the rear panel. Input impedance is greater than 10kΩ; balanced and floating. Inputs can accommodate up to +27 dBu (0 dBu = 0.775Vrms).

The left and right analog outputs are on XLR-type male connectors on the rear panel. Output impedance is 50Ω; balanced and floating. They can drive 600Ω or higher impedances, balanced or unbalanced. The peak output level is adjustable from –6 dBu to +24 dBu.

Level control of the analog inputs and outputs is accomplished via software control through System Setup. (See step 4 on page 2-27 and step 5 on page 2-30.)

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 1-15 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 Outputs

In MPX hardware, there is also 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 breakout cable supplied with your Optimod. This output is fully compatible with and interoperable with the

de-facto industry standard digital connection being implemented by transmitter manufacturers and others.

Subcarriers

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

The 8600 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 8600's remote control feature allows you to reduce the stereo main and subchannel level by connecting an on/off signal from your subcarrier generator (See page 2-9). You define the amount of reduction (in units of percent modulation) on the Input/output screen (See page 2-30). See page 2-60 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

In MPX hardware, two additional subcarrier inputs are provided on the breakout cable. 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 at the end of the breakout cable that connects to the digital composite connector. 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 96 kHz, or a 10 MHz sinewave or squarewave signal, 0.5 to 5 V peak. 10 MHz is a common output frequency produced by GPS and rubidium fre-

quency standards. You can configure the 8600 to lock its 19 kHz pilot tone and output sample frequency to this input.

When a valid reference signal is applied to this input, it locks the Optimod's Internal DSP clock to this input, so if a given AES3 L/R output's sync source is set to INTERNAL, the sample frequency at that output will be locked to this reference signal.

Do not apply an AES3 signal to this input.

Remote Control Interface

The Remote Control Interface is a set of eight optically isolated inputs on a DB-25 connector that can be activated by 5-12V DC. They can control various functions of the 8600. These functions are listed in step 1.1.1.12 on page 2-60.

Computer Interface

On the rear panel of the 8600 is a serial port and a 100 Mbps Ethernet port for interfacing to IBM-compatible PCs. These computer interfaces support remote control and metering, and downloading software upgrades.

Each 8600 package ships with 8600 PC Remote software, a program for any IBM-compatible PC with 600x800 graphics or higher (running Microsoft Windows XP or higher). 8600 PC permits you to adjust any 8600 preset by remote control, or to do most anything else that you can do from the 8600's front panel controls. The program displays all of the 8600's LCD meters on the computer screen to aid remote adjustment.

RS-232 Serial Port (Serial 1)

8600 PC Remote can communicate via modem or direct connection between the computer and the 8600 through their RS-232 serial ports. See *Administrative Operations Available via Serial Port #2* on page 2-50.

RS-232 Serial Port (Serial 2)

A computer (running a simple ASCII terminal program like Hyperterminal®) can communicate with the 8600 through direct cable connection between their RS-232 serial ports. This connection can administer communications and security, and can recall presets. It is also useful for connecting to automation systems that can emit ASCII strings through an RS-232 output. See *Connecting via Serial Port #2 Using a Terminal Program on a PC* on page 2-48.

100 Mbps Ethernet Port

This port will connect to any Ethernet-based network that supports the TCP/IP protocol. See step 1.1.1.11 on page 2-63.

Location of OPTIMOD-FM

Optimal Control of Peak Modulation Levels

The audio processing circuitry in OPTIMOD-FM produces a signal that is pre-emphasized to either the 50μs or 75μs standard preemphasis curve. It is precisely and absolutely high-frequency-controlled and peak-controlled to prevent over-modulation and is filtered at 15 kHz to protect the 19 kHz pilot and prevent distortion caused by aliasing-related non-linear crosstalk. If this signal is fed directly into a stereo encoder, peak modulation levels on the air will be precisely controlled. However, if the audio processor's signal is fed to the stereo encoder through any circuitry with frequency response errors and/or non-constant group delay, the peaks will be magnified. Peak modulation will increase, but average modulation will not. The modulation level must consequently be reduced to accommodate the larger peaks. Reduced average modulation level will cause reduced loudness and a poorer signal-to-noise ratio at the receiver.

Landline equalizers, transformers, and 15 kHz low-pass filters and preemphasis networks in stereo encoders 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 (30-15,000Hz). If low-pass filters are present, this may require the use of delay equalization. The deviation from linear-phase must not exceed $\pm 10^{\circ}$ from 30-15,000Hz.

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 stereo encoder must be canceled by a precisely complementary de-emphasis: Every pole and zero in the preemphasis filter must be complemented by a zero and pole of identical complex frequency in the de-emphasis network. An all-pole de-emphasis network (like the classic series resistor feeding a grounded capacitor) is not appropriate.

   In this example, the network could be fixed 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 most easily met when the audio processor directly feeds the stereo encoder. In the 8600, no circuit elements that might distort the shape of the waveform are interposed between the audio processor and the stereo encoder. We therefore recommend using the 8600 with its built-in stereo encoder whenever practical.

Best Location for OPTIMOD-FM

The best location for OPTIMOD-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 OPTIMOD-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 8600's AES3 digital output through an all-digital, uncompressed path to the transmitter's exciter, although this will preclude using the 8600's composite limiter.

Use the 8600'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 8600'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 8600'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 8600's output sample rate to 44.1 kHz or higher.

If only an audio link is available, use the 8600's left and right audio outputs and feed the audio, without preemphasis, directly into the link. If possible, request that any transmitter protection limiters be adjusted for minimum possible action—OPTIMOD-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. (See *Optimal Control of Peak Modulation Levels* on page 1-13).

If the transmitter is accessible:

You can achieve the most accurate control of modulation peaks by locating OPTIMOD-FM at the transmitter site and then using its stereo encoder to drive the transmitter. You can usually also obtain good results by locating OPTIMOD-FM at the studio and connecting the baseband output of its stereo encoder to the transmitter through a composite baseband STL (see page 1-18). However, many analog

composite baseband STLs do not control peaks perfectly because of bounce (see page 1-19) and locating OPTIMOD-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 8600's composite limiter.

Because OPTIMOD-FM controls peaks, it is irrelevant whether the audio link feeding OPTIMOD-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 non-linear distortion.

We strongly recommend that you use the 8600'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 most external stereo encoders. An exception is Orban's 8218 stereo encoder, which does not add overshoot, and, in fact, contains its own overshoot limiter. However, because it accepts audio in left/right form, the 8218 will not let you exploit the 8600's composite limiter.

The shorter the baseband cable from OPTIMOD-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 transformer[1] to break any ground loops. This transformer will ordinarily cure even the most stubborn hum or noise caused by the composite connection between OPTIMOD-FM and the exciter. For connection instructions, please refer to the Orban CIT25 Composite Isolation Transformer schematic diagram, which is shown on page 12 of the CIT25 manual. This is available from: ftp.orban.com/CIT25/CIT25_Instructions.pdf (Please note that Orban no longer manufactures the CIT25, which used the Jensen transformer.)

If a separate stereo encoder must be used, feed the encoder directly from the 8600's left and right analog outputs. If possible, bypass the preemphasis network and the input low-pass filters in the encoder so that they cannot introduce spurious peaks. Because of their special design, OPTIMOD-FM's preemphasis network and low-pass filters perform the same functions while retaining tight peak control.

Connect the composite output of the 8600 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 (see Figure 2-3 on page 2-10). This separation is com-

[1] http://www.jensen-transformers.com/wp-content/uploads/2014/08/jt-123-bmcf.pdf

fortably above the separation (approximately 20dB) that starts to cause perceptible changes in the stereo image.[2]

Studio-Transmitter Link

Transmission from Studio to Transmitter

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

STLs are used in three fundamentally different ways. They can either (1) pass unprocessed audio for application to the 8600's input, (2) they can pass the 8600's peak-controlled analog or digital left and right audio outputs, or (3) they can pass the 8600'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 non-linear 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 8600 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

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[2] Julie M. Adkins and Robert D. Sorkin: "Effect of Channel Separation on Earphone-Presented Tones, Noise, and Stereophonic Material," *J. Audio Engineering Society*, vol. 33 pp. 234-239, 1985.

Subjects listened to 500-Hz tones, broadband noise, and stereophonic program material through earphones and adjusted the channel separation, via a manual control, until the degradation of the spatial effect became detectable. Mean channel separations ranged from 10 to 15.9 dB for the musical selections employed and from 13.7 to 16.8 dB for the noise and tonal stimuli. The results are discussed in terms of existing data on detectable stereo separation and on the discrimination of interaural time differences. [Abstract ©Audio Engineering Society, Inc.]

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 8600'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 8600 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 8600'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 provided they do not truncate the digital words produced by the devices driving their inputs. Because the 8600's output is tightly band-limited to 16.5 kHz, it can be passed without significant overshoot by equally well by any link with 44.1 kHz 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 8600'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.

NICAM is a sort of hybrid between PCM and lossy data reduction systems. It uses a block-companded floating-point representation of the signal with J.17 preemphasis.

Older technology converters (including some older NICAM encoders) may exhibit quantization distortion unless they have been correctly dithered. Additionally, they can exhibit rapid changes in group delay around cut-off because their analog filters are ordinarily not group-delay equalized. The installing engineer should be aware of all of these potential problems when designing a transmission system.

Any problems can be minimized by always driving a digital STL with the 8600's AES3 digital output, which will provide the most accurate interface to the STL. The digital input and output accommodate sample rates of 32 kHz, 44.1 kHz, 48 kHz, 88.2 kHz, and 96 kHz.

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 8600'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 8600 installation because they allow you to use the 8600'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 preemphasis, 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 8600 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 8600 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, Optimod 6300, 8200ST Compressor / Limiter / HF Limiter / Clipper or an 4000 Transmission Limiter protects the link from overload. Of these, the 1101 and 6300 are currently manufactured as of this writing and are the preferred choices because their AGCs are identical to the AGC in the 8600.

If the 8600 is located at the transmitter and fed unprocessed audio from a microwave STL, it may be useful to use a companding-type noise reduction system (like dbx Type 2 or Dolby SR) around the link. This will minimize any audible noise buildup caused by compression within the 8600.

Some microwave links can be modified so that the deviation from linear phase is less than $\pm 10^\circ$ from 20 Hz to 15 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 (even if a composite FM stereo signal is not actually being applied to the link).

Nevertheless, in a dual-microwave system, the 8600 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 8200ST, 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 8600's processing and optimize the signal-to-noise ratio obtainable from the dual-microwave link.

If the STL microwave uses preemphasis, its input preemphasis filter will probably introduce overshoots that will increase peak modulation without any increases in average modulation. If the studio level control system is capable of producing a preemphasized output, we strongly recommend that the microwave STL's preemphasis be defeated and preemphasis performed in the studio level control system. This frees the system from potential overshoot. (The Orban 8200ST can be readily configured to produce a pre-emphasized output.)

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.

Some consultants presently offer modifications to minimize or eliminate this problem. If your exciter or STL has this problem, you may contact Orban Customer Service for the latest information on such services.

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.

Using the Orban 8100AST (or 8100A/ST) External AGC with the 8600

If you have an OPTIMOD-FM 8100A1 (or 8100A or 8100A/1) installation that uses an Orban 8100AST (or 8100A/ST) external AGC at the studio to protect an STL (with the main 8100A, 8100A1 or 8100A/1 chassis at the transmitter), you may wish to contin-

ue to use the external AGC to protect the STL when you install the 8600 at the transmitter.

If you are keeping your analog OPTIMOD-FM as a standby processor, you will probably want to use the external AGC to drive both the 8600 and the 8100A1 (also called 8100A/1) transmitter chassis in parallel. This is usually practical. However, complications will occur if you are not using an Orban 8100AXT2 (also called 8100A/XT2) Six-Band Limiter Accessory with your 8100A1, because, to correctly drive an 8600, the external AGC must be strapped as if it were driving an 8100A1 (or 8100A/1) + 8100AXT2 (or 8100A/XT2) system. Therefore, if you have only an 8100A1 (or 8100A/1), you will have to re-strap the external AGC for operation *without* the XT2 before you can put the standby 8100A1 (or 8100A/1) on the air.

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 8600's remote control feature offers the means to reduce the peak level of the 8600's audio output as necessary. This way, you can still use the 8600's line-up tone to adjust the steady-state deviation to ±75 kHz. Yet, the reduced peak level of the audio emitted from the 8600 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. See step (8.D) on page 2-22.

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 OPTIMOD-FM'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, 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. OPTIMOD-FM'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

Meters

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 on page 1-22 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.

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.

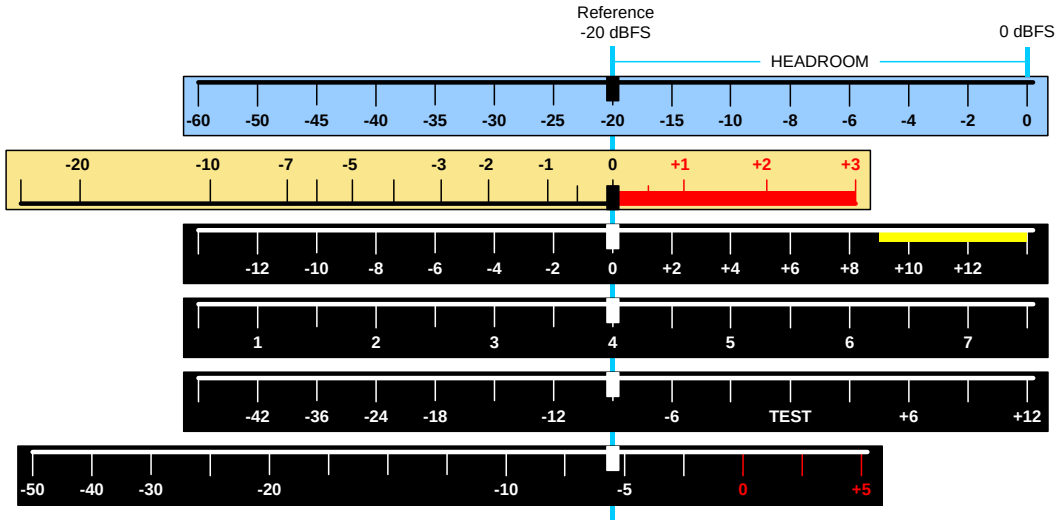


Figure 1-1: Common Audio Meter Scales, Aligned to the Same Reference Level

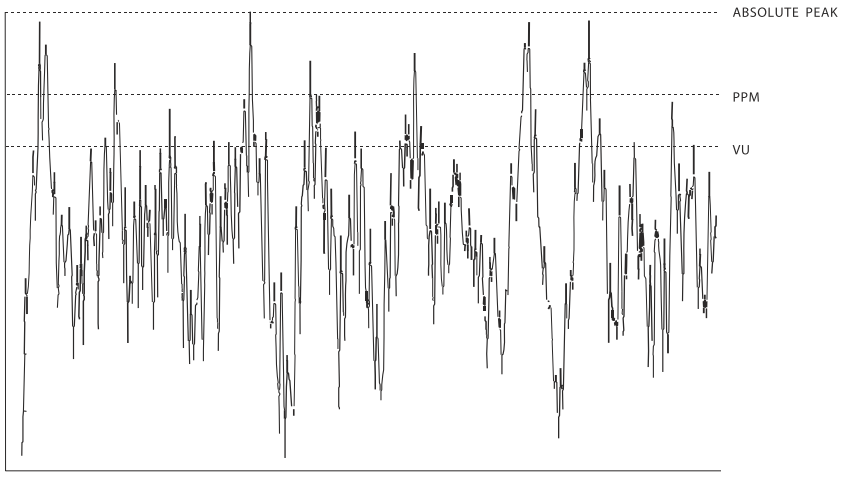


Figure 1-2: Absolute Peak Level, VU and PPM Indications

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 8600 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 8600 can accept. If you are using the AES3 digital input, a full-scale digital word corresponds to the 0 dB point on the 8600's input meter.

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

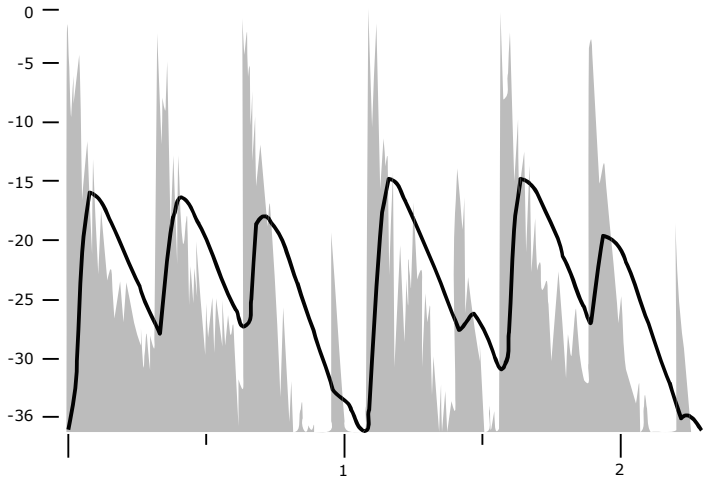


Figure 1-3: VU Meter Indication Compared to Peak Level vs. Time

meter indication is not affected by the setting of the output level control.

Composite Output Level

The Orban 8600 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 8600'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

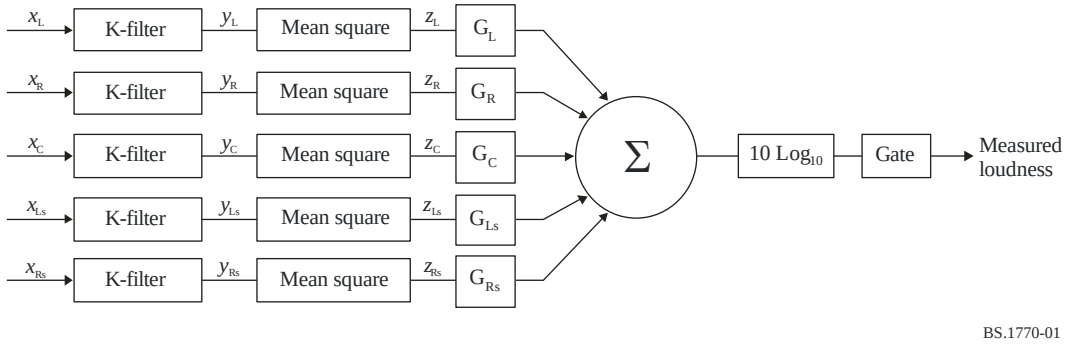


Figure 1-4: BS.1770 Loudness Meter Block Diagram (from ITU-R document)

The subjective loudness meters, labeled LOUDNESS in the 8600'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[3] algorithm.

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

[3] 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 Tech 3342, EBU Tech 3343, and EBU Tech 3344. These are free downloads that can be found easily with a search engine.

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 8600's digital radio processing chain is adjusted to make the dominant program material indicate "0 dB" on the 8600'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 3344[4], Section 5.9. This calls for a 1 kHz sinewave at –23 LUFS to produce an FM carrier deviation of ±10 kHz without pilot tone. This corresponds to 13.33% modulation without pilot tone. This calibration includes 50μs transmission preemphasis. 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 8600 to the transmission system that it is driving, the 8600 contains an adjustable test tone oscillator that produces sine waves at 8600'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 (as described on page 3-85).

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.

---

[4] EBU – TECH 3344: "Practical guidelines for distribution systems in accordance with EBU R 128; Supplementary information for EBU R 128," available for free download. Use a search engine to find the latest version.

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 8600's left/right analog output is switched to FLAT, a de-emphasis filter is inserted between output of the 8600's audio processing and its line output. Thus, as the frequency of the Test Tone is changed, the level at the 8600's line output will follow the selected de-emphasis curve. In most cases, the preemphasis filter in the driven equipment will undo the effect of the 8600's internal de-emphasis, so the 8600's output level should be adjusted such that the tone produces 100% modulation of the transmission link as measured after the link's preemphasis filter. At 100Hz, switching the de-emphasis out or in will have negligible effect on the level appearing at the 8600'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 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 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.

Monitoring on Loudspeakers and Headphones

In live operations, highly processed audio often causes a problem with **the DJ or presenter's headphones.** When an "MX" preset is active, the delay through the 8600 can be as much as 270 milliseconds. This delay will be audible as a distinct echo that almost no one can tolerate in his or her headphones while speaking. The "Multipath Mitigator" phase skew corrector adds about 250 ms of delay.

The 8600 offers two main solutions to this problem:

- Using a non-MX preset, which provides the same algorithms and performance as Optimod-FM 8500.
- Using the 8600's low-delay "monitor" output to drive talent headphones.

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

Because of their higher performance, we recommend using MX presets in HD Radio operations. In this context, the preset delay becomes part of the HD Radio diversity

delay and the 8600 automatically compensates for the preset delay to ensure that the diversity delay always agrees with the value you specified when setting the 8600 up.

With non-MX presets, the normal delay through the 8600 (from input to FM outputs) is about 18 ms when HARD or MEDIUM bass clipping is selected, as it is in all non-MX 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 8600's full look-ahead processing for low latency and about 2.5 dB loudness decrease for ultra-low latency.

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

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

  LLHARD automatically defeats the compressor lookahead. (This action is functionally equivalent to setting the LOOKAHEAD control to OUT, except that it reduces input/output delay by 5 ms).

  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 8600. These presets differ from Optimod 8200 processing in two main ways: (1) the 8600 UL presets still use the 8600's stereo enhancement, equalization section, advanced-technology AGC, composite limiter, and multiplex power controller, and (2) the 8600 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 8600's analog outputs can be switched to provide a **low-delay monitoring feed** with a delay of about 5 ms (see step 15 on page 2-38). 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 8600 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.

The front panel headphone jack provides output matching the Analog Output, except that it is always de-emphasized (even if the Analog Output is set with preemphasis).

EAS Test

For stations participating in the Emergency Alert System (EAS) in the United States, broadcast of EAS tones and data is accomplished in two different ways:

Note: Normal 8600 processing may not allow the full modulation level as required by EAS standards. It is therefore necessary to temporarily defeat the 8600's processing during the broadcast of EAS tones and data. Placing the 8600 in its BYPASS Test Mode can defeat the processing. The BYPASS GAIN control allows a fixed gain trim through the 8600. See "Test Modes," on page 3-85 for more information.

1. **Place the 8600 in Bypass mode locally.**

   A) *Locate* to SYSTEM SETUP on the pop-up Menu display, then press *Enter* button.

   B) Select TEST MODES > *Locate* to TEST MODES icon and press *Enter* button.

   C) *Locate* to BYPASS, then press *Enter* button.

   D) Begin EAS broadcast.

   After the EAS broadcast, resume normal processing:

   E) *Locate* to OPERATE in the Test Modes screen, then press *Enter* button.

   This will restore the processing preset in use prior to the Test Mode.

2. **Place the 8600 in Bypass mode by remote control.**

   To do this, you must first program any two Remote Interface inputs for "Bypass" and "Exit Test," respectively:

   A) Connect two outputs from your station remote control system to the REMOTE INTERFACE connector on the rear panel of the 8600. See Figure 2-2 on page 2-4.

   B) *Locate* to SYSTEM SETUP on the pop-up Menu display, then press *Enter* button.

   C) Select NETWORK REMOTE > *Locate* to NETWORK REMOTE MODES icon and press *Enter* button.

   D) Press and hold *Locate* right to the System Setup > Network / Remote 2 screen.

   E) Select the desired Remote Interface input (1-8), using *Locate* button.

   F) Turn the control knob to display BYPASS, then press the *Enter* button.

   G) *Locate* to a different Remote Interface input, turn the control knob to display EXIT TEST, then press the *Enter* button.

   Once you have completed these initial steps, you can place the 8600 in Bypass mode by remote control at any time:

   A) Switch the 8600 into BYPASS mode by a momentary command from your station's remote control to the input programmed as BYPASS.

   B) Begin EAS broadcast.

Be sure that you have set drive levels into the 8600 to prevent overmodulation during the test, as the 8600 provides no peak limiting in BYPASS mode.

C) When the EAS broadcast is finished, switch the 8600 from BYPASS mode by a momentary command from your station's remote control to the input programmed as EXIT TEST.

You may also choose to insert EAS broadcast tones and data directly into the transmitter, thus bypassing the 8600 for the duration of the EAS tones and data broadcast.

PC Control and Security Passcode

PC software control provides access to OPTIMOD-FM via network, modem or direct (null modem cable) connection, with IBM PC-compatible computers running Windows. PC access is permitted only with a valid user-defined passcode.

PC remote control can be ended from the front panel; this feature effectively prevents simultaneous remote and local control.

See *Security and Passcode Programming* on page 2-43.

Warranty, User Feedback

User Feedback

We are very interested in your comments about this product. We will carefully review your suggestions for improvements to either the product or the manual. Please email us at custserv@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. (See *Technical Support* on page 5-12.)

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

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

INTERNATIONAL WARRANTY

Orban warrants Orban products against evident defects in material and workmanship for a period of five years from the date of original purchase for use. This warranty does not cover damage resulting from misuse or abuse, or lack of reasonable care, or inadequate repairs performed by unauthorized service centers. Performance of repairs or replacements under this warranty is subject to submission of this Warranty/Registration Card, completed and signed by the dealer on the day of purchase, and the sales slip. Shipment of the defective item is for repair under this warranty will be at the customer's own risk and expense. This warranty is valid for the original purchaser only.

Section 2 Installation

Installing the 8600

Allow about 2 hours for installation.

Installation consists of: (1) unpacking and inspecting the 8600, (2) checking the line voltage setting, fuse, and power cord, (3) mounting the 8600 in a rack, (4) connecting inputs, outputs and power, (5) optional connecting of remote control leads and (6) optional connecting of computer interface control leads.

When you have finished installing the 8600, proceed to "Quick Setup," on page 2-18.

DO NOT connect power to the unit yet!

1. **Unpack and inspect.**

   A) If you note obvious physical damage, contact the carrier immediately to make a damage claim. Packed with the 8600 are:

Quantity	Item
1	Operating Manual
2	Line Cords (domestic, European)
2	Fuses (½ A-250V Slow-Blow for 115V; 250 mA-250V for 230V)
2	Fuse holders (gray for 115V fuses and black for 230V fuses)
4	Rack-mounting screws, 10-32 x ¾—with washers, #10
1	Null modem cable (for software upgrades and PC Remote connection)
1	Ethernet crossover cable
1	PC Remote Software CD

   B) Save all packing materials! If you should ever have to ship the 8700 (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.

   C) <u>Complete the Registration Card and return it to Orban. (please)</u>

      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. (The Registration Card is located after the cover page).

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

2. Check the line voltage, fuse and power cord.

We strongly recommend powering your Optimod from an Uninterruptable Power Supply (UPS). This will protect your Optimod from damage caused by line surges and will prevent audio interruptions caused by brownout-induced reboots.

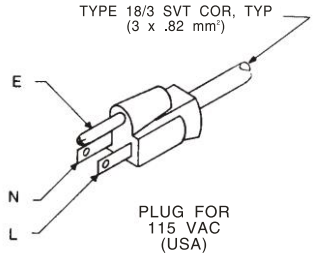
A) **DO NOT connect power to the unit yet!**

B) **Check the VOLTAGE SELECT switch. This is on the rear panel.**

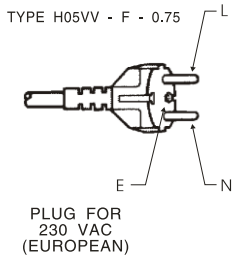
The 8600 is shipped from the factory with the VOLTAGE SELECT switch set to the 230V position. Check and set the VOLTAGE SELECT switch to your local voltage requirements. To change the operating voltage, set the VOLTAGE SELECT to 115V (for 90-130V) or 230V (for 200-250V) as appropriate.

C) **Install the proper fuse and fuse holder, per your country's standards.**

The 8600 is shipped from the factory with the fuse and fuse holder removed. Select the appropriate fuse holder and fuse from the supplied parts in the accessory kit. Use the gray fuse holder for domestic / 115V operation, or the black fuse holder for European / 230V operation. For safety, use ½ A-250 V Slow-Blow for 115V and 250mA-250V for 230V.



CONDUCTOR		WIRE COLOR	
		NORMAL	ALT
L	LINE	BROWN	BLACK
N	NEUTRAL	BLUE	WHITE
E	EARTH GND	GREEN-YELLOW	GREEN



CONDUCTOR		WIRE COLOR
L	LINE	BROWN
N	NEUTRAL	BLUE
E	EARTH GND	GREEN-YELLOW

Figure 2-1: AC Line Cord Wire Standard)

D) Check 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). The green / yellow wire is connected directly to the 8600 chassis.

If you need to change the plug to meet your country's standard and you are qualified to do so, see *Figure 2-1: AC Line Cord Wire Standard)*. Otherwise, purchase a new mains cord with the correct line plug attached.

3. Mount the 8600 in a rack.

The 8600 requires three standard rack units (5 inches / 12.7 cm).

There should be a good ground connection between the rack and the 8600 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.

4. Connect inputs and outputs.

See the hookup and grounding information on the following pages.

Audio Input and Output Connections .....Page 2-6
AES3 Digital Input and Output .....Page 2-7
Connecting a Ratings Encoder .....Page 2-8
Composite Output and Subcarrier Inputs .....Page 2-9
Grounding .....Page 2-12

5. Connect remote control interface. (optional)

For a full listing of 8600's extensive remote control provisions, refer to "Remote Control Interface Programming" on page 2-60.

Optically isolated remote control connections are terminated in a type DB-25 male connector located on the rear panel. It is wired according to Figure 2-2.

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

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

PIN ASSIGNMENT

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

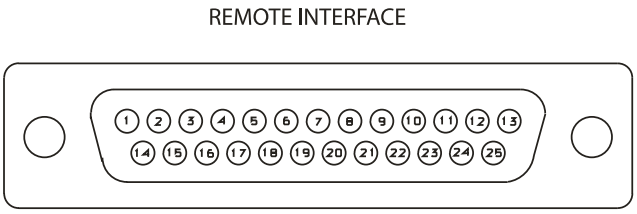


Figure 2-2: Wiring the 25-pin Remote Interface Connector

6. Connect tally outputs (optional)

See the schematic on page 6-42.

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 310 Ω resistor.

The tally outputs are protected against reverse polarity.

To avoid damaging the 8600, 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.

See *Tally Output Programming* on page 2-62 for instructions on using the tally outputs.

7. **Connect to a computer**

   You can connect to a computer via the 8600's serial connector or via an Ethernet network. (See *Networking* on page 2-63.)

   See *Installing 8600 PC Remote Control Software* on page 2-66 for more detail.

8600 Rear Panel

The **Voltage Select switch** can be set to 115V (for 90-130V operation) or 230V (for 180-260V operation).

Fuse values can be changed to support 115V or 230V operation. For safety, use ½ A-250V Slow-Blow for 115V and 250 mA-250V for 230V.

The **Power Cord** is detachable and is terminated in a "U-ground" plug (USA standard), or CEE7 / 7 plug (Continental Europe), as appropriate to your 8600's Model Number.

An **RS-232 (PC Remote) Computer Interface,** labeled **Serial 1**, is provided to connect the 8600 to IBM PC-compatible computers, directly or via modem, for remote control, metering and software downloads. An additional RS-232 port, labeled **Serial 2**, can be used for administering security and for recalling presets via simple ASCII strings. See *Administering the 8600 through its Serial Ports or Ethernet* on page 2-48.

A **Remote Interface Connector** is provided to connect the 8600 to your existing transmitter remote control. The 8600 remote control supports user-programmable selection of up to eight optically isolated GPI inputs. The 8600 remote control accepts a DB-25 connector.

> For a list and description of the programmable GPI functions, see *Remote Control Interface Programming* on page 2-60.

The **Ethernet port** is a female RJ45 connector for use with CAT5 cable in 100 Mbps Ethernet networks running the TCP/IP protocol. Use a normal Ethernet cable to connect the port to a switch or hub, or a crossover Ethernet cable to connect it directly to your computer's Ethernet port. See *Networking and Remote Control* on page 2-63.

Digital AES3 Input and **Outputs** are provided to support two-channel AES3-standard digital audio signals through XLR-type connectors. In addition, an **AES11 Sync Input** is provided to accept house sync, if required.

Analog Inputs and **Outputs** are provided to support left and right audio signals through XLR-type connectors. The digital outputs and the analog output can all be independently switched to emit the FM-processed signal, the digital radio processed signal, or the low-delay monitor signal.

Two **Composite Baseband Outputs** are provided, each with independent output level control. Each output uses a BNC connector.

Two **SCA Inputs** are provided for stations that use additional subcarriers (SCAs). Each input uses a BNC connector. The second SCA input can be reconfigured via an internal hardware jumper as a Pilot Reference Output useful for RDS (RBDS) subcarrier generators that require an external sync reference.

Audio Input and Output Connections

Cable

We recommend using two-conductor foil-shielded cable (such as Belden 8451 or equivalent), because signal current flows through the two conductors only. The shield does not carry signal and is used only for shielding.

Connectors

- Input and output connectors are XLR-type connectors.

  In the XLR-type connectors, pin 1 is CHASSIS GROUND, while pin 2 and pin 3 are a balanced, floating pair. This wiring scheme is compatible with any studio-wiring standard: If pin 2 or 3 is considered LOW, the other pin is automatically HIGH.

Analog Audio Input

- The 8600 will operate normally with nominal input levels between –14 dBu and +8 dBu.

  (0 dBu = 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 depends on the setting of the Analog Input CLIP LEVEL control. It is adjustable from 0 dBu to +27.0 dBu.

- The electronically balanced input uses an ultra low noise and distortion differential amplifier for best common mode rejection. It 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), connect the cable shield at 8600 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 8600 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 can drive loads of 600Ω or higher; the Analog OUT LEVEL control adjusts the 100% modulation level over a –6 dBu to +24 dBu range. The outputs are EMI suppressed.

- If an unbalanced output is required (to drive unbalanced inputs of other equipment), take it 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 circuit ground and 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 8600'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 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 and Output

In a standard 8600, there are two digital inputs and two digital outputs. One input accepts program audio; the other accepts AES11 house sync, although sync is not required. Either output can be programmed to emit the signal intended for the analog FM channel and/or the signal intended for the digital channel.

If you intend to use the BS.1770 Safety Limiter to drive a digital radio chain, we suggest that you use Digital Output 1 to drive the digital radio transmitter. The BS.1770 safety limiter in the analog processing chain is calibrated with reference to 100% modulation (see page 1-25) and is usable with the analog, digital, and composite outputs.

The program input and output are both equipped with sample rate converters and can operate at 32 kHz, 44.1 kHz, 48 kHz, 88.2 kHz, and 96 kHz.

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 BNC connectors. A 110Ω/75Ω balun transformer is required to interface an AES3id connection to your Optimod's digital input or output.

The digital input clip level is fixed at 0 dB relative to the maximum digital word. The maximum digital input will make the 8600 input meters display 0 dB. The reference level is adjustable using the Digital REFERENCE LEVEL control.

The 8600 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-limited to 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 8600's output signal will continue to be compatible with 44.1 kHz links even if it undergoes intermediate sample rate conversions (for example, 44.1 kHz to 96 kHz to 44.1 kHz) at various points in the program chain.

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

Connecting a Ratings Encoder

This allows a ratings encoder having AES3 inputs and outputs to be driven from your Optimod's DIGITAL OUTPUT #2, while the rating encoder's output drives your Optimod's SYNC INPUT. The ratings encoder may be embedded in one of two places:

- At the output of the AGC before the analog-FM and HD processing paths split, allowing a single ratings encoder to be used for both the analog FM and HD channels while still receiving level-controlled audio. Set the AES 2 SOURCE control to AGC>RATINGS in PC Remote or AGC>RE on the Optimod.
- Between your Optimod's left/right audio processing and stereo encoder, maintaining consistently high audio drive levels into the ratings encoder while allowing your Optimod's composite limiter to be used. Set the AES 2 SOURCE control to LIMIT>RATINGS in PC Remote or LIM>RE on the Optimod.

In Rating Encoder mode, if a valid AES3 signal is not detected at its SYNC input, your Optimod will automatically bypass the loop-through. While this will prevent dead air, it means that ratings encoding will not be applied to your broadcast.

Because of the importance of proper ratings encoder operation, your Optimod will flash a warning on its screen and on PC Remote if it enters the fallback mode. This will help detect potential failures in the ratings encoder hardware.

All 8600 units manufactured before April 2013 require a minor hardware modification to support ratings encoder loop-through operation. The modification consists of adding one jumper and removing one resistor. Field Engineering Bulletin #2, downloadable from ftp.orban.com/8600/Field_Engineering_Bulletins, provides rework instructions.

CAUTION: You will be working on a surface-mount circuit board, so to avoid damaging the circuit board you must perform the rework with tools and procedures compatible with surface-mount rework. Do not attempt the rework if you are not skilled in these procedures. Your warranty does not cover damage caused by incorrect rework tools or procedures.

Orban can perform this rework for a flat fee of $100. If you are in doubt, please contact Orban customer service to obtain a Return Authorization. See *Technical Support* starting on page 5-12.

Composite Output and Subcarrier Inputs

There are two **composite outputs**. These carry the encoded stereo signal, the stereo pilot tone, and any subcarriers that may have been applied to the 8600's subcarrier inputs.

Each output's level is independently adjustable from –13.7 dBu to +10.6 dBu.

The output impedance of composite output #1 and composite output #2 can be set to 0Ω or 75Ω via jumpers J2 and J3 respectively (located on the I/O Board). 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 6-56 and the parts locator diagram on page 6-53.)

> Each output can drive up to 75Ω in parallel with 0.047μF before performance deteriorates significantly (see Figure 2-3on page 2-10).

Connect the 8600'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 8600 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 ex-

citer end. This also eases thermal stresses on the output amplifier in the stereo encoder and can thus extend equipment life.

If a composite isolation transformer is used to break a ground loop, the exciter *must* present a 1kΩ or greater load to the transformer for proper transformer operation. See page 1-15.

Even when its composite limiter is being used heavily, the 8600 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). See Figure 5-1 on page 5-4.

The **subcarrier 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 8600's composite output in the analog domain, after D/A conversion of the 8600 stereo encoder's output but before the digitally controlled attenuators that set the composite output levels.

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 J400 (on the I/O board) from pins 3 and 4 to pins 1 and 2.

To access J400, remove the 8600's top cover according to the instructions in step 1 on page 4-2. To find J400, see page 6-53 for the I/O board parts locator drawing. To find the I/O board, see the circuit board locator

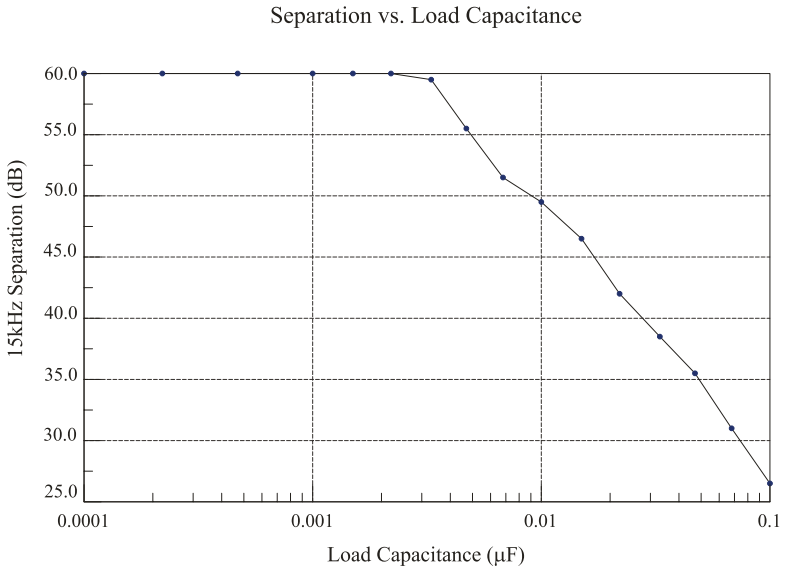


Figure 2-3: Separation vs. load capacitance

drawing on page 6-37. The schematic diagram showing J400 is on page 6-56.

Connect your subcarrier generator(s) to the 8600's subcarrier input(s) with coaxial cable terminated with BNC connectors.

The subcarrier inputs have greater than 600Ω load impedance and are unbalanced. Their sensitivity is variable from 220 mV p-p to > 10 V p-p to produce 10% injection via trimmers that are accessible for screwdriver adjustment through holes in the rear panel.

Using the PILOT REFERENCE control (in the INPUT/OUTPUT > COMPOSITE screen), you can set the phase of the reference to 0°, 90°, 180°, or 270° with respect to the pilot tone appearing at the composite output.

You can use the 19K REF control in 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.

Digital Composite Outputs

In MPX hardware, there is also 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 breakout cable supplied with your Optimod. This output is fully compatible with and interoperable with the de-facto industry standard digital connection being implemented by transmitter manufacturers and others.

Digitized Subcarrier Inputs

In MPX hardware, two additional subcarrier inputs are provided on the breakout cable. 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 at the end of the breakout cable that connects to the digital composite connector. 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 96 kHz, or a 10 MHz sinewave or squarewave signal, 0.5 to 5 V peak. 10 MHz is a common output frequency produced by GPS and rubidium frequency standards. You can configure the 8600 to lock its 19 kHz pilot tone and output sample frequency to this input.

When a valid reference signal is applied to this input, it locks the Optimod's Internal DSP clock to this input, so if a given AES3 L/R output's sync source is set to INTERNAL, the sample frequency at that output will be locked to this reference signal.

Do not apply an AES3 signal to this input.

Grounding

Very often, grounding is approached in a "hit or miss" manner. Nevertheless, 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 8600 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.

Power Ground

- Ground the 8600 chassis through the third wire in the power cord. Proper grounding techniques never leave equipment chassis unconnected to power or earth ground. A proper power ground is essential for safe operation. Lifting a chassis from power ground creates a potential safety hazard.

Circuit Ground

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

- Circuit and chassis ground are always connected

If you encounter a ground loop when feeding an unbalanced exciter input, you can balance and float the it with a Jensen JT-123-BMCF transformer (see page 2-15).

- In high RF fields, the system is usually grounded through the equipment rack in which the 8600 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.

8600 Front Panel

- **Headphone Jack** allows you to monitor the output of the processing through headphones. Headphone impedance should be 75Ω or higher.
- You can switch the **headphone feed** to receive the digital radio (HD) signal, the low-delay monitor signal, or the analog FM-processed signal before the diversity delay. This control is located on the INPUT/OUPUT > OUTPUT 2 screen.
- **Headphone Level Control** (the small blue control knob to the right of the jack) adjusts headphone output.
- The red **Enter** button allows you to choose pop-up menu items, icons and buttons. If you are in the Preset screen, it allows you to put a Factory or User Preset on-air once you have selected it.

  If you edit a Factory Preset, you must save it as a new User Preset to retain your edit permanently. Even if not saved, your edited preset will be retained automatically even if the 8600 is powered down and will be restored on-air upon power-up. If not saved, your edited preset will appear in the RECALL list of available presets as the name of its parent present prefixed by the abbreviation "modif" (for "modified").

  However, if you edit another preset before you save your old modified preset, your old edited preset will be lost—the 8600 automatically retains only one "modified" preset. Therefore, it is wise to rename and save any edited preset you wish to keep, using the 8600's SAVE main menu item. This ensures that your edited preset will not be overwritten accidentally.

- The green joystick, labeled **Locate**, is a pointing device that allows you to navigate to settings and controls on each screen. Pressing and holding the knob left or right moves you to the previous and next function screens (when multiple screens are available).
- A yellow **Escape** button allows you to navigate quickly to underlying screens, higher-level screens or the Meters screen and displays the pop-up menu.

  When a pop-up item, like Menu, is onscreen, *Escape* always returns you to the underlying screen.

  Pressing *Escape* from a secondary screen page, like System Setup > Place / Date / Time 1 takes you back to the top level; in this case, the System Setup screen.

Escape from top-level screens (like the System Setup screen), brings you back to the Meters screen. (If you are already in the Meters screen, *Escape* displays the pop-up Menu.)

- The **Control Knob** is the large blue knob on the front panel. Turning the knob scrolls through displayed lists (like the Preset screen list) or changes a setting that is highlighted onscreen (e.g., the setting last selected by the *Locate* joystick). Pushing the knob in, towards the front panel, displays the pop-up Menu over the previous screen.

- **Screen Display** supplies control setting information and screen help and displays the gain reduction and level meters (described directly below).

  The 8600's screen displays the following meters and indicators:

  - IN METERS show the peak input level applied to the 8600's analog or digital inputs with reference to 0 dB = digital full-scale.

  - AGC METERS show the gain reduction of the slow AGC processing that precedes the multiband compressor. Full-scale is 25 dB gain reduction.

    Because the AGC is a two-band unit with Orban's patented bass coupling system, the two meters indicate the gain reduction of the AGC Master and Bass bands.

  - GATE INDICATORS show gate activity. They light up when the input audio falls below the threshold set by the gate threshold controls. (There are two gating circuits—one for the AGC and one for the multiband limiter—each with its own Gate Thresh control.) When gating occurs, the AGC and compressor's recovery times are slowed drastically to prevent noise rush-up during low-level passages.

  - MULTIBAND GAIN REDUCTION METERS show the gain reduction in the multiband compressor. Full-scale is 25 dB gain reduction. The MB GR METER switch (in INPUT/OUTPUT > UTILITIES) determines what signals the 2-Band and 5-Band Compressor gain reduction meters indicate. The switch can be set to FM, HD, or SPLIT. In SPLIT mode, the 8600's front panel display shows the gain reduction of the FM analog multiband compressors on the left side of the split meters and the gain reduction of the digital radio compressors on the right. If the left and right channel gain reductions are not identical in a given band, its meter displays the larger of the left or right channel gain reductions.

  - 2B HF METERS DISPLAY THE GAIN REDUCTIONS IN DB OF THE INDEPENDENT LEFT AND RIGHT CHANNEL HIGH FREQUENCY LIMITERS IN THE 8600'S TWO-BAND STRUCTURE. THESE METERS APPEAR ONLY WHEN THE 8600 IS IN TWO-BAND MODE.

  - OUT METERS display 8600's instantaneous peak output level.

  - COMP METER displays the stereo encoder's output level before the COMP 1 or COMP 2 attenuators, in percent scale over a 125 to 0 range.

- HD METERS display the gain reduction of the left and right look-ahead limiters that feed the HD outputs.
- The LOUDNESS LEVEL meters, which uses the ITU-R BS.1770-3 algorithm, indicates the relative loudness level at the output of the digital and analog radio processing chains. The reference level for the analog and digital processing chains are set independently. The loudness level for the analog radio chain is shown as a number at the bottom of the main meter screen on the 8600 unit; it is shown as a meter on PC Remote. See *BS.1770 Loudness Level* on page 1-24.
- The LOUDNESS GR meters indicate the amount of gain reduction produced by the BS.1770 Safety Limiters in the digital and analog radio processing chains.
- MULTIPLEX POWER GR / BS.1770 GR METER uses a two-color display to indicate the action of the ITU Multiplex Power controller and the BS.1770 safety limiter in the analog radio processing chain. It shows how much the Multiplex Power Controller and BS.1770 Safety Limiter have reduced the clipper drive, reducing the average power in the processed audio. The BS.1770 gain reduction is shown in cyan and the multiplex power gain reduction is shown in dark blue.

  This meter, labeled "PWR," is displayed on the 8600's color LCD. It always appears when the Two-Band Structure is active. When the Five-Band Structure is active, the meter only appears when the Multiplex Power Controller is turned on.

External AGC Installation (optional)

[Skip this section if you are not using an external AGC ahead of the 8600. Continue with "Quick Setup" below.]

- As of this writing, the currently manufactured Orban products that can be used as external AGCs are Optimod-PC 1101/1101e and Optimod 6300. Their manuals contain instructions on how to use them in this application. They are the preferred choices because their AGCs are identical to the AGC in the 8600.
- Discontinued Orban products usable as external AGCs include the 8200ST, 464A "Co-Operator," 8100AST, and Optimod-PC1100. In this manual, we do not provide step-by-step instructions for setting up all of these older products, although it should be easy to extrapolate from the instructions we provide for the 8200ST below.
- If you are using an Orban 8100AST (or 8100A/ST) external AGC, refer to page 1-19.

If you are using an Orban 8200ST external AGC:

If the STL uses preemphasis, its input preemphasis filter will probably introduce overshoots that will increase peak modulation without any increase in average modulation. We therefore strongly recommend that the STL transmitter's preemphasis be defeated (freeing the STL from such potential overshoot) and that the 8200ST be used to provide the necessary preemphasis.

If the STL transmitter's preemphasis cannot be defeated, then configure the 8200ST for flat output. In this case average modulation levels of the STL may have to be re-

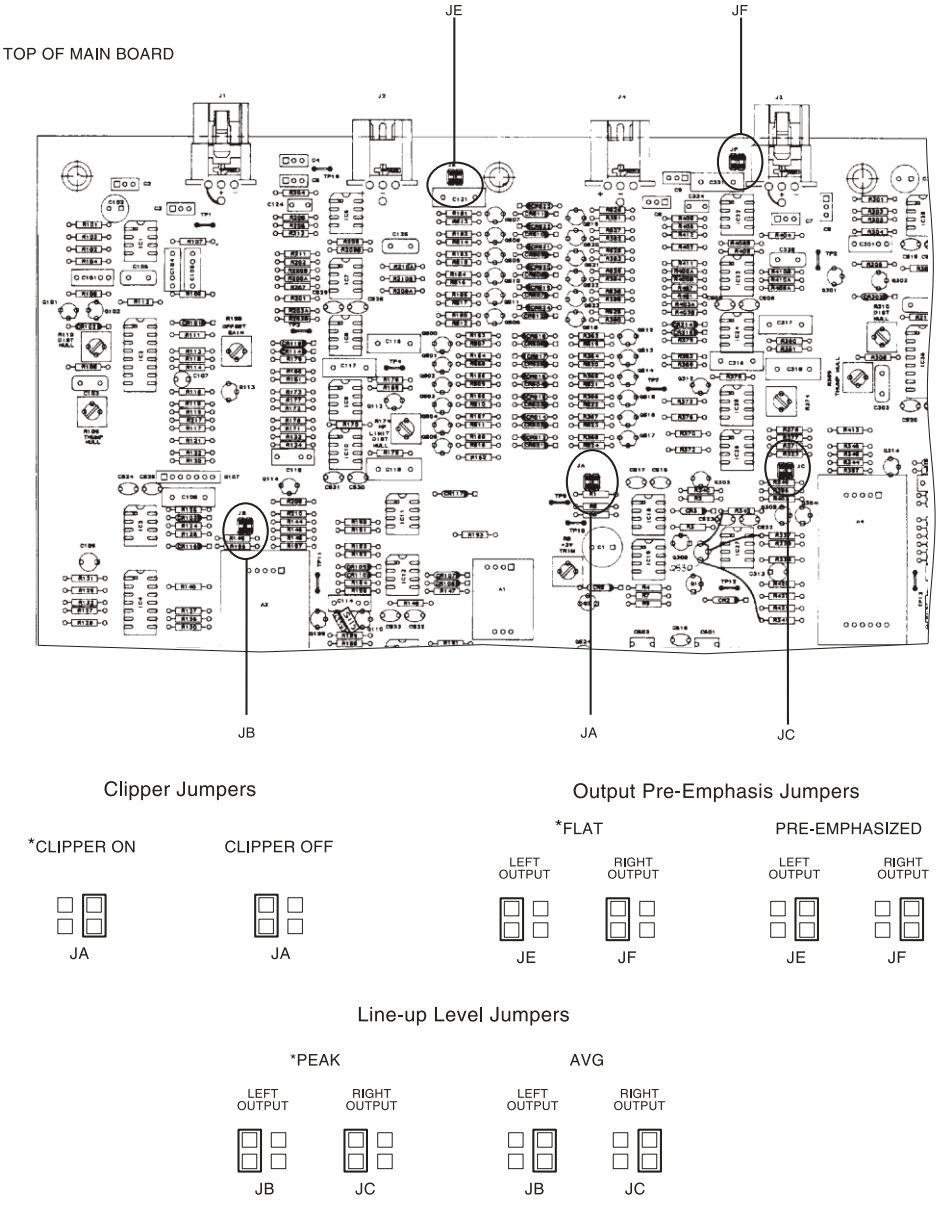


Figure 2-4: 8200ST Jumper Settings (*Factory Configuration)

duced to accommodate the overshoots.

1. **Configure the 8200ST's internal jumpers.**

   A) Remove all screws holding the 8200ST's cover in place; then lift it off.

   Refer to Figure 2-4 on page 2-16.

   B) Place jumper JA in the CLIPPER ON position.

   C) If you have defeated the STL transmitter's preemphasis, place jumpers JE and JF in the PRE-EMPHASIZED position.

   D) If you cannot defeat the STL transmitter's preemphasis, place jumpers JE and JF in the FLAT position.

   E) Replace the top cover and then replace all screws snugly. (Be careful not to strip the threads by fastening the screws too tightly.)

2. **Install the 8200ST in the rack. Connect the 8200ST's audio input and output.**

   Refer to the *8200ST Operating Manual* if you require information about installation, audio input, and audio output connections to the 8200ST.

3. **Set 8200ST Output Level with tone.**

   A) Press the TONE button on the 8200ST.

   The TONE lamp should light and the modulation meters should indicate "0." If they do not, re-strap jumpers JB and JC to "peak." (Refer to Figure 2-4 on page 2-16.)

   The 8200ST is now producing a 400Hz sine wave at each output. The peak level of this tone corresponds to 100% modulation.

   B) Adjust the 8200ST's L OUT and R OUT controls so that the STL transmitter is being driven to 100% modulation.

   The L OUT and R OUT controls are now correctly calibrated to the transmitter. If no significant overshoot occurs in the transmitter, the MODULATION meter will now give an accurate indication of peak modulation of the STL.

   C) Turn off the tone by pressing the TONE button.

   If the STL transmitter suffers from bounce or overshoot, you may have to reduce the L OUT and R OUT control settings to avoid peak overmodulation caused by overshoots on certain audio signals.

4. **Set controls for normal operation with program material.**

   The following assumes that a VU meter is used to determine 8200ST line drive levels with program material.

   A) Set controls as follows:

   HF LIMITER.... Set to match the preemphasis of the transmission system
   L&R Out ..... do not change
   GATE 12:00

RELEASE 12:00
VOICE ..... OFF
AGC ..... ON
COUPLE ..... ON

B) Feed the 8200ST either with tone at your system reference level (0VU), or with typical program material at normal levels.

C) Adjust the GAIN REDUCTION control for the desired amount of gain reduction.

We recommend 8-15 dB gain reduction for most formats.

If the STL uses preemphasis, its input preemphasis network will probably introduce overshoots that will increase peak modulation without any increase in average modulation. We therefore strongly recommend that the STL transmitter's preemphasis be defeated (freeing the STL from such potential overshoot) and that the 464A be used to provide the necessary preemphasis.

If the STL transmitter's preemphasis cannot be defeated, configure the 8200ST for flat output. In this case, average modulation levels of the STL may have to be reduced to accommodate the overshoots.

Quick Setup

The 8600's Quick Setup feature provides a guided, systematic procedure for setting up the 8600. It should be adequate for most users without special or esoteric requirements. Following this section, you can find more detailed information regarding setup outside the Quick Setup screens. Mostly, you will not need this extra information.

Quick Setup configures the 8600 for an analog-FM facility only. If you are setting up an digital radio facility (HD Radio, DAB/DAB+, etc.), you must use the detailed instructions found after this Quick Setup section.

For the following adjustments, use *Locate* (the green joystick, between *Escape* and *Enter*) to select parameters. After you have highlighted the desired parameter on the screen, use the front panel control knob to adjust the parameter settings, as desired.

1. **From the pop-up Menu display, Locate to System Setup, then press the Enter button.**

   If the pop-up Menu isn't onscreen, press the control knob in.

2. **From the System Setup screen, Locate to the Quick Setup icon, then press the Enter button.**

   Quick Setup presents a guided sequence of screens into which you must insert information about your particular requirements.

Each Quick Setup page is titled in the top right corner (e.g., page 1 is System Set-up > Quick Setup 1).

3. **Set time and date.**

   A) *Locate* to the Time & Date screen (System Setup > Quick Setup 2).

   B) Choose Time Format as desired (either 24-hour time or 12-hour AM / PM-style time).

   C) Set hours, minutes, and seconds, in that order, using a 24-hour format for entering hours even if you have set the time format to 12-hour.

      Seconds will stop advancing when you set hours and minutes. So set seconds last.

   D) Choose the desired date format.

   E) Set today's date.

   F) If you want the clock to reset itself automatically to conform to Daylight Saving Time (Summer Time), use the BEGINS and ENDS fields to specify when Daylight Saving Time begins and ends in your area. If you do not wish to use this feature, leave the BEGINS and ENDS fields set to Off.

      Note that the clock will set itself automatically if you have set the 8600 to synchronize to an Internet timeserver. See *Synchronizing Optimod to a Network Time Server* in page 2-69.

4. **Set preemphasis in regional settings screen.**

   A) *Locate* to the Regional Settings screen (SYSTEM SETUP > QUICK SETUP 3).

   B) Select the preemphasis (either 75μS or 50μS) used in your country.

      Because there is only one field in this screen, you do not have to *LOCATE* to the Preemphasis field; it will automatically be active.

      There may be a slight time delay between when you move the knob and when the preemphasis indication changes.

      You can change the preemphasis later from the INPUT/OUTPUT > UTILITIES screen.

5. **Set External AGC mode.**

   A) *Locate* to Studio Configuration screen (SYSTEM SETUP > QUICK SETUP 4).

   B) Set the External AGC mode.

   - Set the field to YES if you have an external AGC (such as an Orban 1100, 1101, 6300, 8200ST OPTIMOD-Studio, Orban 464A Co-Operator, or similar AGC) installed at your studio feeding the studio-to-transmitter link. This setting defeats the 8600's AGC for all presets.
   - If you do not have an external AGC installed, set the field to NO; this setting allows the selected preset to determine the 8600 AGC status.

- If you are using an Orban 4000 Transmission Limiter, set field to NO (so that the AGC function in the 8600 continues to work). The Orban 4000 is a transmission system overload protection device; it is normally operated below threshold. It is not designed to perform an AGC or gain-riding function and it cannot substitute for the AGC function in the 8600.

  Most of the processing structures in the 8600 control level with a preliminary AGC (Automatic Gain Control). If you are using a suitable Automatic Gain Control at the studio, the AGC in the 8600 should be defeated. This is so that the two AGCs do not "fight" each other and so they do not simultaneously increase gain, resulting in increased noise.

6. **Set input levels.**

A) Prepare to adjust the Input Reference Levels.

a) Feed normal Program material to the 8600.

Play program material from your studio, peaking at normal program levels (typically 0VU if your console uses VU meters).

b) *Locate* to the Reference Levels screen (SYSTEM SETUP > QUICK SETUP 5).

The Reference Level screen allows you to match the 8600 to the normal operating level to be expected at the 8600, so the 8600's AGC can operate in the range for which it was designed. There are separate settings for the analog and digital inputs. If you provide both analog and digital inputs to the 8600, optimum adjustment is achieved when the same amount of processing is indicated for either analog or digital inputs. This will allow you to switch between analog and digital inputs without sudden level changes.

B) Using the SET INPUT TO field, set the input to ANALOG.

[Skip this step if you are not using the analog input.]

a) Adjust the ANALOG REFERENCE LEVEL so that the meter reads an average of 10 dB gain reduction.

[–9 dBu to +13 dBu (VU), or –2 to +20 dBu (PPM)] in 0.5 dB steps

The ANALOG REFERENCE LEVEL VU and PPM settings track each other with an offset of 8 dB. This compensates for the typical indications with program material of a VU meter versus the higher indications on a PPM.

If you know the reference VU or PPM level that will be presented to the 8600, set the ANALOG REFERENCE LEVEL to this level, but please verify it with the steps shown directly below.

b) If the AGC gain reduction meter averages less than 10 dB gain reduction (higher on the meter), re-adjust the ANALOG REFERENCE LEVEL to a lower level.

c) If the AGC gain reduction meter averages more gain reduction (lower on the meter), re-adjust the ANALOG REFERENCE LEVEL to a higher level.

This control has no effect on the AES3 digital input.

C) Using the SET INPUT TO field, set the input to DIGITAL.

[Skip this step if you are not using the digital input.]

a) Adjust the DIGITAL INPUT REFERENCE so that the meter reads an average of 10 dB gain reduction.

[–30 to –10 dBFS (VU), or –23 to –3 dBFS (PPM)] in 0.5 dB steps.]

The DIGITAL REFERENCE LEVEL VU and PPM settings track each other with an offset of 8 dB. This compensates for the typical indications with program material of a VU meter versus the higher indications on a PPM.

If you know the reference VU or PPM level that will be presented to the 8600, set the DIGITAL REFERENCE LEVEL to this level, but do verify it with the steps shown directly below.

b) If the AGC gain reduction meter averages less than 10 dB gain reduction (higher on the meter), re-adjust the DIGITAL REFERENCE LEVEL to a lower level.

c) If the AGC gain reduction meter averages more gain reduction (lower on the meter), re-adjust the DIGITAL REFERENCE LEVEL to a higher level.

This control has no effect on the analog inputs.

D) Select primary Input Source:

Using the SET INPUT TO field, set the input to the source (analog, digital, or DIG+J17) that you will use for normal programming.

DIG+J17 applies J.17 de-emphasis to the incoming digital signal. It is only applicable to certain STLs (like NICAM) that use this type of preemphasis and that have not applied their own de-emphasis prior to their3 AES outputs.

7. Configure output.

A) *Locate* to the Output Configuration screen (SYSTEM SETUP > QUICK SETUP 6).

B) Set the ANALOG OUTPUT PRE / FLAT control to PRE-E (for preemphasis) or FLAT.

[Skip this step if you will not be using the analog output.]

If you will use the analog output to drive a stereo encoder, PRE-E provides the best performance because this stereo encoder does not have to restore the preemphasis. However, if you cannot defeat the preemphasis in your stereo encoder, or if you will use the analog output for monitoring, set the output FLAT.

If you are sending the output of the 8600 through a digital link that uses lossy compression (like MPEG, APT-X, or Dolby), set the output FLAT. Lossy codecs cannot handle pre-emphasized signals.

C) Set the DIGITAL OUTPUT #1 PRE / FLAT control to PRE-E (for preemphasis), FLAT, or PRE-E+J17.

[Skip this step if you will not be using the "AES1" digital output.]

(See the notes immediately above.)

PRE-E+J17 applies both FM preemphasis and J.17 preemphasis (in cascade) to the signal and is only used with STLs using J.17 preemphasis when their own J.17 preemphasis filters are bypassed. These are rare.

D) Set the DIGITAL OUTPUT #1 SAMPLE RATE to 32, 44.1, 48, 88.2, or 96 kHz.

[Skip this step if you will not be using the "AES1" digital output.]

The 8600's fundamental sample rate is always 64 kHz, but the internal sample rate converter sets the rate at the 8600's digital output. This adjustment sets the output sample rate to ensure compatibility with equipment requiring a fixed sample rate.

8. Set output levels.

A) Locate to the Set Output Levels screen (SYSTEM SETUP > QUICK SETUP 7).

B) You can use either program material or tone to set the output level (and thus, the on-air modulation). If you want to use tone, set the 400HZ CALIBRATION TONE to ON.

C) Using a modulation monitor or modulation analyzer, adjust the outputs you are using (analog, digital, composite 1 and composite 2) to make the modulation monitor read 100% modulation (usually ±75 kHz deviation).

D) If you are using program material, make sure that the program material is loud enough to produce peaks of frequent recurrence that hit the 8600's peak limiting system, thereby defining the maximum peak level that the 8600 will produce. In the U.S., we recommend using 900μs peak weighting on the peak modulation indicator, as permitted by F.C.C. rules. This will cause the monitor to ignore very low energy overshoots and will produce the highest peak modulation permitted by law.

In other countries, use a peak-indicating instrument as specified by the regulatory authority in your country.

If you are required to implement the multiplex power limits specified by ITU-R 412, you may seldom see peaks hitting ±75 kHz deviation. In this case, we advise you to set the output level using the 8600's reference 400Hz tone.

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 8600 can reduce audio modulation to compensate for subcarriers. Once you are finished with Quick Setup, navigate to SYSTEM SETUP > NETWORK REMOTE 1 and program the Remote Interface Terminal for MOD. REDUCTION 1 or MOD. REDUCTION 2. Set the amount of modulation reduction by navigating to INPUT/OUTPUT > COMPOSITE and adjusting the MOD. RED. 1 and MOD. RED. 2 parameters. When both are active, the modulation reduction is the sum of their settings. In general, 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 MOD. RED. 1 to "5%" and MOD. RED. 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 MOD. REDUCTION function is active as long as signal is applied to its associated GPI input.

The advantage of using the MOD. REDUCTION function is that the pilot injection stays constant when the audio modulation is reduced. However, using the MOD. REDUCTION function is slightly inconvenient because it requires programming and activating at least one 8600 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 LEVEL control (in the INPUT / OUTPUT > COMPOSITE screen) until the injection equals 9% modulation.

9. Choose a factory preset.

A) *Locate* to the Choose Preset screen (SYSTEM SETUP > QUICK SETUP 8).

B) Using the *Locate* joystick up/down control or turning the control knob, highlight a preset corresponding to your format. Press *Enter* to put the highlighted preset on the air.

Preset names are *just suggestions*. Some of the most competitive presets (the "Loud" and "Impact" families) are intentionally not named for formats because these presets can be used in a wide variety of competitive mass-appeal music formats. 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.

You can easily modify a preset with the 8600's one-knob LESS-MORE feature. After you have finished with Quick Setup, Navigate to the BASIC MODIFY screen. If you do not see the LESS-MORE screen immediately, press and hold the *Locate* joystick to the right or left until you find the screen. Turning LESS-MORE up will produce more loudness but also more processing artifacts like distortion and unpleasant density. Turning LESS-MORE down will make the sound cleaner, more open, and easier to listen to, but will also make it quieter.

C) Congratulations! You are now on the air with your initial sound. Feel free to read the material in Section 3 of this manual, which describes the various presets and how you can customize them to an almost unlimited extent.

10. Complete Station ID.

The Station ID is an optional setting that you can provide to associate the 8600 with the station providing the program material (e.g., "KABC"). The Station ID appears on the Meters screen to the left of the date, and on many other screens, in the left pane, above the date.

A) Locate to the Station Identifier screen (System Setup > Quick Setup 9).

B) To erase the default Station ID name, use *Locate* to highlight CLEAR, then press *Enter*.

C) Enter in your Station ID name.

For each keypad item, Locate to the item and press *Enter.*

For upper case letters, first *Locate* to the SHIFT key and then press *Enter*.

D) When finished entering your name, highlight SAVE and press *Enter*.

11. Complete Quick Setup.

A) Locate to the Finished screen (SYSTEM SETUP > QUICK SETUP 10).

B) Press *Escape* once to return to the System Setup screen, or twice to display the Meters screen. Alternatively, press the control knob to display the pop-up Menu.

Quick Setup is finished, unless your country is required to meet ITU-R 412-7 requirements (see next step)

Note that Quick Setup only guides you through setting up the processing for the analog FM output. To set up the HD output, see About the 8600's HD / Digital Radio Processing *on page 3-71*.

12. If you are required to meet the "multiplex power" limitations of ITU-R 412-7 in your country, activate the 8600's ITU-R 412 controller.

[Skip this step if your country does not enforce ITU-R 412. At the time of this writing, it is only enforced in certain European countries.]

A) Navigate to the INPUT/OUTPUT > UTILITIES screen.

a) Press the control knob to display the pop-up Menu.

b) Turn the knob to highlight INPUT/OUTPUT and press the knob.

c) *Locate* to the INPUT/OUTPUT > UTILITIES screen.

B) Set the MULTIPLEX POWER THRESHOLD to "0.0 dB."

If your transmission system introduces overshoot in the signal path after the 8600 (including the transmitter), instead set the MULTIPLEX POWER THRESHOLD so that it equals the amount of peak overshoot (in dB) in the transmission system. If you do not do this, the 8600's ITU-R 412 controller will set the average multiplex power too low.

> The easiest way to measure system overshoot is to turn the multiplex power controller off temporarily. Then set the 8600's output level, using its built-in 400Hz reference tone, so that the transmitter produces ±75 kHz deviation. Finally, play program material with lots of high frequency energy and bass transients (like bright rock music with heavy kick drum) and observe the peak deviation produced by the program material. The overshoot is the amount (in dB) by which the deviation with program material exceeds ±75 kHz deviation.

If your country does not enforce ITU-R 412, the MULTIPLEX POWER THRESHOLD should be set to OFF.

Because the multiplex power controller sometimes uses the output of the 8600's stereo encoder as its reference, set the COMPOSITE LIMIT DRIVE control (page 3-43) to OFF If you are using the 8600's analog or digital output (not its composite output) to drive the transmission system.

The total gain reduction of the BS.1770 controller (see step 13 below) and MPX power controller is split before and after the FM peak limiter such that the first 3 dB of gain reduction occurs before the peak limiter. Hence, you can achieve full ±75 kHz deviation when the gain reduction is between 0 and 3 dB. Any gain reduction greater than 3 dB is applied after the peak limiter, permitting you to achieve more of the "FM limiter sound" with higher amounts of gain reduction. (Because some of the gain control is applied after the peak limiter, the peak deviation may be limited to less than ±75 kHz.)

See the notes on the Multiplex Power Offset control on page 3-45. For more information on the multiplex power controller, refer to *ITU-R Multiplex Power Controller* on page 3-81.

For "MX" presets or anytime the diversity delay is active, the 8600 computes the multiplex power from the left and right outputs of the audio processing because there is too much delay to use the output of the stereo encoder. Under these circumstances, the 8600 cannot directly take into account the setting of the COMPOSITE LIMIT DRIVE control.

13. Set up and activate the BS.1770 Safety Limiter in the analog radio processing chain (optional)

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

The MPX POWER OFFSET control allows you to reduce the drive level into the BS.1770/BS.412 chain to prevent excessive amounts of gain reduction. This arrangement allows you to apply as much "FM processor sound" as you want to the audio. Turn the MPX POWER OFFSET control up (closer to 0) to drive the peak limiters harder.

We recommend using as little gain reduction as possible to prevent unnatural sounding source-dependent loudness variations—a limitation of the BS.1770 algorithm itself, which can over-indicate the loudness of material with substantial dynamics compression by up to 3 dB compared to subjective impressions by lis-

teners[1]. We suggest using a maximum of 3 dB gain reduction in the BS.1770 safety limiter.

A) Navigate to the INPUT/OUTPUT > UTILITIES screen.

B) Turn the 1770 SAFETY LIMITER control ON.

C) Set the 1770 LOUDNESS THRESH control to the target loudness specified by your country's governing authority. EBU B 128 specifies –23 LUFS.

> The BS.1770 Loudness Meter is calibrated in relative units (LU) so that it indicates "0" when the loudness is equal to the target loudness set by the 1770 LOUDNESS THRESH control. In an overshoot-free transmission system, a 1 kHz sinewave at –23 LUFS must produce an FM carrier deviation of ±10 kHz without pilot tone. This corresponds to 13.33% modulation without pilot tone. This calibration includes 50μs transmission preemphasis.
>
> **Note for version 2.0.2 software**: We have found that with typical pop music, a –23 LUFS setting typically produces about –5 dB BS.412 MPX Power and rarely produces ±75 kHz peak carrier deviation. 0dB MPX power is not reached unless the BS.1770 THRESHOLD control is set to approximately –17 LUFS. This behavior seems odd (the low modulation level significantly compromises the received S/N ratio in weak signal areas and in moving vehicles), so we are unsure if our interpretation of EBU Tech 3344 is correct in this initial release and we welcome any feedback. Please email Bob Orban at the address shown at http://www.orban.com/contact/ if you discover an error in our calibration.

D) The calibration of the 1770 LOUDNESS THRESH control is correct if the transmitter's modulation level was calibrated using your Optimod's built-in lineup tone (step 8 on page 2-22). If you turned down the drive level to the FM transmitter (via the COMPOSITE OUT, DO 100%, or AO 100% controls depending on how you are driving your transmitter) to compensate for overshoots in the signal path following the Optimod, turn the 1770 LOUDNESS THRESH control up by the same amount that you turned the output level control down. This will preserve correct loudness at the receiver.

> For example, assume that you turned down the output level control by 2 dB compared to its calibration using your Optimod's tone generator, so that a line-up tone that would produce ±75 kHz deviation in an overshoot-free transmission system instead produces ±59.6 kHz deviation to allow for 2 dB of overshoot with program material. In this case, set the 1770 LOUDNESS THRESH control to –21 LUFS instead of –23 LUFS.

E) If you have not already done so in step 12 on page 2-24, activate the MPX Power Controller by setting it to some setting other than OFF. This is necessary to enable the MULTIPLEX POWER OFFSET control.

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[1] See Begnert, Fabian; Ekman, Håkan; Berg, Jan, "Difference between the EBU R-128 Meter Recommendation and Human Subjective Loudness Perception," AES Convention Paper 8489, 131st AES Convention, (October 2011). This paper states, "These loudness-equalized signals gave rise to a perceived maximum loudness difference of 2.8 dB."

Using program material typical of your format, adjust the MULTIPLEX POWER OFFSET control so that the BS.1770 Safety Limiter produces about 3 dB of gain reduction on typical program material. This control is part of the processing preset, so after you have set the control, save the preset as a User Preset.

> You must adjust the MULTIPLEX POWER OFFSET control for each processing preset you plan to use and then save each one as a user preset.

The following material provides detailed instructions on how to set up the 8600. If Quick Setup does not fully address your setup needs or if you wish to customize your system beyond those provided with Quick Setup, then you may need the additional information in the sections below. You will need this information if you are setting up a digital radio facility. However, for most users, this material is only for reference, because Quick Setup has enabled them to set up the 8600 correctly.

Analog and Digital I/O Setup

For the following I/O calibration parameters, use the *Locate* joystick to highlight input/output parameters. When the desired parameter is highlighted, turn the front panel control knob to adjust the parameter settings as desired.

Analog and digital parameters appear on the same screen. If you are not using a given input or output, ignore the parameters associated with it.

1. **Specify processing preemphasis.**

   Navigate to the INPUT/OUTPUT > UTILITIES screen. Use the PROCESS PREEMPHASIS control to select the preemphasis (either 75μS or 50μS) used in your country

2. **Temporarily set the External AGC mode to "No."**

   Navigate to the INPUT/OUTPUT > UTILITIES screen and set EXTERNAL AGC to NO.

   > If you are using an external AGC, you will restore this setting to YES after the setup procedure is complete.

3. **Adjust input selector.**

   A) Navigate back to the INPUT/OUTPUT > INPUT screen.

   B) Set the INPUT to ANALOG.

4. **Adjust Clip Level control.**

   [0 dBu to +27 dBu] in 0.5 dB steps

   This step matches the level at which the 8600's A-D (Analog-to-Digital) converter clips to the absolute maximum peak level that your installation supplies to the 8600's analog input.

This setup maximizes the 8600's signal-to-noise ratio. If the clip level is set too low, the 8600's analog-to-digital converters will overload and distort on program peaks. If the clip level is set too high, the signal-to-noise ratio will suffer. Use care and attention in setting this adjustment.

We have found that the single most common reason for distorted sound on-air in other Orban digital processors is maladjustment of the CLIP LEVEL control, such that the A/D converter is clipping and distorting. This will *always* be clearly indicated by the INPUT meters' going into the red part of their scale.

If you are adjusting the 8600 during normal programming and cannot interrupt or distort the program to play program material from your studio at a much higher level than normal, follow the directions to:

- Calibrate while on air with normal programming: step (A) on page 2-28.

If you are able to interrupt or distort normal programming, you can achieve calibration that is more precise. Follow the directions to:

- Calibrate with unprocessed audio: step (B), page 2-28, or
- Calibrate with a Studio Level Control System that has a built-in 100% Calibration Tone, such as the Orban 6300, 8200ST-Studio Level Controller or 4000 Transmission Limiter: step (C), page 2-29, or
- Calibrate with an Orban 464A Co-Operator: step (D), page 2-29, or
- Calibrate with an Orban 1100 or 1101 OPTIMOD-PC

as appropriate.

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

A) Calibrate with program material and normal programming levels.

[Skip this step if you are calibrating in another manner.]

a) Adjust the CLIP LEVEL so that program peaks indicate approximately –15 dB on the input meters.

Observe the meters on the 8600 screen for a long period; be sure to observe live announcer voice. If this setting is misadjusted, distortion will result.

0 dB indicates input clipping on the 8600. These meters should never peak as high as 0 dB with program material; always leave a safety margin of headroom.

B) Calibrate with program material and worst-case programming levels (best method):

[Skip this step if you are calibrating in another manner.]

a) *Play program material from your studio at a much higher level than normal—turn the faders up all the way!*

This will usually produce the highest peak level output that your system can produce.

b) Adjust the 8600's CLIP LEVEL so that on program peaks the input meters indicate no more than approximately –2 dB.

0 dB indicates input clipping on the 8600. These meters should never peak as high as 0 dB with program material; always leave a safety margin of headroom.

C) Calibrate with a Studio Level Control System that has a built-in 100% Calibration Tone, such as the Orban 6300 Optimod-DAB, 8200ST-Studio Level Controller or 4000 Transmission Limiter:

[Skip this step if you are calibrating in another manner.]

a) Turn on the Studio Level Control System's 100% Calibration Tone.

On the Orban 4000 Transmission Limiter, press both of the 4000's front panel *TONE* buttons.

b) Adjust the output level of the Studio Level Control System for 100% modulation of the STL.

c) Adjust the 8600's CLIP LEVEL to indicate –2 dB on the input meters.

D) Calibrate with an Orban 464A Co-Operator:

[Skip this step if you are calibrating in another manner.]

The 464A does not have a built-in 100% tone. The easiest way to set the 8600 input peak clipping level is to temporarily re-adjust the 464A to produce clipped waveforms on program material to give a clear indication of peak clipping level.

a) Record the normal operating settings of the 464A.

b) Set both channels of the 464A controls as follows:

METER CAL	0
HF LIMIT PREEMPHASIS	set to preemphasis of your STL; if no preemphasis, set to 25μs
OUTPUT ATTEN	0
INPUT ATTEN	10
GATE THRESH	0
RELEASE TIME	0
REL SHAPE	SOFT
LEVEL	OFF
COMPR	OFF
HF LIMIT	OPERATE
SYSTEM	OPERATE
POWER	ON
MODE	DUAL

c) Play program material from your studio.

d) Adjust the 464A's METER CAL controls so that the 0 dB segment on the 464A's PEAK OUTPUT LEVEL meter just illuminates on program peaks.

e) Adjust the 464A's OUTPUT ATTEN controls to drive the STL to 100% modulation.

f) Adjust the 8600's CLIP LEVEL so that the program peaks indicate approximately –2 dB on the meter on the screen.

g) Return the 464A to the normal settings.

E) Calibrate with an Orban 1100 or 1101/1101e Optimod-PC:

[Skip this step if you are calibrating in another manner.]

Refer to the Optimod-PC manual for instructions on setting it up as an external AGC. You will usually use one of its three AGC presets.

Optimod-PC does not have a built-in 100% tone generator. The easiest way to set the 8600 input peak clipping level is to recall a "loud" preset in Optimod-PC that will produce substantial amounts of gain reduction in Optimod-PC's look-ahead limiter. This will produce frequent peaks at the maximum peak level at OPTIMOD-PC's output.

a) If you have customized your normal AGC preset in Optimod-PC, save it as a User Preset.

b) Apply program material at normal level to Optimod-PC's input.

c) Recall the IMPACT preset and verify that the Optimod-PC LIMITER meters indicate substantial gain reduction.

d) Adjust the 8600's CLIP LEVEL so that the program peaks indicate approximately –2 dB on the meter on the screen.

e) Recall your regular AGC preset in Optimod-PC.

5. Adjust the Analog Input's Reference Level.

[–9 dBu to +13 dBu (VU), or –2 to +20 dBu (PPM)] in 0.5 dB steps]

The REFERENCE LEVEL VU and PPM (Peak) settings track each other with an offset of 8 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 8600's gain reduction range to the level to which your studio operators peak their program material on the studio meters. This ensures that the 8600'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.

If you know the reference VU or PPM level that the 8600 will receive, set the REFERENCE LEVEL to this level, but do verify it with the steps shown directly below.

A) From the pop-up Menu, select the PRESETS screen.

B) Highlight the ROCK-MEDIUM preset.

C) Press the *Enter* button to select the preset.

D) Calibrate using Tone.

[Skip this step if you are using program material to calibrate the 8600 to your standard studio level. Skip to step (E).]

a) Verify EXTERNAL AGC is set to NO.

Refer to step 1 on page 2-27 above.

b) Feed a tone at your reference level to the 8600

If you are not using an external AGC, feed a tone through your console at normal program levels (typically 0VU if your console uses VU meters).

If you are using an Orban 4000 Transmission Limiter, press its two *TEST* buttons. Feed a tone through your console at the level to which you normally peak program material (typically 0VU if your console uses VU meters).

If you are using a Studio Level Controller that performs an AGC function, such as an Orban 8200ST OPTIMOD-Studio or 464A, adjust it for normal operation.

c) Adjust the REFERENCE LEVEL to make the 8600's AGC meters indicate 10 dB gain reduction.

d) When finished, reset EXTERNAL AGC to YES, if required (e.g., if that was its setting prior to setting REFERENCE LEVEL).

e) Skip to step 6.

E) Calibrate using Program.

[Skip this step if you are using tone to calibrate the 8600 to your standard studio level—see step (D) above.]

a) Verify EXTERNAL AGC is set to NO.

Refer to step 1 on page 2-27 above.

b) Feed normal Program material to the 8600

Play program material from your studio, peaking at the level to which you normally peak program material (typically 0VU if your console uses VU meters).

c) Adjust the REFERENCE LEVEL to make the 8600's AGC meters indicate an average of 10 dB gain reduction when the console's VU or PPM is peaking at its normal level.

If the AGC gain reduction meter averages less than 10 dB gain reduction (higher on the meter), set the REFERENCE LEVEL to a lower level.

If the AGC gain reduction meter averages more gain reduction (lower on the meter), set the REFERENCE LEVEL to a higher level.

d) When finished, reset EXTERNAL AGC to YES, if required (e.g., if that was its setting prior to setting the REFERENCE LEVEL).

6. Adjust the Analog Input's Right Channel Balance.

[Skip this step if the channels are already satisfactorily balanced.]

[–3 dB to +3 dB] on right channel only, 0.1 dB steps

Adjust RIGHT BALANCE to achieve correct left/right channel balance.

> This is not a balance control like those found in consumer audio products. This control changes gain of the right channel only. Use this control if the right analog input to the 8600 is not at exactly the same level as the left input. Be certain that the imbalance is not from a certain program source, but only through distribution between the console output and 8600 input. This is best accomplished by playing program material that is known to be monophonic, or by setting the mixing console into mono mode (if available).

7. **Adjust the Digital Input Reference Level and Right Balance controls.**

   If you will be using the digital input, set the input to DIGITAL and repeat steps 5 and 6 above using the REFERENCE LEVEL and RIGHT BALANCE controls for the DIGITAL section.

8. **Enable or Defeat the "Multipath Mitigator" Left/Right Skew Corrector.**

   See *"Multipath Mitigator" Left/Right Phase Skew Correction* on page 3-8.

   The Phase Corrector controls are found in the active Processing Preset. Because the Phase Corrector adds about 250 ms of delay to the audio path, if you do not wish to use it, we recommend defeating it in SYSTEM > UTILITIES via the PHASE CORRECTOR control. If you defeat it, this will override the PHASE CORRECTOR control in the active Processing Preset.

9. **Set response to an invalid or missing digital input signal. (optional)**

   Refer to step 17 on page 2-39.

10. **Set up and activate the BS.1770 Safety Limiter in the digital radio processing chain (optional)**

    The BS.1770 Safety Limiter prevents the BS.1770-3 integrated loudness in the digital radio processing chain from exceeding a value set by the BS.1770 LOUDNESS CONTROL THRESHOLD control. If your country requires that the integrated loudness of your digital radio broadcast to be constrained according to the BS.1770-3 loudness measurement algorithm, execute the steps below. *If your country does not require this, leave the BS.1770 Safety Limiter off.*

    A) Navigate to the INPUT/OUTPUT > HD RADIO screen.

    B) Set the BS.1770 LOUDNESS CONTROL THRESHOLD control to be equal to the target integrated loudness of your broadcast as specified by the regulatory authority in your country.

    C) Using the ON radio button, turn the BS.1770 SAFETY LIMITER control ON.

    D) To use the BS.1770 Safety Limiter, you must assign the output source of one of the Digital Outputs to HD. We suggest using Digital Output 1 to drive the digital radio transmitter.

    > To facilitate error-free loudness calibration through your transmission system, activating the BS.1770 Safety Limiter changes the behavior of the

Digital Output 100% PEAK LEVEL control. Normally, changing the setting of this control changes both the loudness and peak level of the signal appearing at the Digital Output driving your digital radio transmitter. However, when the BS.1770 Safety Limiter is turned ON, the Digital Output becomes the "Loudness Reference Output" and the Digital Output 100% PEAK LEVEL control sets *only the peak level, not the loudness*. In this case, the output loudness is always calibrated correctly with respect to 0 dBFS.

(**Note:** If both Digital Output 1 and Digital Output 2 are sourced from "HD," then Digital Output 2 becomes the "Loudness Reference Output" instead of Digital Output 1.)

For example, setting the Digital Output 1 100% PEAK LEVEL control to –3 dBFS always constrains the maximum peak level appearing at Digital Output 1 to –3 dBFS, but changing the setting of the Digital Output 1 100% PEAK LEVEL control does not change loudness. This is because when the BS.1770 Safety Limiter is turned ON, a second "hidden" gain control is inserted before the input of the peak limiter. The gain set by the "hidden" control complements the setting of the Digital Output 1 100% PEAK LEVEL control. For example, when the Digital Output 1 100% PEAK LEVEL control is set to –3 dBFS, the "hidden" control adds 3 dB of gain before the peak limiters to compensate for the 3 dB of loss that the 100% PEAK LEVEL control introduces after the peak limiters. Unless the peak limiters produce gain reduction, this arrangement keeps the overall gain constant from the input of the peak limiter to the Digital Output.

If the peak limiters produce enough gain reduction to reduce loudness audibly, this gain reduction will cause the output loudness to be lower than the setting of the BS.1770 LOUDNESS CONTROL THRESHOLD. This will never occur with BS.1770 reference levels normally used in digital radio (–18 LUFS) and digital television (–23 LUFS or –24 LUFS) because with these reference levels, any peak limiting that occurs will be too brief to audibly affect loudness.

Unless the 100% PEAK LEVEL control is set to 0 dBFS, this arrangement causes the output loudness to change when you toggle the BS.1770 SAFETY LIMITER control ON and OFF. In our example, the loudness will decrease by 3 dB when you set the BS.1770 SAFETY LIMITER to OFF.

E) After you have chosen a Processing Preset and adjusted it to your taste (step 20 on page 2-41, you must set the preset's HD FINAL LIMITER DRIVE to create the appropriate amount of gain reduction in the BS.1770 Safety Limiter. Then save the result as a User Preset. We recommend allowing the BS.1770 Safety Limiter to produce no more than 3 dB of gain reduction.

11. Set output and configuration level.

A) Navigate to the INPUT/OUTPUT > OUTPUT1 screen.

You can use either program material or tone to set your output level (and thus, your on-air modulation). If you want to use tone, turn the 400Hz calibration tone on.

B) Set the ANALOG OUT SOURCE to FM or FM+DELAY, depending on whether you need to apply diversity delay to that source. See *Diversity Delay* on page 2-75.

Be aware that this setting can be toggled between FM and FM+DELAY by several means in addition to the ANALOG OUT SOURCE control in the

INPUT/OUTPUT > OUTPUT1 screen. These additional means include the 8600's GPI inputs (page 2-60), its serial ports (page 2-48), its Ethernet input (via 8600 PC Remote or a terminal program), and its clock-based automation (page 2-42).

C) Set the ANALOG OUTPUT PRE / FLAT control to PRE-E (for preemphasis) or FLAT.

[Skip this step if you will not be using the analog output.]

If you will use the analog output to drive a stereo encoder, PRE-E provides the best performance because this stereo encoder does not have to restore the preemphasis. However, if you cannot defeat the preemphasis in your stereo encoder or if you will use the analog output for monitoring, set the output FLAT.

If you are sending the analog output of the 8600 through a digital link that uses lossy compression (like MPEG, APT-X, or Dolby), set the output FLAT. Lossy codecs cannot handle pre-emphasized signals.

[Skip steps (D) through (I) if you will not be using the"AES1" digital output. Note that both the "AES1" and "AES2" digital outputs have identical functionality, so you could use either one in the steps below.]

D) Set the AES1 OUT SOURCE control to the desired source: FM, FM+DELAY, HD, or MONITOR.

In most facilities, you will set it to FM.

HD is not available in 8600FM units.

E) Set the AES1 PRE/FLAT control to PRE-E (for preemphasis), FLAT, or PRE-E+J17.

See the notes immediately above.

PRE-E+J17 applies both FM preemphasis and J.17 preemphasis (in cascade) to the signal and is only used with STLs using J.17 preemphasis when their own J.17 preemphasis filters are bypassed. These are rare.

F) Set the AES1 output SAMP RATE (sample rate).

[32], [44.1], [48], [88.2], or [96], in kHz

The 8600's base sample rate is always 64 kHz, but the internal sample rate converter sets the rate at the 8600's digital output. This adjustment sets the 8600's output sample rate to ensure compatibility with equipment requiring a fixed sample rate.

G) Adjust the AES1 SR SYNC control.

[Internal / Sync In / Input]

You can lock the sample rate of each of the 8600's AES3 outputs to the sample rate of a reference AES3 signal that is applied to the AES3 input or to an AES11 signal that is applied to the SYNC input.

If you wish to operate the two AES3 outputs at different sample rates, only one output can be synced to the signal at the SYNC input. However, in this case the other output could be synced to the signal appearing at the digital input.

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 8600'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 8600'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 8600 digital input or SYNC IN, set SR SYNC to INTERNAL and select the desired output sample rate.

MPX hardware provides a sync reference input (labeled REF IN) on a female BNC jack at the end of the breakout cable that connects to the digital composite connector. It accepts a 1x 5V p-p squarewave wordclock signal at 32, 44.1, 48, 88.2, or 96 kHz, or a 10 MHz sinewave or squarewave signal, 0.5 to 5 V peak. 10 MHz is a common output frequency produced by GPS and rubidium frequency standards. You can configure the 8600 to lock its 19 kHz pilot tone and output sample frequency to this input.

When a valid reference signal is applied to this input, it locks the Optimod's Internal DSP clock to this input, so if a given AES3 L/R output's sync source is set to Internal, the sample frequency at that output will be locked to this reference signal. This will also lock the 192 kHz sample rate of the digital composite output and the pilot tone frequency appearing at both the analog and digital composite outputs to the reference signal.

See *Digital Composite Outputs* on p. 2-11, *Digitized Subcarrier Inputs* on p. 2-11, and *Wordclock/10 MHz Sync Reference Input* on p. 2-11

H) Set the desired output WRD LENGTH (word length).

[14], [16], [18], [20], or [24] bits

The largest valid word length in the 8600 is 24 bits. The 8600 can also truncate its output word length to 20, 18, 16 or 14 bits. The 8600 can also add dither and you should set it to do so if the input material is insufficiently dithered for these lower word lengths. (See the next step.)

I) Adjust DITHER to IN or OUT, as desired.

[In] or [Out]

It is wise to leave this control set to IN.

When set to IN, the 8600 adds "high-pass" dither before any truncation of the output word. The amount of dither automatically tracks the setting of the WORD LENGTH control. This is first-order noise-shaped dither that reduces added noise in the midrange considerably by comparison to white PDF dither. However, unlike extreme noise shaping, it adds a max-

imum of 3 dB of excess total noise power when compared to white PDF dither. Thus, it is a good compromise between white PDF dither and extreme noise shaping.

J) If you are using the AES2 output, repeat *steps (D) through (I)* for this output. Its controls are located on the INPUT/OUTPUT > OUTPUT2 screen.

K) If you are using the Ratings Encoder Loop-through feature, perform the following steps (see *Connecting a Ratings Encoder* on page 2-8):

a) Set the AES2 OUT SOURCE control to AGC>RATINGS or LIMIT>RATINGS.

b) Connect your Optimod's AES2 output to the input of the ratings encoder.

c) Connect the ratings encoder's output to your Optimod's SYNC INPUT.

Depending on whether you choose AGC>RATINGS or LIMIT>RATINGS, the SYNC input is repurposed automatically to drive the input of the FM/HD split point or your Optimod's stereo encoder. AGC applies a controlled audio level to the encoder and allows you to use one ratings encoder for both FM and HD; LIMIT applies the highest possible audio level to the ratings encoder but requires two ratings encoders if you are transmitting HD.

d) Set the AES2 output SAMP RATE to 48 kHz.

e) Set the AES2 output WRD LENGTH to 24-bit.

- In Ratings Encoder Loop-through mode, you cannot use your Optimod's SYNC INPUT to receive a sync reference signal. You can still lock your Optimod's output sample rate to a signal applied to its AES3 input.

- The Ratings Encoder Loop-through mode has a standard fallback strategy: To prevent an audio mute, your Optimod will automatically bypass the loop-though if there is no valid AES3 signal at your Optimod's Sync Input. When this happens ratings encoding does not occur, so warnings will flash on both the Optimod's front panel and in PC Remote (if previously connected to your Optimod). Nevertheless, we recommend monitoring the presence of valid ratings encoding via the monitor supplied by the ratings provider.

L) Using a modulation monitor or modulation analyzer, adjust the outputs you are using (analog and/or digital) to make the modulation monitor read 100% modulation (usually ±75 kHz deviation).

If you are using program material, make sure that the program material is loud enough to produce peaks of frequent recurrence that hit the 8600's peak limiting system, thereby defining the maximum peak level that the 8600 will produce. In the U.S., we recommend using 900μs peak weighting on the peak modulation indicator, as permitted by F.C.C. rules. This will cause the monitor to ignore very low energy overshoots and will result in the highest peak modulation permitted by law.

In other countries, use a peak-indicating instrument as specified by the regulatory authority in your country.

If you are required to implement the average modulation limits specified by ITU-R 412-7, you may seldom see peaks hitting ±75 kHz deviation. In

this case, we advise you to set the output level using the 8600's reference 400Hz tone.

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). This implies that you must set the 8600's composite output level for the equivalent of 90% modulation, not counting the subcarriers. (90% + 20% = 110%.) This will mean that pilot injection will be about 8% modulation instead of the desired 9%. From the Input/output > Composite screen, adjust PILOT LEVEL control as necessary to produce 9% modulation (±6.75 kHz deviation). This will ordinarily require you to set the PILOT LEVEL parameter to "10%."

12. Set preemphasis mode of Output meters (optional).

When the METER SELECT switch (in INPUT/OUTPUT > HD DIGITAL RADIO) is set to FMOUTLEVEL, the 8600's Output meters normally indicates the peak output level of the FM analog processing, which is always pre-emphasized. If you wish to apply de-emphasis before these meters, navigate to INPUT/OUTPUT > UTILITIES and set the FM OUTPUT METER to DE-EMPH.

The default is PRE-EMP.

13. Set the polarity of the analog FM processing (optional).

In HD Radio installations, this command is useful when switching the 8600 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.

A) Navigate to the INPUT/OUTPUT > UTILITES screen.

B) Set the FM POLARITY control to POSITIVE or NEGATIVE as appropriate for your transmitter.

14. Configure composite outputs.

[Skip this step if you are not using the composite baseband outputs.]

A) Navigate to the INPUT/OUTPUT > COMPOSITE screen.

B) Adjust the composite level at the composite output(s) you are using so that the FM carrier is modulated to 100% on modulation peaks. Alternately, you can use tone.

If you are using subcarriers, this screen allows you to specify the amount by which the 8600 automatically reduces main and stereo subchannel modulation to accommodate the subcarrier within the modulation limits specified by the governing authority.

See step (8.C) on page 2-22 for a more detailed discussion.

C) Set the MODULATION MODE to STEREO

In addition to STEREO. the MODULATION MODE control also allows you to set the modulation mode to MONO-L, MONO-R, and MONO-SUM (mono modu-

lation sourced from the left input, right input, or sum of the left and right inputs). For testing, it also offers a PILOT OFF mode, which transmits a stereo waveform with no pilot tone.

SSB will operate the stereo encoder in its experimental single-sideband mode (see *SSB Stereo Encoder Operation* on page 3-11) and will activate a peak modulation controller to prevent composite modulation from exceeding 100% when this controller is used in conjunction with the 8600's composite limiter. We recommend setting the *Composite Limit Drive* control to 0dB for SSB operation. This control is part of the active processing preset and can be accessed from either the Intermediate Control or Advanced Control screen.

D) Set the PILOT LEVEL to 9% modulation.

If you have to reduce the setting of the COMPOSITE LEVEL control to accommodate overshoots in the transmission path following the 8600 (including the transmitter), you may have to increase the setting of the PILOT LEVEL so that the transmitted pilot is still at 9% modulation.

E) Set the DIVERSITY DELAY control to IN or OUT as appropriate for your installation.

Digital Composite Setup

In MPX hardware, there is one AES3 digital composite output at 192 kHz sample rate. Its sample rate is locked to the 8600's internal clock unless you have applied a valid clock signal to the REF INPUT BNC on the Digital Composite daughter board, in which case it will be locked to this clock, as will the stereo pilot tone frequency. See *Wordclock/10 MHz Sync Reference Input* on page 1-11.

F) If you will be using subcarriers, connect your subcarrier generators to the digitized SCA inputs, located on the breakout cable.

G) Navigate to the INPUT/OUTPUT > COMPOSITE2 screen.

H) Adjust the SCA INPUT 1 and SCA INPUT 2 controls to produce the desired amount of subcarrier injection.

Refer to step 8 on page 2-22 for more information on setting modulation levels.

I) Adjust the COMPOSITE OUT control to achieve the desired modulation level.

In 8600 PC Remote, this control is located in I/O SETUP > DIGITAL ENCODER.

15. Set up a low-delay headphone monitoring system (optional).

If you are using an "MX" preset and/or using the diversity delay, it is impossible for talent to monitor the signal off-air while speaking because the delay is too large. To provide a talent headphone feed, you can configure any of the 8600's left/right outputs (analog, AES1, and AES2) to emit the low-delay monitor signal. 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. The input/output delay is approximately 5-10 milliseconds (depending on the setting of AGC CROSSOVER TYPE) and should be comfortable for talent. See the discussion in *Monitoring on Loudspeakers and Headphones* on page 1-26.

To configure any output for Low-Delay Monitoring:

A) Locate to the INPUT/OUTPUT > OUTPUT1 screen to configure the analog or AES1 outputs. Locate to the INPUT/OUTPUT > OUTPUT2 screen to configure the AES2 output.

B) Choose OUT SOURCE > MONITOR.

C) Set the preemphasis control to FLAT for the output you are using for monitoring.

If you use the low-delay output to drive both your studio monitor speakers and talent headphones (which may be necessary if your console has only one monitor input for both), we recommend connecting a loss-of-carrier alarm to one of the 8600's GPI inputs and programming this input to mute the monitor output in the event that carrier is lost. This simulates normal "off air" monitor functionality and immediately alerts the staff if the transmitter goes off the air unexpectedly.

You can program any GPI terminal for Monitor Mute functionality from SYSTEM SETUP > NETWORK/REMOTE 1 (the REMOTE screen). The OUT SOURCE parameter located in the INPUT/OUTPUT > OUTPUT1 screen needs to be set to MONITOR to make the Monitor Mute feature work.

16. Defeat the final clipper (optional).

If you are using the 8600 to drive a network with protection audio processors (like Orban's Optimod-FM 2300 or 5500) at each transmitter, you may wish to defeat the 8600's final clipper to prevent double clipping, which can unnecessarily increase distortion on-air. To do this:

A) Put a non-MX preset on air. (Clipper defeat is not supported in MX presets.)

B) Locate to the INPUT/OUTPUT > UTILITIES screen.

C) Set FINAL CLIP DEFEAT to DEFEAT.

This will also defeat the overshoot compensator.

Note that defeating the final clipper and overshoot compensator will increase peak output levels, possibly to the point where the 8600's output amplifier and/or digital output clips. It is wise to customize any presets you are using by reducing the FINAL CLIP DRIVE and OSHOOT COMP DRIVE controls to their minimum values. This will greatly reduce the probability of inadvertently clipping the 8600's output.

Unless defeated, the 8600's composite limiter will continue to control the composite output level. However, you should not use the composite output when the 8600's final clipper is defeated because this will remove the advantages of the distortion cancellation in the 8600's final clipper. (The composite limiter does not cancel distortion.)

17. Program Silence Sense (optional)

You can program the 8600 to switch automatically from its digital input to its analog inputs if the INPUT SOURCE is set to DIGITAL and the signal at the digital input falls silent. Conversely, you can also program the 8600 to switch automatical-

ly from its analog inputs to its digital input if the INPUT SOURCE is set to ANALOG and the signal at the analog input falls silent.

There are two silence detectors, one for the analog input and one for the digital input. The silence sense parameters apply to both simultaneously. Both detectors are available to drive the 8600's tally outputs, and either silence detector can be used to instigate automatic input switching. (See *Tally Output Programming* on page 2-62.)

Silence sense will be activated if either channel falls silent, protecting against "loss-of-one-stereo-channel" faults.

If silence is detected at both the analog and digital inputs (as in the case of a studio operational fault), automatic switching will not occur.

When an active signal is restored to the preferred input, the 8600 will automatically switch back to that input.

When the digital input is in the silence detect fallback state, the unit and PC will show the input as being DIGSILENCE; when the analog input is in silence detect fallback, they will indicate ANASILENCE.

A) Navigate to INPUT/OUTPUT > SILENCE DETECT.

B) Set the SILENCE THRESHOLD to the level below which the 8600 will interpret the input as being silent.

This setting is with respect to the current analog reference level and digital reference level.

C) Set the SILENCE DELAY to the amount of time that the input must be below the SILENCE THRESHOLD before the 8600 automatically switches to the analog input.

D) Set the ANALOG FALLBACK to YES if you wish the 8600 to automatically switch from the digital to analog input when silence is detected. Set the control to NO to defeat automatic switching.

E) Set the DIGITAL FALLBACK to YES if you wish the 8600 to automatically switch from the analog to digital input when silence is detected. Set the control to NO to defeat automatic switching.

18. Set the PILOT REFERENCE control

This control is located in the INPUT/OUTPUT > COMPOSITE screen. It determines the phase relationship between the 19 kHz pilot reference output (see page 2-10) and the pilot tone present in the composite output. 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. 180 DEG and 270 DEG take into account installations where there is a polarity reversal in the RDS generator.

19. End Analog and Digital I/O setup.

If you are using an external AGC and you temporarily set the EXTERNAL AGC to NO in step 1 on page 2-27, set the EXTERNAL AGC to YES.

When you are finished adjusting input/output parameters, press the *Escape* button to return to the Meters screen.

20. Select a processing preset.

This step selects the processing to complement the program format of your station.

After this step, you can always select a different processing preset, program the 8600 to automatically change presets on a time/date schedule, modify presets to customize your sound, and store these presets as User Presets.

A) Navigate to the Presets screen.

a) Press the control knob to display the pop-up Menu.

b) Turn the knob to highlight PRESETS and press the knob.

B) Using the *Locate* joystick up/down control or turning the control knob, highlight a preset corresponding to your format. Press *Enter* to put the highlighted preset on the air.

> For a discussion of the presets, see *Factory Programming Presets* starting on page 3-21. Preset names are *just 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.
>
> You can easily modify a preset with the 8600's one-knob LESS-MORE feature. Navigate to the Basic Modify screen. (There may be a short delay while the 8600 computes Less-More values.) If you do not see the LESS-MORE screen immediately, press and hold the *Locate* joystick to the right or left until you find the screen. Turning LESS-MORE up will produce more loudness but also more processing artifacts like distortion and unpleasant density. Turning LESS-MORE down will make the sound cleaner, more open, and easier to listen to, but will also make it quieter.

C) If you are using the digital radio processor's BS.1770 Safety Limiter (step 10 on page 2-32) you must set your chosen processing preset's HD FINAL LIMITER DRIVE to create the appropriate amount of gain reduction in the BS.1770 Safety Limiter. Then save the result as a User Preset. We recommend allowing the BS.1770 Safety Limiter to produce no more than 3 dB of gain reduction.

> BS.1770-3 is an imperfect metric. It has been established by listening tests that BS.1770-3 can over-indicate loudness by 3 dB or more when the program material has an unusually low peak-to-RMS ratio because of production techniques, dynamic compression, etc. Whenever the BS.1770 meter would otherwise read above 0 LU, the BS.1770 Safety Limiter produces gain reduction that is the inverse of the BS.1770 meter's reading. Hence, the Safety Limiter can cause such material to be unnaturally quiet sounding compared to speech or other program material with higher peak-to-RMS ratios, so we recommend using as little gain reduction as possible in the BS.1770 Safety Limiter. We believe that the Optimod processing sounds most natural with no BS.1770 limiting at all, so you should only use it if government regulations in your country require it.

D) If you are using the analog FM processing chain's BS.1770 Safety Limiter and/or its MPX Power Controller, you must adjust the MULTIPLEX POWER OFFSET control to obtain an appropriate amount of gain reduction. See step 13 on page 2-25.

Using Clock-Based Automation

1. **If you have not already done so, set the system clock.**

   If you can connect your 8600 to the Internet through its Ethernet port, you can then specify an Internet timeserver to set your 8600's clock automatically. In addition, Optimod PC Remote software can automatically set your Optimod's local time, OFFSET, and TIME SERVER to reflect the Windows settings in the machine running PC Remote software. See *Synchronizing Optimod to a Network Time Server* on page 2-69. If you are planning to set your Optimod's time via PC Remote and/or the Internet, skip to step (C).

   A) Navigate to SYSTEM SETUP > PLACE / DATE / TIME.

   B) Navigate to the TIME AND DATE screen.

      a) Choose TIME FORMAT as desired (either 24-hour time or AM / PM-style time).

      b) Set hours, minutes, and seconds, in that order.

         Seconds will stop advancing when you set hours and minutes. So set seconds last.

      c) Choose the desired date format.

      d) Set today's date.

      e) If you want the clock to automatically reset itself to conform to Daylight Saving Time (Summer Time), use the BEGINS and ENDS fields to specify when Daylight Saving Time begins and ends in your area. If you do not wish to use this feature, leave any BEGINS and ENDS field OFF.

   C) (Optional) > Navigate to the STATION screen to specify your station's identifier (call sign or call letters).

2. **Navigate to System Setup / Automation.**

3. **If the far left button reads "Disabled," choose it and press Enter to enable automation.**

   This button lets you enable or disable all automation events easily without having to edit individual automation events.

4. **To add an automation event:**

   A) Select ADD.

   B) You can program an event that occurs only once or an event that occurs in a weekly preset pattern. Highlight either SET BY WEEK or SET BY DATE and press the ENTER button.

   C) For SET BY WEEK:

      a) Navigate to the each day of the week in turn; then use the knob to turn the day on or off.

         You can program the event to occur on as many days as you wish.

b) Navigate to the Event Time field and set the hour, minute, and second when the automation event is to occur.

Automation events have a "start" time but no "stop" time. The 8600 will stay in the state specified by an existing automation event indefinitely, until its state is changed by another automation event or by another action (like a user's interacting with the front panel or with the PC Remote application).

c) Navigate to the Event Type field and set the desired event. You can recall any factory or user preset and can activate BYPASS mode (for scheduled network testing) or EXIT TEST. Other automation events include: STEREO, MONO-R, MONO-L, MONO-SUM, DIVERSITY DELAY IN, DIVERSITY DELAY OUT, MOD REDUCTION 1, MOD REDUCTION 2, and EXIT MOD REDUCTION.

D) For SET BY DATE, set the desired date and time for the event and specify the Event Type.

E) Choose DONE and press ENTER.

You will return to the automation event list. *You may have to scroll the list (using the knob) to see the event that you just added*.

5. **To edit an existing event:**

A) Using the knob, highlight the event you wish to edit.

B) Select the EDIT button and press *Enter*. The edit screen appears.

C) Edit the event as desired.

D) When you have finished making edits, choose DONE and press *Enter*.

6. **To delete an event:**

A) Highlight the event to delete with the knob.

B) Choose DELETE and press *Enter*.

7. **Choose DONE and press Enter to leave the Automation screen.**

Security and Passcode Programming

You can use multi-level passcodes to control access to the 8600 via the front panel and via PC Remote. You can configure a given passcode to allow one of the following levels of access:

1. All Screens (i.e., administrator level)
2. All Screens except Security
3. All screens except Modify and Security
4. Presets, Modify, Save, Memory, and Automation
5. Presets and Automation
6. Presets

Only passcodes with ALL SCREENS access let you do software updates and set passcode permissions.

Each Passcode is unique; the software will not let you create duplicate Passcodes. Further, to prevent accidental lockout, the software requires you to have at least one passcode with ALL SCREENS (administrator) privileges.

Your Optimod secures User Presets by encrypting them (using the Advanced Encryption Standard algorithm with the session passcode as its key) when PC Remote fetches them. Hence, a packet sniffer cannot intercept User Presets in plaintext form. PC Remote then writes the fetched User Presets in encrypted form on your hard drive, where they remain for the duration of your PC Remote session.

If PC Remote exits normally, it will erase these temporary User Preset files from your computer's hard disk. If it does not exit normally, these files will remain in encrypted form. However, the next time that PC Remote starts up, it will automatically clean up any orphaned files.

1. **From the main menu, navigate to SYSTEM SETUP and then to SECURITY.**

   The Security screen lets you set front-panel lockout time, create new passcodes, review and/or assign authorization levels for existing passcodes, and delete passcodes.

   If the 8600 is already under security control, you must enter an ALL SCREENS-level passcode to enter the Security screen.

2. **Set the Security Screen "Lockout" parameter.**

   The choices are 1, 5, 15, or 30 minutes, 1, 2, 4, or 8 hours, or OFF.

   Front Panel lockout only occurs when the lockout value is not OFF.

   The Lockout field sets the time delay between the last user interaction with the front panel and automatic front-panel lockout. Once the front panel is locked out, you can only regain access by entering a valid passcode.

   The Lockout field does not affect PC Remote connections. Once connected, the PC Remote application does not time out automatically; it remains connected until explicitly disconnected by its user.

   The lockout timer begins at the top of the next minute. For example, if you set Lockout to be 1 minute at 9:10:33 AM and do not touch the front panel again, the front panel will lock out at 9:12:00 AM.

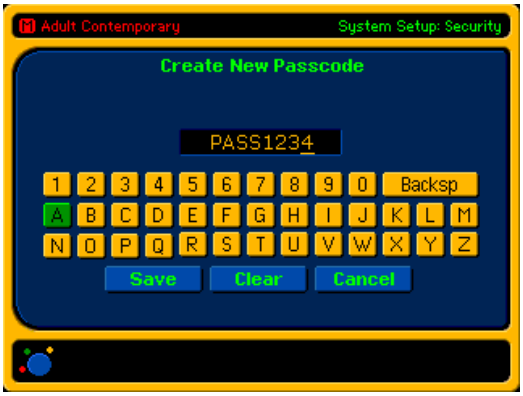
3. **Set the Security Screen "View Meters" parameter.**

   Select YES to display meters and NO to hide meters when lockout is active.

4. **Create a new passcode (optional).**

   A) Select the "New" button from the Security screen.

   The "Create New Passcode Screen" appears.



   B) Use the "virtual keyboard" to create a passcode.

   Use the LOCATE button to navigate to each character. Then press ENTER to accept that character.

   The letters on the virtual keyboard are all uppercase. When you use the passcode later, you must enter it using capital letters because the passcode is case sensitive. For example, if you set up your passcode as OOPS25, you must enter it as OOPS25, not as oops25 or Oops25.

   C) When you have finished creating your passcode, write it down so you do not forget it.

   D) Choose SAVE. The Security screen reappears.

   E) Initially, your new passcode has ALL SCREENS (administrator) privileges. To change its privileges, navigate to the PASSCODE AUTHORIZES ACCESS TO field. Then turn the knob to choose the desired privilege level.

   F) Choose DONE or press *Escape* when you are finished. The System Setup screen appears.

5. **Edit or delete an existing passcode (optional).**

   A) Navigate to SYSTEM SETUP and then to SECURITY

   If the 8600 is already under security control, you must enter an ALL SCREENS-level passcode to enter the Security screen.

   B) Navigate to the CURRENT PASSCODE field. Use the blue knob to scroll through the passcodes until you see the one you wish to edit or delete.

   C) To delete the passcode, choose the DELETE button.

   At least one passcode must have "All Screens" privileges. If you try to delete the last "All Screens" passcode, the following dialog box will appear:

   > You cannot delete this Passcode because you must have at least one Passcode with All Screens privileges. Press OK to continue.

   D) To edit the passcode, navigate to the PASSCODE AUTHORIZES ACCESS TO field. Then turn the knob to choose the desired privilege level.

   E) Choose DONE when you are finished. The System Setup screen appears.

   You may edit or delete more than one passcode before choosing DONE.

Choosing DONE on the Security Screen automatically saves all of your Passcode settings.

To Unlock the Front Panel

A) On the 8600 front panel, operate any button or the knob.

The ENTER PASSCODE screen appears.

B) Enter your passcode using the virtual keyboard.

If you enter a Passcode that does not exist, you are returned to the ENTER PASSCODE screen.

C) Choose UNLOCK.

You will be able to access 8600 functions allowed by the privilege level of your passcode.

After you have finished working, the panel will automatically re-lock after the time delay set in step 2 on page 2-44. Provided you have ALL SCREENS privileges, you can set the delay as desired by following the instructions in that step.

8600 User Interface Behavior during Lockout

Meters are not visible during lockout unless you enter a passcode of any privilege level. Instead, a Lockout screen replaces the Meters screen. It displays Input Status, Time, Date, Studio Name, Mod. Reduction Status, and Help Text.

The On-Air Preset and Meters do not appear in order to prevent your competitors from seeing them if your 8600 is installed in a shared facility.

The diagnostic screens are unavailable during lockout unless you enter a passcode of any privilege level.

Default ADMIN Passcode

When you first open to the Security screen on the 8600, there is one default passcode: ADMIN (all capitals), which has ALL SCREENS privileges. This passcode permits an initial connection to the 8600 and 8600X via PC remote; you must enter ADMIN when PC Remote asks you for a passcode.

The 8600X has a blank front panel. Only an external PC running the 8600 PC Remote application can control it.

The front panel lockout feature's default setting is OFF, so standard 8600s (with full-featured front panels) will not have the lockout feature functioning until a lockout time is set.

Any passcode you have programmed into the 8600 (via step 3 on page 2-44) allows PC Remote connections with the same privileges. For example, if you connect to the PC Remote and use a Passcode with ALL SCREENS access, this Passcode will allow full access to the 8600 from that PC. Conversely, if you connect to the 8600 with a Passcode that only allows access to the "Presets" on the 8600, you will only be able to recall presets from the PC Remote.

To ensure good security, you should first create a new ALL SCREENS passcode and then delete the ADMIN passcode (in that order) to prevent others from accessing your 8600 with the ADMIN passcode. The longer a passcode is, the more secure it is. Moreover, the most secure passcodes use a random combination of letters and numbers.

Security and Orban's PC Remote Application

Any passcodes set on the 8600 will allow the PC Remote application to connect via direct, modem and Ethernet connections at the level authorized by the passcode.

- *If no Passcodes are assigned to 8600 except the ADMIN default Passcode:*

  When you attempt a connection to the 8600 via Direct, Modem, and Ethernet connections, the "Enter Passcode Screen" will prompt you to enter a Passcode. Type in ADMIN from your keyboard. This will allow you full access to the 8600 via the PC Remote.

  To ensure that your 8600 is fully protected, create a new passcode that has ALL SCREENS access. Then delete the ADMIN passcode.

  See step 3 on page 2-44 for instructions on how to create a new passcode and step (5.C) on page 2-45 for instructions on how to delete a passcode.

- *Using passcodes to end PC Remote connections from the 8600 front panel:*

  If you try to access an 8600 from its front panel while a remote connection exists, a message will appear asking you whether you want to disconnect the remote connection. If you choose to disconnect the connection, the "Enter Passcode Screen" will appear if the unit is locked out.

Passcodes and Software Updates

PC Remote allows a software update to occur regardless of passcode level of the 8600 ↔ PC Remote connection. However, PC Remote will *only* offer to perform a software update if the version of PC Remote higher than the version of the software installed in your 8600. Hence, this does not create a significant security issue.

If you have forgotten your "All Screens" passcode…

You can access the 8600 even if you have forgotten your ALL SCREENS passcode.

A) Press the ENTER button within two seconds after the 8600 displays its "Please wait while Optimod initializes" screen upon boot-up.

B) Choose whether you want the 8600 to reset all passcodes while retaining other customizations (like I/O levels and user presets) or if want the 8600 to reset itself to its factory defaults. In either case, all existing passcodes will be erased.

- If you choose to reset the unit to factory defaults, the 8600 will subsequently ask whether it should erase all user presets or retain them.

- If you reset only the passcodes, the front panel will not unlock automatically. After passcode reset, there will be one passcode, ADMIN, with All Screens privileges. Use this passcode to unlock the front panel normally.
- If you reset the unit to its factory defaults, the panel will unlock automatically. Please note that resetting the unit to its factory defaults:

  - Resets all global parameters to factory default settings
  - Deletes all Automation Events
  - Restores Remote Interface inputs 1-8 to "no function"

Administering the 8600 through its Serial Ports or Ethernet

You can connect a PC to the 8600's serial ports or to its Ethernet port by using a terminal program like HyperTerminal to administer security and to recall presets using simple ASCII commands. The behaviors of Serial Ports #1 and #2 are different. This section gives details.

For both ports:

- Valid commands are in either upper or lower case, not a combination.
- Only one valid command is permitted per line.
- The 8600 will not respond to unrecognized commands.
- The character code supported is ASCII.

Connecting via Serial Port #2 Using a Terminal Program on a PC

- The 8600's Serial 2 interface can be used with any computer or terminal that is compatible with the RS-232 standard interface.

  Unlike the 8600's Ethernet and Serial 1 ports, Serial 2 does not use the TCP/IP and PPP protocols.

- Users will connect their computer or terminal to the 8600 with the supplied null modem cable. Only direct connections are supported; there is no provision for communications via modem at Serial 2.
- Communications configuration is 9600, N, 8, 1, no handshaking (flow control = none).
- To facilitate maintaining security at sites shared with others, the 5500 monitors Serial 2 for 30 minutes after power-up or after the last valid command is re-

ceived, after which the security, restore and networking commands at Serial 2 are no longer available.

If you have forgotten your "All Screens" passcode on page 2-47 provides a simple means to regain access to an 8600 from which you are locked out. You can also do this via a PC running a terminal program like HyperTerminal.

A) Connect an available RS232 serial port (COM port) on your computer to Serial Port #2 on the 8600.

> You do not need to remove power from either your computer or the 8600 when you do this.

B) Start HyperTerminal. (You can usually access it from Start / Programs / Accessories / Communication.)

The NEW CONNECTION dialog box appears.

C) Give your new connection a name and choose OK.

The CONNECT TO dialog box appears.

D) Set the CONNECT USING field to "Direct to COMx," where "x" is the COM port you are using on your PC.

E) Choose OK.

The PORT SETTINGS dialog box appears.

F) Set the port properties as follows:

Bits per second 9600
Data Bits.....8
Parity .....none
Stop bits.....1
Flow control.....none

G) Choose OK.

H) Navigate to File / Properties / Settings / ASCII Setup. Set the ASCII Setup properties as follows:

Check:
- Send line ends with line feeds
- Echo typed characters locally
- Wrap lines that exceed terminal width

Uncheck:
- Append line feeds to incoming line ends
- Force incoming data to 7 bit ASCII

Leave "Line delay" and "Character delay" at their default values.

I) Activate the CAPS LOCK on your computer to ensure that you type in uppercase.

You can now type in commands described in the specification in *Administrative Operations Available via Serial Port #2* below.

Administrative Operations Available via Serial Port #2

In the following tables of commands and responses:

- Text that the user enters appears in **MONOSPACED BOLD**.
- Responses that the 8600 transmits appear in `monospaced normal`.
- The symbol "↵" means CR (for received commands) and CR+LF (for transmitted responses from 8600).

1. To restore factory defaults:

Command	Response
RESTORE FACTORY DEFAULTS↵	`Are you sure (yes / no)?↵`
YES↵	(factory defaults restored) `Restored↵`
NO↵ (or any response other than **YES↵**")	(abort) `Defaults not restored↵`

To protect against accidental loss of settings, you must enter the entire command string and a "YES" response in uppercase.

Restoring factory defaults does the following:

- Deletes all User Presets
- Resets all global parameters to factory default settings
- Deletes all Automation Events
- Restores Remote Interface inputs 1-8 to "no function"
- Clears all passcodes

2. To unlock an 8600 / 8600X:

The following command assigns an ALL SCREENS passcode. This passcode is then available from the front panel or when you connect normally via the 8600PC application (through the 8600's Serial Port #1 or optional Ethernet connections).

Command	Response
PW ########↵	(valid passcode entry)`Accepted↵` (invalid passcode entry) `Denied↵`

Valid arguments follow the same rules for passcode entries made from the front panel and via 8600PC:

- Passcode length must be 1 to 8 characters.
- Only alphanumeric characters are allowed (0…9 and A…Z). No punctuation or extended characters are allowed.
- Lower case letters included in the argument will automatically be converted to upper case.

3. **To change the IP Address:**

Command	Response
IP XXX.XXX.XXX.XXX↵	(valid IP address) IP: xxx.xxx.xxx.xxx entered↵ (invalid IP address) ERROR. Using IP: yyy.yyy.yyy.yyy↵
IP?↵	Using IP: yyy.yyy.yyy.yyy↵

In the above table: **XXX.XXX.XXX.XXX** = the specified IP address;
yyy.yyy.yyy.yyy = the present (or default) IP address in use.

192.168.254.254 is the factory default.

Any out-of-range or invalid characters render invalid the whole IP address that you entered.

4. **To change the Subnet Mask:**

Command	Response
SN XXX.XXX.XXX.XXX↵	(valid subnet) SN: xxx.xxx.xxx.xxx entered↵ (invalid subnet) ERROR. Using SN: yyy.yyy.yyy.yyy↵
SN?↵	Using SN: yyy.yyy.yyy.yyy↵

In the above table: **XXX.XXX.XXX.XXX** = the specified subnet mask;
yyy.yyy.yyy.yyy = the present subnet mask in use.

255.255.255.0 is the factory default.

Valid subnet masks are defined according to existing standards. Any out-of-range or invalid characters render the whole argument invalid.

5. **To change the Gateway:**

Command	Response
GW XXX.XXX.XXX.XXX↵	(valid gateway) GW: xxx.xxx.xxx.xxx entered↵ (invalid gateway) ERROR. Using gw: yyy.yyy.yyy.yyy↵
GW?↵	Using GW: yyy.yyy.yyy.yyy↵

In the above table: **XXX.XXX.XXX.XXX** = the specified gateway;
yyy.yyy.yyy.yyy = the present gateway in use.

Valid gateways are defined according to existing standards. Any out-of-range or invalid characters render the whole argument invalid.

6. **To change the Port:**

Command	Response
PO XXXXX ↵	(valid port) PO: xxxxx entered↵ (invalid port) ERROR. Using PO: yyyyy↵
PO?↵	Using PO: yyyyy↵

In the above table: **xxxxx** = the specified port;
yyyyy = the present port in use.

Port 6201 is the factory default.

Valid ports are defined according to existing standards. Any out-of-range or invalid characters render the whole argument invalid.

7. **To change the Terminal Port:**

Command	Response
TP XXXXX ↵	(valid terminal port) TP: xxxxx entered↵ (invalid terminal port) ERROR. Using TP: yyyyy↵
TP?↵	Using TP: yyyyy↵

In the above table: **xxxxx** = the specified terminal port;
yyyyy = the present terminal port in use.

Port 23 is the factory default.

Valid ports are defined according to existing standards. Any out-of-range or invalid characters render the whole argument invalid.

8. **To change the Modem Init string:**

Applies only for modem connections (via 8600's Serial 1 port)

Command	Response
MO ATF0S0=4↵	MO[ATF0S0=4] entered↵
MO?	Using MO[ATF0S0=4]↵

The 8600 appends CR+LF to the modem init string as transmitted to a modem (physically connected to Serial 1). The 8600 will not perform any case conversion to the argument (i.e., lower case arguments will be transmitted to the modem as lower case).

9. To change the Interface Type:

Command	Response
TY M|D↵	(valid argument) Ty: Modem|Direct entered↵ (invalid argument) ERROR. Using Ty: Modem|Direct↵
TY?↵	Using Ty: Modem|Direct↵

10. To recall a preset:

Command	Response
RP XXXXXXX[PASSCODE]↵	(valid passcode and preset name) ON AIR: XXXXXXX (invalid passcode) [no error message is issued]

In the above table: **XXXXXXX** = the preset name;
PASSCODE = any valid passcode.

- If a non-existent control value and/or an invalid passcode is entered, the 8600 will ignore the command. If a non-existent preset name is entered, your Optimod will return an error message to indicate that the preset you entered does not exist.
- You can apply this command anytime after the 8600 boots up. The 30-minute timeout does not apply.
- This command is useful in interfacing automation systems to the 8600.

11. To turn the analog diversity delay on or off for a given output:

Command	Response
DE XXX[PASSCODE]	(valid passcode and argument) DIVERSITY DELAY: [ON,OFF] (for each output) (invalid passcode) [no error message is issued]

In the above table: **XXX = ANALOG_ON, ANALOG_OFF, DIGITAL1_ON, DIGITAL1_OFF, DIGITAL2_ON, DIGITAL2_OFF, COMPOSITE_ON, COMPOSITE_OFF,** or **?**
? returns the current on/off status for each output that is configured to emit the analog FM processed audio
PASSCODE = ANY VALID PASSCODE.

- If a non-existent control value and/or an invalid passcode is entered, the 8600 will ignore the command.
- You can apply this command anytime after the 8600 boots up. The 30-minute timeout does not apply.

12. To set the analog diversity delay:

Command	Response
TR X.XXXXXXXXX[PASSCODE]	(valid passcode and argument) DELAY: X.XXXXXXXXX (invalid passcode) [no error message is issued]

In the above table: **X.XXXXXXXXX =** the delay time or ?. Acceptable values are 0.000015625 to 8.192 for the 8-second board and 0.000015625 to 13.0000000000 for the 16-second board (10 total digits) in increments of 0.000015625

? returns the current delay time.
PASSCODE = ANY VALID PASSCODE.

- As long as the delay change does not exceed ±1.3 seconds, this command changes the only delay in the HD Radio processing path to prevent glitching in the analog FM path. Larger changes cause the analog path delay to change so that the digital path delay is re-centered and a ±1.3 second range is again available for future adjustments.
- This command is useful when a station has two transmission chains (typically main and backup) that require different diversity delay settings. We recommend first using 8600 PC Remote to set the correct delays for the main and backup chains. When you have found the correct delay for each transmission chain, write down the delay times that PC Remote displays. Then program these into "main" and "standby" batch files that, when executed, send the appropriate TR command to the 8600 when the on-air transmission chain is swapped.
- If a non-existent control value, and/or an invalid passcode is entered, the 8600 will ignore the command.
- You can apply this command anytime after the 8600 boots up. The 30-minute timeout does not apply.

13. To change the polarity of the analog FM processing's output:

Command	Response
FM XXXXXXXX[PASSCODE]	(valid passcode and argument) FM POLARITY is set to XXXXXXXX (invalid passcode) [no error message is issued]

In the above table: **XXXXXXXX = POSITIVE, NEGATIVE, OR ?;**
? returns the current polarity
PASSCODE = ANY VALID PASSCODE.

- For an explanation of why this can be useful, see step 13 on page 2-37).
- If a non-existent control value and/or an invalid passcode is entered, the 8600 will ignore the command.
- You can apply this command anytime after the 8600 boots up. The 30-minute timeout does not apply.
- In HD Radio installations, this command is useful when switching the 8600 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.

14. To select the analog or digital input as the audio source:

Note: When the input is set to DIGITAL and no valid digital signal is present at the input, the setting of the DI ANALOG FALLBACK control will determine whether the Optimod automatically switches to its analog input. See step 9 on page 2-32.

Command	Response
IN X[PASSCODE]	(valid passcode and argument) INPUT: [ANALOG, DIGITAL] (for each output) (invalid passcode) [no error message is issued]

In the above table: **X = A, D**
where A = analog input, D = digital input
PASSCODE = ANY VALID PASSCODE.

- If a non-existent control value and/or an invalid passcode is entered, the 5500 will ignore the command.
- You can apply this command anytime after the 5500 boots up. The 30-minute timeout does not apply.

15. To fetch real-time operational status information from the Optimod:

This provides a real-time status report including the following information:

Command	Response
RT [PASSCODE]↵	5500 Status: digital input 1 lock [active][inactive] digital input 2 lock [active][inactive]

	remote contact closure 1 [active][inactive] remote contact closure 2 [active][inactive] remote contact closure 3 [active][inactive] remote contact closure 4 [active][inactive] remote contact closure 5 [active][inactive] remote contact closure 6 [active][inactive] remote contact closure 7 [active][inactive] remote contact closure 8 [active][inactive] tally out 1 [active][inactive] tally out 2 [active][inactive]

16. To fetch information about the active processing preset:

Command	Response
AP [PASSCODE]?↵	Returns active processing preset name
AP [PASSCODE]??↵	Returns active processing preset control settings
LP [PASSCODE] ↵	Returns a list of all available processing presets

17. To fetch diagnostic information from the 8600:

This provides a status report indicating technical parameters:

Command	Response
ST [PASSCODE]↵	8600 Status: Ver. 1.0.x.x Bootrom: 4 mmm. dd, yyyy HH:MM:SS On Air: LOUD-HOT+BASS User Presets: 0 Memory Total: 16318464 Init Available: 13606912 Mem Available: 10022480 Mem Contiguous: 9420800 Heap Available: 15952 Heap Contiguous: 2048 Disk Size: 23998 Free Disk Space: 18978

18. To set the security timeout of the terminal connection:

Sets the delay (minutes) between when the last terminal command is received and when the Optimod disconnects automatically. Default is 15 minutes. 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.

Command	Response
TIMEOUT [PASSCODE]↵	Delay (minutes) between when the last command is received and when the

```
Optimod disconnects automatically.
```

Connecting to the 8600's Ethernet Port or Serial Port #1 via a Terminal Program on a PC

You can connect a terminal emulation application to the 8600's Ethernet or Serial 1 ports via TCP/IP, port 23 (which is the standard Telnet port and the 8600 factory default). When connected like this, you can:

- recall presets (step 10 on page 2-53)
- turn the diversity delay on and off for a specified output (step 11 on page 2-53)
- trim the diversity delay (step 12 on page 2-54)
- change the polarity of the analog-processed output (step 13 on page 2-54)
- Select the analog or digital input as the audio source (step 14 on page 2-55)
- Fetch real-time operational status information from the Optimod (step 15 on page 2-55)
- Fetch information about the active processing preset (step 16 on page 2-56)
- Fetch diagnostic information from the 8600 (step 17 on page 2-56)

This interface can be used to allow custom third-party applications (including automation systems) to recall presets, view status and set the controls listed above. However, you cannot restore defaults, set security passcodes and change networking controls via Ethernet or Serial 1. You can set those controls using a terminal program with Serial Port #2 (see *Administrative Operations Available via Serial Port #2* on page 2-50) or using 8600 PC Remote software. (See *Installing 8600 PC Remote Control Software* on page 2-66.)

- Unlike Serial 2, Serial 1 uses the PPP protocol.
- To set a different port number:

  A) From the main menu, navigate to SYSTEM SETUP > NETWORK REMOTE > NETWORK. The current setting of the Terminal Port will appear.

  B) If you wish to change the Terminal Port, *Locate* to TERMINAL PORT. Press the ENTER button to access the SET TERMINAL PORT screen.

  C) *Locate* to CLEAR, then press the ENTER button.

  D) This will allow you to enter the Terminal Port number.

  E) *Locate* to the first number and press the ENTER button; repeat until you have selected all the numbers in the Terminal Port. When the Terminal Port entry is complete, *Locate* to SAVE and press the *Enter* button.

- The IP address for this Ethernet connection is the same as the IP address set in step (1.A) on page 2-63 and is visible in the SYSTEM SETUP > NETWORK REMOTE > NETWORK screen. A serial connection through the Serial 1 uses a fixed IP address: 192.168.168.101.

To control the 8600 directly through its Serial 1 or Ethernet port, you can use the freeware terminal emulation application PuTTY. If you wish to automate control, download Plink. Both of these applications are available for free download. Search "PuTTY" with Google to find a download site.

> Note that Windows 7 does not install a Telnet Client by default, so you must do this manually if you wish to initiate a Telnet connection between a Windows 7 machine and your Optimod. Refer to:
> http://technet.microsoft.com/en-us/library/cc771275(WS.10).aspx

Direct Control Using PuTTY

A) If you are using a serial connection, establish a Direct Serial Connection between your computer and the Optimod. See *Appendix: Setting Up Serial Communications* starting on page 2-75.

> A connection through the Serial 1 port uses a fixed IP address: 192.168.168.101

B) Start PuTTY.

The SESSION window appears.

C) Click the TELNET button, which is hard-wired for Port 23.

D) In the TERMINAL category, check "Implicit CR in every LF."

You should not have to change any other PuTTY Terminal, Window, or Connection defaults

E) Specify the host name or IP address:

- If you are connecting through the 8600's Serial 1 port, type 192.168.168.101 into the "Host Name (or IP address)" field.
- If you are connecting through the 8600's Ethernet interface, type the 8600's IP address into the "Host Name (or IP address)" field.

> The IP address for this connection is the same as the IP address set in step (1.A) on page 2-63 and is visible in the SYSTEM SETUP > NETWORK REMOTE > NETWORK screen.

F) Name and save the Session if you wish.

G) Click OPEN.

H) Activate the CAPS LOCK on your computer to ensure that you type in uppercase.

You can now recall presets. Refer to step 10 on page 2-53.

To automate control of the 8600 externally, establish a Telnet/SSH connection and issue commands and parameters, either by typing them directly into a Telnet/SSH client or by placing them within batch files. Then process them with a scriptable

Telnet/SSH client that supports this operation, such as PuTTY, along with its companion command-line interpreter, Plink. You can also use netcat.exe. Below, we provide instructions for both PuTTY/Plink and Netcat.

Custom third party applications can be developed to use this protocol. Additionally, you can include this protocol in an existing application by using small subsets of the standards-based Telnet/SSH protocols directly, or for simplicity, by using scripting or by calling batch files with a Telnet/SSH client such as PuTTY along with its companion command-line interpreter, Plink.

Automating control changes is possible using the Windows Task Scheduler to launch batch files at the desired time.

> **CAUTION**: Because of the powerful features and potential security risks of Netcat, many virus programs may detect this software as a threat. If Netcat is detected as such, configure virus software to allow, and use the software with the normal security precautions. The outbound configurations shown here do not provide any security risks. Inbound connections, if used for other applications, require careful security configuration.

In the examples below, replace "123.45.67.89" with the IP address of the 8600 you are controlling. Replace "23" with the terminal port you specified using the method described on page 2-57. Port 23 is the factory default.

Automated Control Using PuTTY/Plink

This method is scripted with a .cmd file and calls a .txt file. Via plink.exe, the .cmd file calls putty.exe, which then makes the network connection between the computer executing putty.exe and 8600, and specifies the .txt file to use.

The .txt file contains the 8600 commands.

The following two examples recall the presets GREGG and GREGG OPEN respectively.

The file "8600_P1_Gregg.cmd" contains:

```
plink -raw -P 23 123.45.67.89 < 8600_P1_Gregg.txt
```

The file "8600_P1_Gregg.txt" contains:

```
RP GREGG [PASSWORD]
disconnect
```

The file "8600_P2_GreggOpen.cmd" contains:

```
plink -raw -P 23 123.45.67.89 < 8600_P2_GreggOpen.txt
```

The file "8600_P2_GreggOpen.txt" contains

```
RP GREGG OPEN [PASSWORD]
disconnect
```

Automated Control Using Netcat

Only one utility is required to use this method: netcat.exe. This is available as a free download. Google "netcat."

This method is scripted with a .cmd file and calls a .txt file.

- The .cmd file calls netcat.exe to make the network connection between the computer executing netcat.exe and 8600, and specifies the .txt file to use.
- The .txt file contains the 8600 commands.

The following two examples recall the presets GREGG and GREGG OPEN respectively.

The file "8600_P1_Gregg.cmd" contains:

```
nc.exe -v -n -w 1 123.45.67.89 23 < 8600_P1_Gregg.txt
```

The file "8600_P1_Gregg.txt" contains:

```
RP GREGG [PASSWORD]
disconnect
```

The file "8600_P2_GreggOpen.cmd" contains:

```
nc.exe -v -n -w 1 123.45.67.89 23 < 8600_P2_GreggOpen.txt
```

The file "8600_P2_GreggOpen.txt" contains:

```
RP GREGG OPEN [PASSWORD]
disconnect
```

Remote Control Interface Programming

[Skip this section if you do not wish to program the opto-isolated GPI remote control interface at this time.]

1. **Navigate to the System Setup > Network Remote screen.**
2. **Program one or more remote control interfaces.**

   To program a given remote input, use LOCATE to highlight the input. As you turn the control knob, the functions listed below will appear in the highlighted field. A momentary pulse of voltage will switch most functions, except as noted.

   - **Preset Name**: switches that preset on the air. Any factory or user preset may be recalled via the control interface.
   - **Input**: Analog: selects the analog inputs.

- **Input:** Digital**:** selects the digital input. No J.17 de-emphasis is applied to the digital input.
- **Input**: Digital+J.17: selects the digital input and applies J.17 de-emphasis to the digital input.
- **Bypass**: switches the Bypass Test Mode on the air.
- **Tone**: switches a Tone Test on the air.
- **Exit Test**: If a test mode (Tone or Bypass) is switched on the air, EXIT TEST reverts to the normal operating mode using the previous processing preset.
- **Stereo**: switches the 8600's stereo encoder on using normal double sideband modulation of the stereo subchannel.
- **SSB:**. switches the 8600's stereo encoder on using compatible single sideband/vestigial sideband modulation of the stereo subchannel. See *SSB Stereo Encoder Operation* on page 3-11.
- **Mono From Left, Mono From Right, or Mono From Sum**: switches the 8600's stereo encoder off, using the Left, Right, or Sum (L+R) respectively, as the program source.
- **MOD Reduction 1,** or **MOD Reduction 2**: reduces the program modulation by the percentage programmed in the Input/output screen. When voltage is removed, these functions are deactivated.
- **Reset Clock To Hour**
- **Reset to Midnight**
- **Monitor Mute** is a non-latching function that mutes the analog output if it has been programmed to emit the monitor signal (in the INPUT/OUTPUT: OUTPUT1 screen).
- **Analog Out Delay In**: Activates the diversity delay for the analog output if it is configured to emit the output of the analog FM processing chain (see page 3-75).
- **Digital Out 1 Delay In**: Activates the diversity delay for Digital Output #1 (Analog, Digital #1, Digital #2, Composite) if it is configured to emit the output of the analog FM processing chain (see page 3-75). There is also a DIGITAL OUT 2 DELAY IN choice available.
- **Analog Out Delay Out:** Bypasses the diversity delay for a specified output (Analog, Digital #1, Digital #2, Composite) emitting the output of the analog FM processing chain (see page 3-75). DIGITAL OUT 1 DELAY OUT and DIGITAL OUT 2 DELAY OUT are also available.
- **FM Polarity Positive**: Makes positive the output polarity of the 8600's FM analog channel processing. In HD Radio installations, this command is useful

when switching the 8600 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 Polarity Negative**: (see above)
- **No Function**: remote input is disabled.

3. **End remote control interface programming.**

   When you are finished programming the remote control interface, press the *Escape* button once to return to the System Setup screen.

Tally Output Programming

[Skip this section if you do not wish to use the tally outputs.]

See step 6 on page 2-4 for wiring instructions.

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

1. **Navigate to the SYSTEM SETUP > NETWORK REMOTE > TALLY OUTPUT screen.**
2. **Program one or both of the tally outputs.**

   To program a given tally output, use LOCATE to highlight the output. As you turn the control knob, the functions listed below will appear in the highlighted field.

   - **Input: Analog**: Indicates that the 8600 is processing audio from its analog input.
   - **Input: Digital**: Indicates that the 8600 is processing audio from its AES3 digital input.
   - **Analog Input Silent**: Indicates that the level at either or both analog input channels is below the threshold set in step 17 on page 2-39.
   - **AES Input Silent**: Indicates that the level at either or both digital input channels is below the threshold set in step 17 on page 2-39.
   - **AES Input Error**: Indicates that the 8600's AES input receiver chip has detected an error in the received data that makes it unusable. When the chip detects such an error, it automatically switches the 8600's input to ANALOG.
   - **No Function**: Tally output is disabled.

Networking and Remote Control

[Skip this section if you do not wish to connect to your 8600 remotely, either for downloading software upgrades or for PC Remote Control. For terminal and automated control using simple ASCII commands, see Networking and Remote Control *starting on page 2-63.]*

The 8600 has a built-in 100 Mbps Ethernet port that can be used with 10 Mbps, 100 Mbps, and 1 Gbps networks using the TCP/IP protocol. You can also connect a PC to the 8600 through the 8600's RS-232 serial port, either by modem or directly through a null modem cable.

1. **Prepare the 8600 for a network connection.**

   [Skip this step if you will not be using an Ethernet connection.]

   A) Set the IP address:

      a) From the main menu, navigate to SYSTEM SETUP > NETWORK REMOTE > NETWORK.

      b) If you wish to change the IP Address, Locate to SET ADDRESS. Press the ENTER button to access the SET IP ADDRESS screen.

         192.168.254.254 IS THE 8600 DEFAULT SETTING.

      c) *Locate* to CLEAR, then press the ENTER button.

         This will allow you to enter your IP address.

      d) *Locate* to the first number and press the ENTER button; repeat until you have selected all the numbers in the IP address assigned by your network administrator. When the IP address entry is complete, *Locate* to SAVE and press the *Enter* button.

   B) If necessary, set the Subnet Mask assigned by your network administrator:

      a) *Locate* to SET SUBNET.

         Unless previously set away from the factory default, the Subnet Mask should indicate 255.255.255.0.

      b) Press the ENTER button to access the SET SUBNET MASK screen.

      c) *Locate* to the first number and press the ENTER button; repeat until you have selected all the numbers in the Subnet Mask. When the Subnet Mask entry is complete, toggle to SAVE and press the *Enter* button.

         The 8600 will not accept an invalid Subnet Mask (like 255.255.255.300).

   C) If necessary, set the Gateway assigned by your network administrator.

      To cross subnets, you must specify a gateway. If the PC and 8600 are on the same subnet, then it is unnecessary to specify a gateway.

      If the gateway, port, and firewall (if used) are configured correctly, it is possible to connect 8600 PC Remote to an 8600 via a VPN.

a) *Locate* to SET GATEWAY.

Unless previously set away from the factory default, the Gateway should indicate 0.0.0.0.

b) Press the ENTER button to access the Set Gateway Address screen.

c) *Locate* to the first number and press the ENTER button; repeat until you have selected all the numbers in the Gateway. When the Gateway entry is complete, toggle to SAVE and press the *Enter* button.

D) If necessary, set the Port assigned by your network administrator.

If you are behind a firewall, this port needs to be opened in order to communicate with the 8600 PC Remote application.

a) *Locate* to SET PORT.

The default port is 6201.

b) Press the ENTER button to access the SET PORT ADDRESS screen.

c) *Locate* to the first number and press the ENTER button; repeat until you have selected all the numbers in the Port. When the Port entry is complete, toggle to SAVE and press the *Enter* button.

E) Connect your network's Ethernet cable to the card.

This completes setup of network parameters.

2. **Prepare the 8600 for direct serial connection through serial port 1:**

[Skip this step if you will not be using a direct serial connection.]

A) Configuring the 8600 network screen settings:

a) From the main menu, navigate to SYSTEM SETUP > NETWORK REMOTE > NETWORK.

b) *Locate* to INTERFACE TYPE.

c) Turn the blue knob until DIRECT appears in the INTERFACE TYPE field.

B) Set 8600 passcodes as desired. See *Security and Passcode Programming* on page 2-43.

C) Connect the cable:

a) Connect one end of the null modem cable that we supplied with your 8600 to the SERIAL 1 CONNECTOR on the 8600's rear panel.

b) Connect the other end to your computer's COM port.

You are now ready to connect your computer to your 8600 through a null modem cable connected to your computer's serial port. Refer to *Installing 8600 PC Remote Control Software* (page 2-66).

3. **Prepare the 8600 for modem connection through serial port 1:**

[Skip this step if you will not be using a modem connection.]

A) Configure the network screen settings on your 8600:

a) From the main menu, navigate to SYSTEM SETUP > NETWORK REMOTE > NETWORK.

b) *Locate* to INTERFACE TYPE.

c) Turn the blue knob until MODEM appears in the INTERFACE TYPE field.

B) Set the modem initialization string:

a) LOCATE to INIT STRING and the SET STRING button.

b) If the INIT STRING is S0=4, this is correct. Skip to step (C) below.

> S0=4 is the 8600 default setting. This activates auto-answer functionality in the modem.

c) If the INIT STRING reads UNDEFINED, press the ENTER button to access the MODEM INIT STRING screen.

d) Using the LOCATE button, toggle to CLEAR. Then press the ENTER button.

> This will clear the INIT STRING field of UNDEFINED so that you can enter your Init String.

e) Set the INIT STRING to S0=4.

> LOCATE to the first number and press the ENTER button; repeat until you have selected all the numbers in the Init String.

C) Set 8600 passcodes as desired. See *Security and Passcode Programming* on page 2-43.

D) Modem setup:

You will need two modems and two available phone lines, one of each for your PC and your 8600.

> **Reminder**: Orban Customer Service supports only the 3Com / U.S. Robotics® 56kbps fax modem EXT on the 8600 side (although other 56kbps modems will often work OK).

a) Connect the modem to the 8600's SERIAL 1 port with a standard (not null) modem cable. The cable provided with your 8600 is a null modem cable and will not work.

> You can use either an internal or an external modem with your PC.

b) Connect the telephone line from the wall phone jack to the wall connection icon on the back of the modem (modem in).

c) Connect the modem cable from the modem to the SERIAL 1 port of the 8600.

d) Set the modem to AUTO ANSWER and turn it on.

> For 3Com / U.S. Robotics® 56kbps fax modem EXT, set dip switches 3, 5, and 8 in the down position to activate the AUTO ANSWER setting. All other dip switches should be set to the up position.

Installing 8600 PC Remote Control Software

8600 and 8600FM do not use the same PC Remote software. Be sure to use 8600 PC Remote with an 8600 and 8600FM PC Remote with an 8600FM.

This section briefly summarizes the procedure for installing 8600 PC Remote software on existing 8600s. If required, you will find more detailed instructions in the .pdf file automatically installed on your computer by Orban's installer program, `Setup8600_x.x.x.x.exe`, where "x.x.x.x" represents the software version you are installing. (For example, for version 1.0 software, this would be 1.0.0.0.)

The PC Remote software is supplied on a CD shipped with your 8600. You can also download it from `ftp.orban.com/8600`.

Instructions for using the PC Remote software start on page 3-88.

Installing the Necessary Windows Services

The 8600 PC Remote application uses Windows' built-in communications and networking services to deal with the low-level details necessary to communicate with the 8600's serial port. (These services are also used to upgrade your 8600's firmware when updates are available from Orban.) The exact process will vary, depending on how you wish to set up the communications. That is:

If you want to communicate through a local PC, you will need to establish a connection between a serial (COM) port of the PC and the COM port of your 8600 through a null modem cable (supplied with your 8600). You will then use Windows Direct Serial Connect to make the basic connection. Alternatively, you can use a crossover Ethernet cable to communicate to your PC through its Ethernet port. Only the latter is supported for Windows Vista and 7.

If you want to communicate through a pair of modems, you will use the Windows Dial-Up networking service to make the connection.

You must install the appropriate communications services in Windows (if they are not already installed) before you can run 8600 Remote software. *You may therefore need to have access to the Windows install disk(s)—or have their image copied onto your computer's hard drive—before you attempt to use the 8600 update or remote applications.*

> In all cases, regardless of whether your PC communicates to the 8600 through its serial port or Ethernet connector, it uses the ppp and the TCP/IP protocols to communicate with the 8600.

Check Hardware Requirements

To connect your PC to your 8600, regardless of the method you choose, you will need the following:

- Orban 8600 OPTIMOD-FM.

- *If connecting by serial cable:* a null modem cable (also called a "reverse" cable), supplied by Orban with your 8600 when it was shipped. This cable has DB9 female connectors at both ends for connecting the 8600 to the serial port on your computer. If your computer has a DB25 connector, you will need to obtain an adapter.
- *If connecting by modem:* a 3Com / U.S. Robotics® 56kbps fax modem EXT and normal (not null) modem cable for the 8600 side of the connection. Note that Orban Customer Service does not support any other type of modem for connecting to the 8600 although many modems will work.
- *If connecting by network:* a standard Ethernet cable (with RJ45 connectors) to connect to a network hub or router, or a crossover Ethernet cable to connect directly to your PC's Ethernet jack.
- PC running Windows 2000 (SP3 or higher)/XP/Vista/7.

  8600 PC Remote and 8600 Updater will not run on older Windows versions.

Recommended Components

Computer	Pentium II or higher
Available Disk Space	25MB
RAM	256MB
Display	SVGA or higher
Microsoft Windows	XP/Vista/7/8/10
COM Port	16550 (or compatible) UART

WARNING!

When connecting your 8600, use shielded cable to protect the pins in the RS-232 connector from electrostatic discharge.

Running the Orban Installer Program

Insert the installer CD into your computer's CD drive.

The installer should start up and ask you if you wish to install the PC Remote application on your computer. If it fails to do so, navigate to Start \ Run on your computer and type `X:setup` (where "X" is the drive letter of your CD drive).

Follow the prompts on your screen to install the PC Remote software automatically on your computer.

- You might have obtained the automatic installer application from some other source than Orban's CD, like Orban's ftp site or another computer on your network. If so, run the application and follow the on-screen instructions.
- This program installs the necessary files and adds an Orban / Optimod 8600 folder to your computer's Start Menu. This folder contains shortcuts to the PC Re-

mote application and to the documentation. If you accepted the option during installation, there is also a shortcut to the PC Remote application on your desktop.

You have now installed all files necessary to use the PC Remote software. If you are using a direct serial or a modem connection, the next step is to install and configure the Windows communications services that allow your computer to communicate with your 8600. *Appendix: Setting Up Serial Communications* (starting on page 2-75) provides details.

Setting Up Ethernet, LAN, and VPN Connections

If you are using an Ethernet connection and your computer can successfully connect to the Internet through its Ethernet port, it already has the correct (TCP/IP) networking set up to communicate with the 8600. In most cases, all you need is your 8600's IP address, Port, and Gateway number, as set in step 1 on page 2-63. You will enter these when you create a "connection" to your 8600 from the 8600 PC Remote application—see *To set up a new connection* on page 3-89. If your computer does not have an Ethernet port, you will need to add one (usually by adding a PCI Ethernet card) and then following the instructions provided by Microsoft to set it up to enable TCP/IP networking.

If you are using a crossover Ethernet cable to connect your Optimod directly to your computer, you must set your Windows networking to provide a static IP address for your computer because your Optimod does not contain a DHCP server.

If you wish to connect to your 8600 through your LAN or VPN (through a WAN or the Internet), consult your network administrator. Note that to cross subnets, you must specify a gateway. If the PC and 8600 are on the same subnet, then it is unnecessary to specify a gateway.

If you are behind a firewall, you must open the port you specified in step (1.D) on page 2-64. If the gateway and firewall (if used) are configured correctly, it is possible to connect 8600 PC Remote to an 8600 via a VPN.

Conclusion

By carefully following the instructions in this Section (including the Appendix, if you are connecting via direct serial cable or modem), you should have successfully installed the necessary Windows services and connected to your 8600. (Note that Ethernet connections are preferred because they are easier to configure than serial connections and are faster.) However, if you experience any problems with this process, or have any other 8600 questions, please contact Orban Customer Service:

phone: +1 856.719.9900

email: custserv@orban.com

For details on your new 8600 software, from new features to operational suggestions, refer to our FTP site (ftp.orban.com/8600).

Synchronizing Optimod to a Network Time Server

[Skip this section if you do not wish to automatically synchronize your Optimod's internal clock to a network timeserver, which may be part of your local network or located on the Internet.]

1. **From the main menu, Locate to System Setup > Place/Date/Time > Time Sync.**

   A) Use the SYNC PROTOCOL control to choose either TIME PROTOCOL or SNTP.

   - Select TIME PROTOCOL if the Optimod is behind a firewall that does not pass UDP packets. TIME PROTOCOL selects the Time Protocol as described in the standard RFC868. This method uses TCP on port 37.
   - Select SNTP if your network timeserver supports the Simple Network Time Protocol as described in standard RFC1769. This method uses UDP on port 123.

   Ask your network administrator which protocols are available. SNTP is slightly more accurate.

   B) Locate to SYNC PERIOD. Using the knob, choose how often your Optimod will automatically update its internal clock to the timeserver you selected.

   The choices are OFF, 8 HOURS, and 24 HOURS.

If the connection to the timeserver fails (due to network overload or other problems), your Optimod will try once per hour to synchronize until it is successful.

   C) Locate to OFFSET. Using the knob, set it to the difference (in hours) between your time zone and Universal Time (UTC).

   UTC is also known as GMT, or Greenwich Mean Time.

   - The value can range between –12 and +12 hours. If this value is set to 0, your Optimod's time will be the same as UTC.
   - You can empirically adjust this value until the correct time for your location is displayed after you synchronize your Optimod to a timeserver.

2. **Choose a timeserver.**

   As of August 2011, http://tf.nist.gov/tf-cgi/servers.cgi# provides a current list of timeservers available on the Internet. You network may also have a local timeserver; ask your network administrator.

3. **Set up timeserver parameters.**

   You can specify the timeserver either from your Optimod's front panel or from its PC Remote software. From the front panel, you can only enter the timeserver's IP address (for example, 192.43.244.18). If you specify the timeserver from PC

Remote, you can specify either its named address (for example, time.nist.gov) or its IP address.

4. **Specify the time sync parameters from your Optimod's front panel:**

   [Skip this step if you wish to specify the timeserver and time sync parameters from your Windows computer.]

   A) Locate to the SET SERVER button and press *Enter*.

      a) Using the *Locate* button, locate to CLEAR. Then press the ENTER button.

         This will allow you to enter the IP address of the desired timeserver.

      b) Use the *Locate* button, toggle to the first number and then press the ENTER button; repeat until you have selected all the numbers in the IP address. When the IP address entry is complete, toggle to SAVE and press the *Enter* button.

   B) Locate to the SYNC NOW field and press *Enter* to test your settings. Your Optimod's display should indicate that it is connecting to the IP address that you specified. When the connection is successful, the Optimod's clock will automatically synchronize to the timeserver.

   - If the connection is not successful within five seconds, the display will indicate that the connection failed. This means either that the timeserver is too busy or that your setup cannot connect to the timeserver. Double-check the IP address. If you are behind a firewall, make sure that port 123 is open.

   - If your connection failed, the gateway address might not be set correctly on your Optimod. The gateway address for the timeserver connection is the same gateway address that you set in step (1.C) on page 2-63. If you do not know the correct gateway address, you can often discover it by connecting a Windows computer to the same Ethernet cable that is ordinarily plugged into your Optimod. Ascertain that the computer can connect to the Internet. At the command prompt, type `ipconfig`. The computer will return the "Default Gateway."

5. **Specify the time sync from the Optimod PC Remote software:**

   [Skip this step if you wish to specify the timeserver and time sync parameters from your Optimod's front panel.]

   Optimod PC Remote software can automatically set your Optimod's local time, OFFSET, and TIME SERVER to reflect the Windows settings in the machine running PC Remote software.

   See *Installing 8600 PC Remote Control Software* starting on page 2-66 and *Using the 8600 PC Remote Control Software* starting on page 3-88.

   If you are running Windows 2000, you cannot specify the timeserver from your computer. However, you can still set your Optimod's clock and offset.

   A) In Windows, navigate to the CONTROL PANEL / DATE AND TIME / TIME ZONE tab.

B) Set time zone to correspond to your local time zone.

C) In Windows, navigate to the CONTROL PANEL / DATE AND TIME / INTERNET TIME tab.

D) If you are running Windows XP or higher:

a) Check "Automatically synchronize with an Internet time server" to set your Optimod's SYNC PERIOD to "24."

b) Set "Server" to the desired timeserver.

c) Click the "Update Now" button to synchronize your computer's clock to the selected timeserver. If this is successful, this means that you can connect to the selected timeserver over your network.

- The INTERNET TIME tab is not available in Windows 2000. If you are running Optimod PC Remote on Windows 2000, you must enter the timeserver from your Optimod's front panel as an IP address (step 4 on page 2-70).
- If the timeserver you selected in Windows is a named address (not an IP address), the 8600 will resolve it correctly but the IP address that appears in your Optimod's display will be 0.0.0.0.
- To use PC Remote to turn off your Optimod's automatic synchronization, uncheck "Automatically synchronize with an Internet time server" on your PC. Then click the "Update Now" button on PC Remote.

E) Navigate to Optimod PC Remote's SETUP/ UTILITY tab and click the SET 8600 CLOCK button.

- If you are running Windows XP or higher, PC Remote will download your computer's currently specified timeserver into your Optimod.
- PC Remote will adjust your Optimod's OFFSET setting to correspond to your computer's time zone setting.
- PC Remote will synchronize your Optimod's clock with your computer's clock.

F) It is wise to disconnect from PC Remote and then to *Locate* to the SYNC NOW field on your Optimod and press *Enter* [step (4.B) on page 2-70]. This is to test your Optimod's ability to synchronize to the selected timeserver and to ensure that your Optimod's clock is set accurately. If the test fails, see the comments under step (4.B) on page 2-70, particularly those regarding setting the gateway.

NOTE: Manually setting your Optimod's clock via Set Time, Set Date, Daylight Time, and the remote contact closure Reset to Hour and Reset to Midnight will not work when the automatic synchronization function is active. To inactivate this function (thereby permitting manual setting to work), set the SYNC PERIOD *to* OFF.

Updating your 8600's Software

8600 and 8600FM have different internal software. When upgrading your 8600 or 8600FM, be sure to use 8600 PC Remote with an 8600 and 8600FM PC Remote with an 8600FM. If you attempt to upgrade an 8600FM using 8600 PC Remote software, the software will refuse to connect.

The software version number of PC Remote must be the same as the version number of the software running within your 8600. If the software version of PC Remote is higher than the version running in your 8600, PC Remote will automatically detect this and will offer to update your 8600's software automatically.

1. **If you have not already done so, prepare your computer and the 8600 for a direct serial, modem, or Ethernet connection.**

   See *Networking and Remote Control* starting on page 2-63.

2. **Install the latest version of 8600 PC Remote software on your computer.**

   This is available from

   ftp://orban.com/8600

   See *Installing 8600 PC Remote Control Software* on page 2-66.

   See the readme8600_x.x.x.x.htm file (where x.x.x.x is the version number) for details about the upgrade not given in this manual. The PC Remote installer will install this file on your computer's hard drive.

3. **If you have not previously done so, start 8600 PC Remote and set up a "connection" to the 8600 you will be updating.**

   See *To set up a new connection* on page 3-89.

4. **Update your 8600.**

   A) Attempt to initiate communication to your 8600 via your connection.

   See *To initiate communication* on page 3-90.

   8600 PC Remote will automatically detect that the 8600 software version on your 8600 is not the same as the version of 8600 PC Remote. PC Remote will then offer to update your 8600 automatically.

   This procedure will only work for a connection using an "all-screens" (administrator) passcode.

   B) Choose YES and wait for the update to complete. *Note that this will cause an interruption in the audio of approximately 3 seconds when your 8600 automatically reboots after the update is complete.* If you cannot tolerate such an interruption, choose NO or CANCEL to abort the update.

   Please be patient; this will take several minutes. (The exact time will depend on whether the 8600 has to do any "housekeeping" to its flash memory as part of the update.)

Completion will be indicated by the updater's command-line window's closing automatically and your 8600's rebooting.

Your 8600 will continue to pass audio normally while the update is occurring. *However, the audio will be interrupted for approximately 3 seconds when your 8600 reboots.*

Do not interrupt power to your 8600 or your computer, close PC Remote or the update application's command-line window, or reboot your computer during this time. While doing any of these things is unlikely to damage your 8600 (because of extensive backup and error-checking provisions in your 8600), they will certainly cause the update to fail. It is wise to power your 8600 via a UPS.

C) When the 8600 screen display returns after its automatic reboot, the 8600 will be running with the updated software.

If the update fails for some reason, try repeating the procedure in steps (A) through (C) again.

D) If the 8600 screen remains blank for more than one minute after the update has completed, manually reboot the 8600 by removing AC power from the 8600 for at least ten seconds and then powering the 8600 back up.

E) The 8600 software update is now complete. You should now be able to connect to your 8600 via PC Remote.

NOTE: If you cannot make a connection after a software upgrade, manually reboot the 8600 with a normal "power-off/power-on" sequence.

Appendix: Setting Up Serial Communications

This appendix provides instructions for setting up both direct serial and modem connections from your 8600 to your PC. You must do this when you define a new connection from the 8600 PC Remote application. The appendix provides Direct Serial Connect and Modem Connect procedures for the Windows 2000 and Windows XP operating systems and the Direct Serial Connect procedure for Windows 7. To set up modem communications with Windows 7, use the instructions that Microsoft provides for this operating system.

Please note that the accompanying screen shots were originally prepared for Orban's OPTIMOD-FM 8300 and refer to that product. However, they are fully applicable to the 8600.

Preparing for Communication through a Null Modem Cable

1. **Configure your 8600.**

   A) On your 8600's front panel, navigate to SETUP / NETWORK & REMOTE.

   B) Hold down the PC CONNECT soft button and turn the knob until you see DIRECT on the display.

2. **Connect the cable.**

   A) Connect one end of the null modem cable that we supplied with your 8600 to the DB9 serial connector on the 8600's rear panel.

      Be sure to use a null modem cable. A normal serial cable will not work.

   B) Connect the other end of the cable to your computer's COM port.

Connecting Using Windows 2000 Direct Serial Connection:

Ordinarily, a direct serial connection through a null modem cable is used only when you are controlling one 8600 per available COM port on your computer. If you wish to control multiple local 8600s, it is better to use an Ethernet network connection. However, in principle you could control multiple 8600s serially from one COM port, using a hardware serial switch to select the 8600 you wish to control. In this case, you should set up a separate 8600 "connection" for each 8600 to be controlled, following the instructions below. All connections should reference the same COM port.

This connection is used both for upgrading your 8600 and for connecting the 8600 PC Remote application to your 8600.

Important: The Direct Serial Connection must have exclusive access to the PC COM port that connects to your 8600. Make sure than any software that monitors this COM port (such as HotSync manager, etc) is disabled before running Direct Serial Connection.

If you have already configured your direct serial cable connection, skip to step 2 on page 2-80.

To enforce security, Windows 2000 does not permit a Direct Serial Connection and a LAN connection to coexist. If you wish to have access to your LAN connection at the same time as you are connected to your 8600 by a direct serial cable, you must connect to your 8600 via the RAS Serial Cable modem, not Windows Direct Connect. After you have installed the RAS Serial Cable modem, you can use the instructions below for adding a "Modem Connection."

The RAS Serial Cable modem appears in the list of modems available to Windows in the "Null modem types" category. (It is a piece of software included with Windows, not a piece of hardware.) While we do not support or provide detailed installation instructions for the RAS null modem supplied with Windows 2000 and XP, it can be Added just like any other modem in these operating systems (see your Windows documentation). However, for security reasons, we do not recommend doing this.

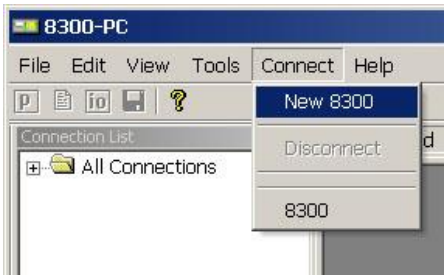
Note that when the RAS modem is installed, it *may cause Windows Direct Connect to stop working*. You may have to completely disable the RAS modem (by uninstalling it) and then reboot to recover Direct Connect functionality.

1. Add and configure a Direct Connection for Windows 2000:

A) Create a New Windows 2000 Direct Connection:

a) Launch 8600 PC Remote.

b) Choose "Connect / New 8600"

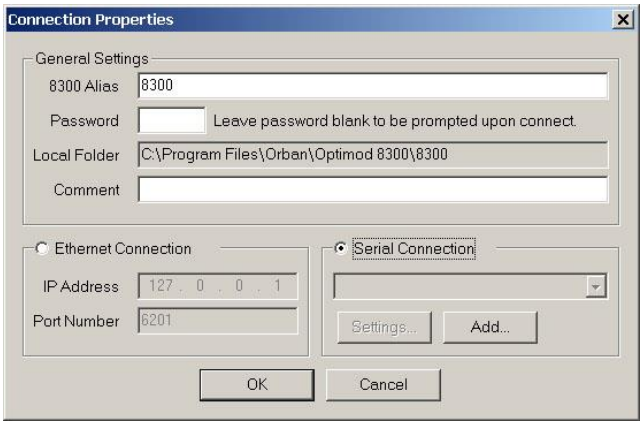


c) Give your 8600 a name (e.g., "KABC") by entering this name in the "8600 Alias" field.

d) If you wish to have 8600 PC Remote remember the password for this Optimod, enter the password in the "Password" field.

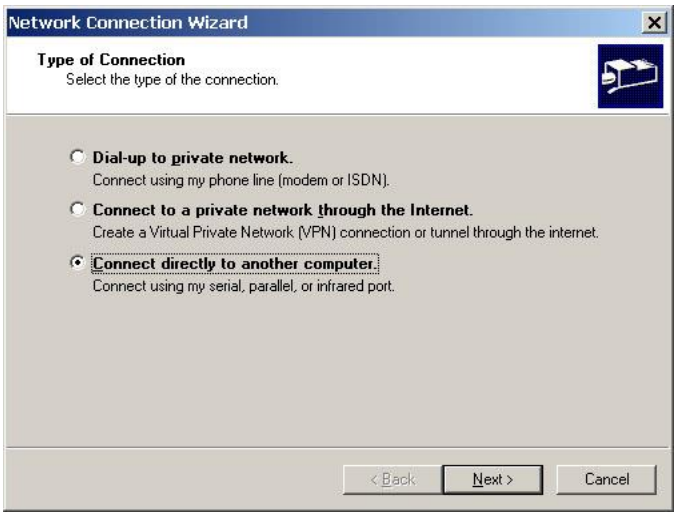
e) Select "Serial Connection."

f) Click "Add."



g) Select "Connect Directly to another computer."

h) Click "Next."



i) In the drop-down box, select the serial port you will be using to make the connection.

j) Click "Next."

k) Select either "For all users" or "Only for myself."

The correct setting depends on how your network and security are configured.

Your wizard may not display this field if your computer is set up for a single user only.

l) Click "Next."

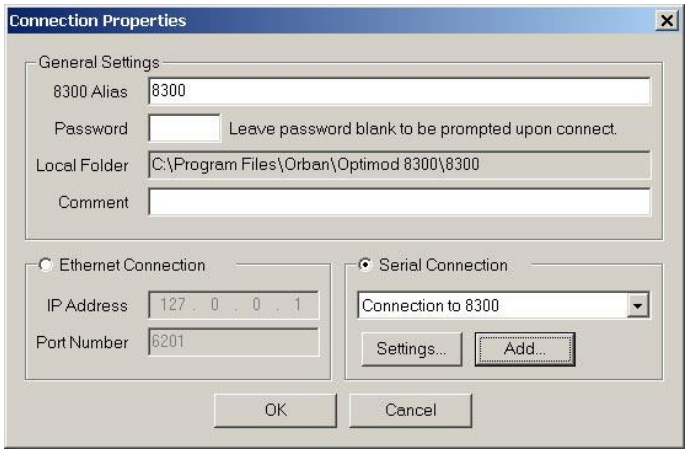
m)Enter a name for your Connection such as: "Connection to 8600."

n) Click "Finish."

o) Click "Yes."

B) Edit your new Direct Connection properties:

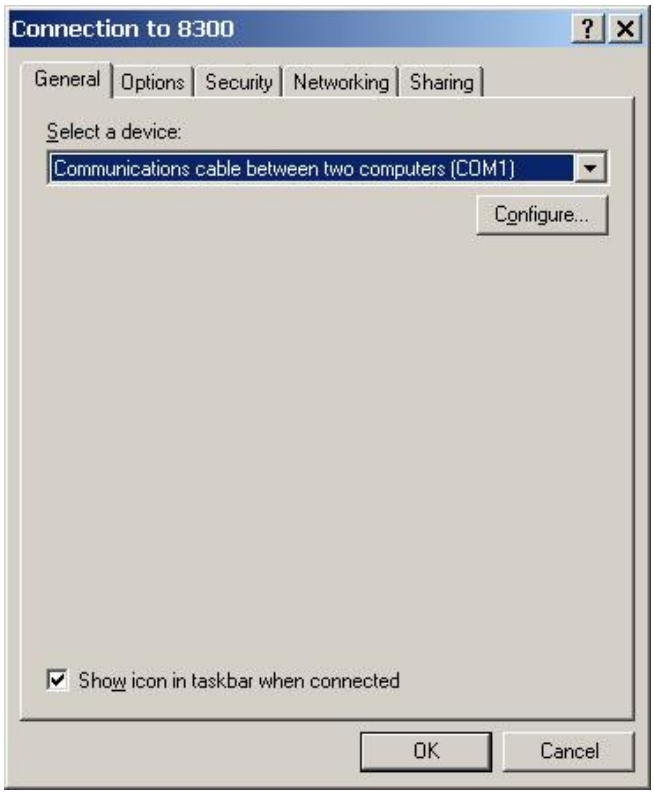
a) Click "Settings."



b) Click the "General" tab.

c) Select the device you set up in step (i) on page 2-77. This will usually be "Communications cable between two computers (COM1)."

d) Click "Configure."

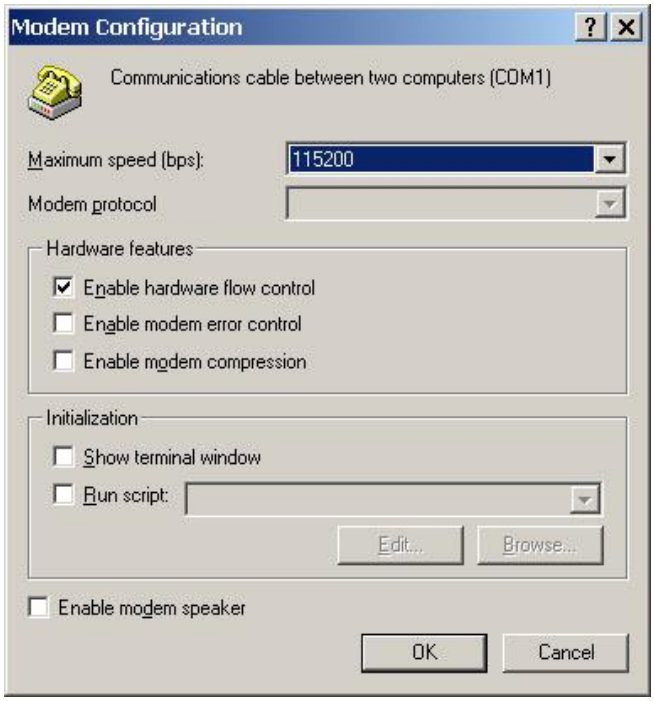


e) Set "Maximum speed (bps)" to "115200."

f) Check "Enable hardware flow control."

g) Make sure that all other boxes are not checked.

h) Click "OK."



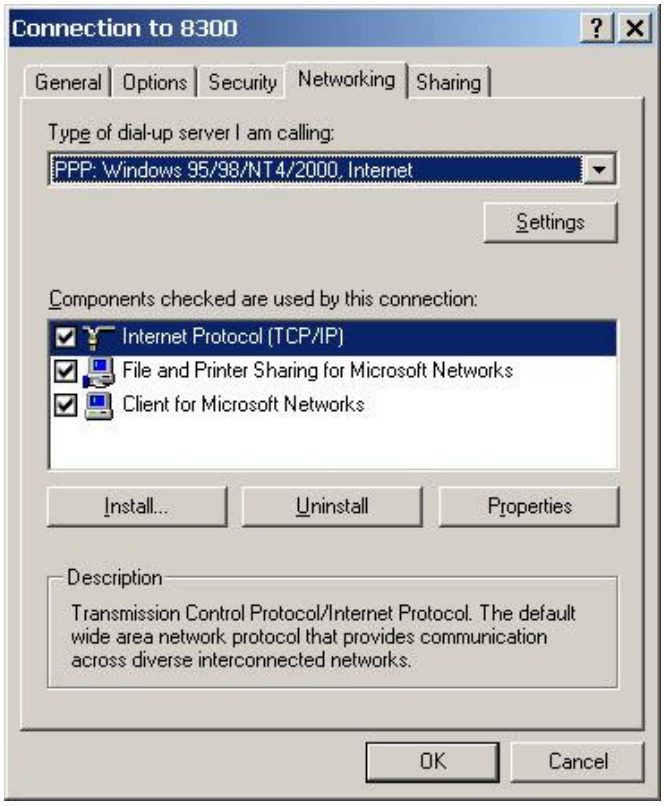
i) Select the Networking tab.

j) Make sure that "PPP: Windows 95 / 98 / NT 4 / 2000, Internet" appears in the "Type of dial-up server I am calling" field.

k) Make sure that "Internet Protocol (TCP/IP) is checked.

You may leave "File and Printer Sharing for Microsoft Networks" and "Client for Microsoft Net-works" checked if you like.

l) Click "OK."



When the "Connection properties" window appears, click "OK."

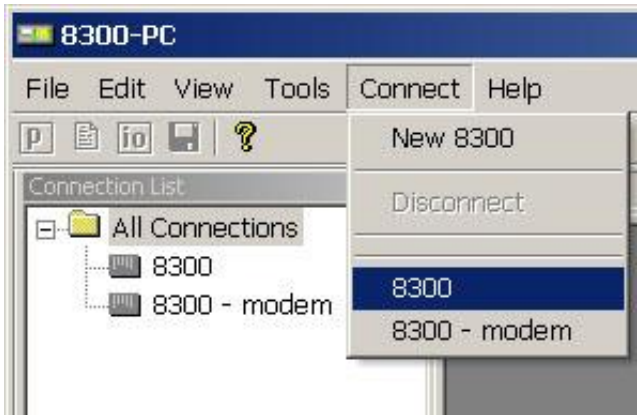
2. Launch an existing Windows 2000 Direct connection.

Once you have set up a "connection" specifying Direct Connect in the 8600 PC Remote application (see *To set up a new connection* on page 3-89), choosing this connection from 8600 PC Remote automatically opens a Windows Direct Connection to your 8600.

You can connect by selecting the desired connection from the drop-down list in the CONNECT menu.

You can also connect by double-clicking the connection in the "Connection List" window.

A dialog bubble will appear on the bottom right hand corner of the screen verifying your connection if the connection is successful.



If you have trouble making a connection, refer to *OS Specific Troubleshooting Advice: Troubleshooting Windows 2000 Direct Connect* on page 5-10. If you have

trouble the first time after creating a connection according to the instructions above, try restarting your computer to clear its serial port.

3. **To change the properties of an existing connection:**

   Right-click the connection in the "connection List" window and choose "Properties." The "Connection properties" window opens (see page 2-76).

Connecting Using Windows XP Direct Serial Connection

If you have already configured your direct serial cable connection, skip to step 2 on page 2-84.

> To enforce security, Windows XP does not permit a Direct Serial Connection and a LAN connection to coexist. If you wish to have access to your LAN connection at the same time as you are connected to your 8600 by a direct serial cable, you must connect to your 8600 via the RAS null modem, not Windows Direct Connect. We do not support or provide installation instructions for the RAS null modem with Windows 2000 and XP, but it can be installed just like any other modem in these operating systems. However, for security reasons, we do not recommend doing this (see page 2- 76).

1. **Add and configure a Direct Connection for Windows XP:**

   A) Create a New Windows XP Direct Connection:

   a) Launch 8600 PC Remote.

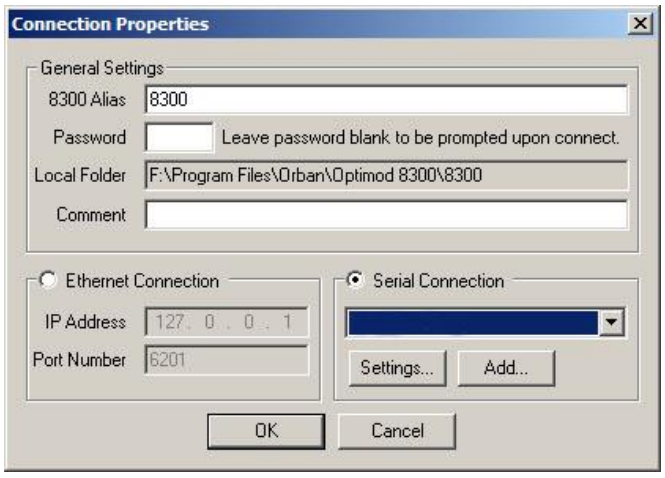
   b) Choose "Connect / New 8600"

   c) Give your 8600 a name (e.g., "KABC") by entering this name in the "8600 Alias" field.

   d) If you wish to have 8600 PC Remote remember the password for this Optimod, enter the password in the "Password" field.

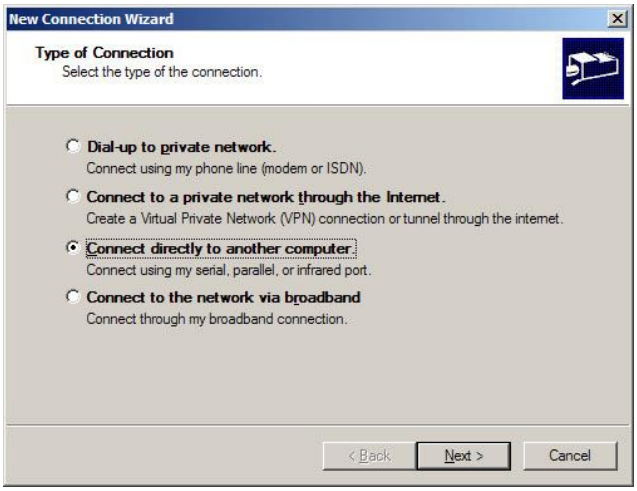
   e) Select "Serial Connection."

   f) Click the "Add" button.



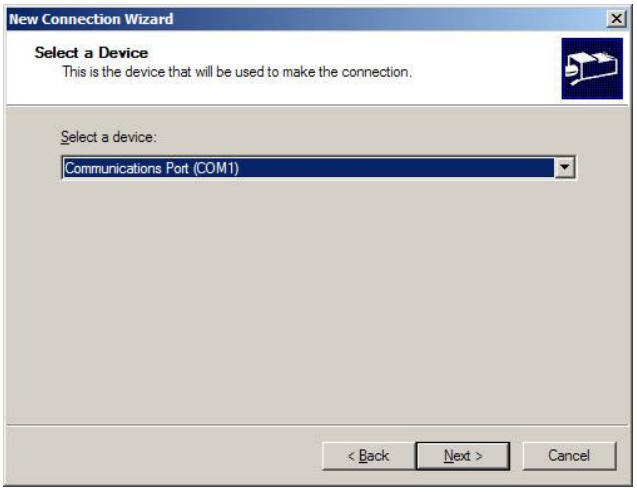
g) Choose "Connect directly to another computer."

h) Click "Next."



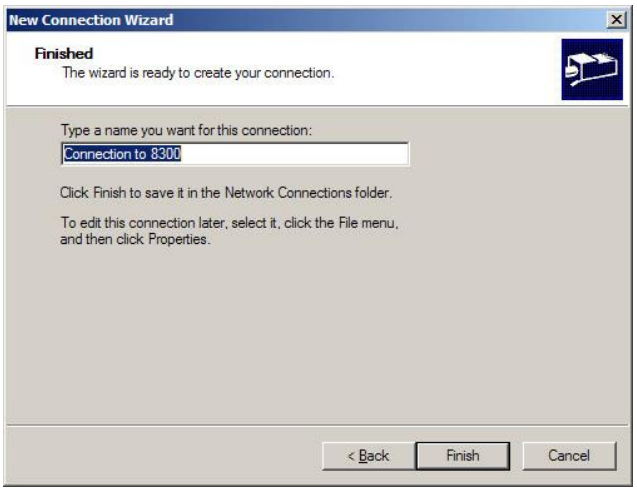
i) In the drop-down box, select the serial port you will be using to make the connection.

j) Click "Next."



k) Type in a name for your Connection such as: "Connection to 8600."

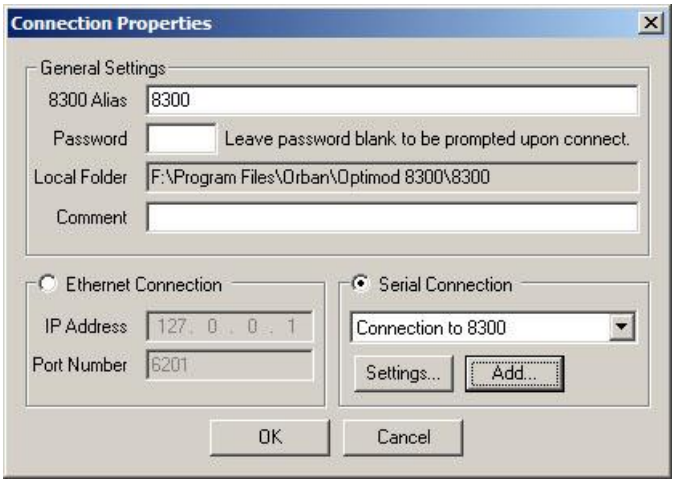
l) Click "Finish."



m)Click "Yes."

B) Edit your new Direct Connection properties:

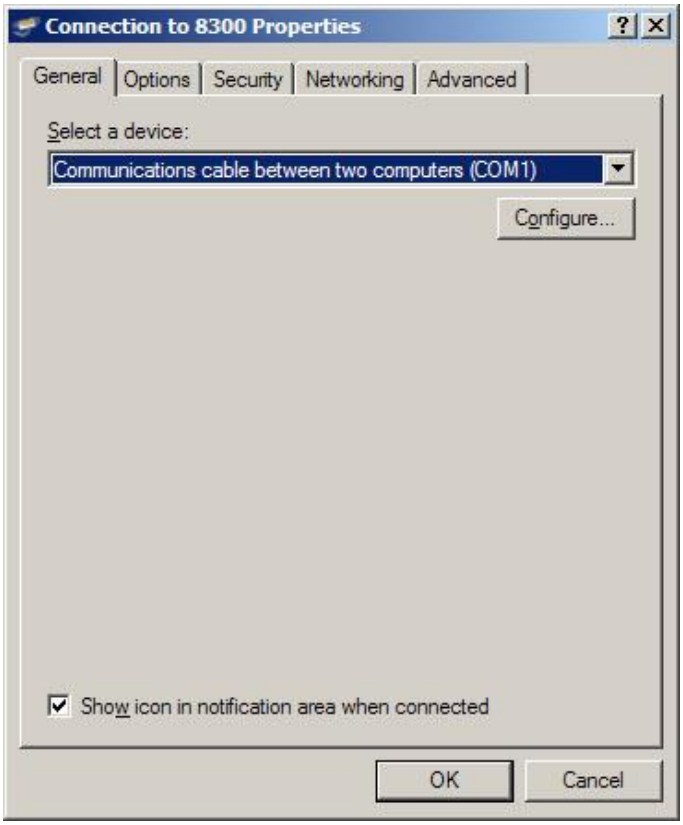
a) Click "Settings."



b) Click the "General" tab.

c) Select the device you set up in step (i) on page 2-82. This will usually be "Communications cable between two computers (COM1)."

d) Click "Configure."

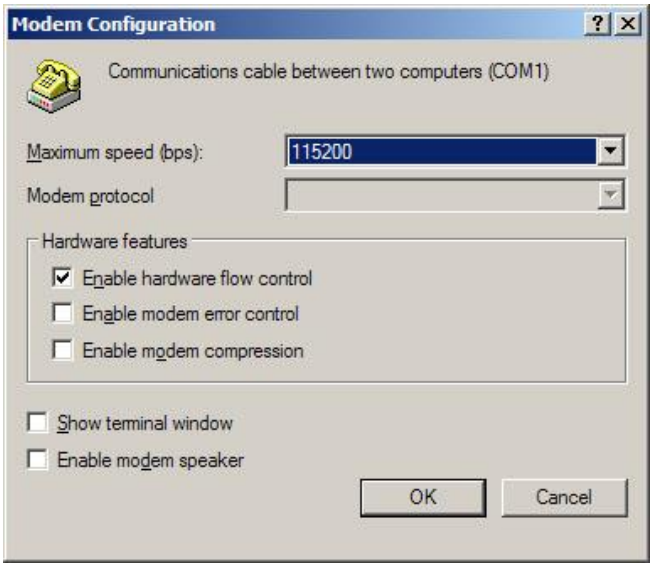


e) Set the "Maximum Speed (bps)" to 115200.

f) Check "Enable hardware flow control."

g) Make sure all other hardware features are unchecked.

h) Click "OK."



i) Select the Networking tab.

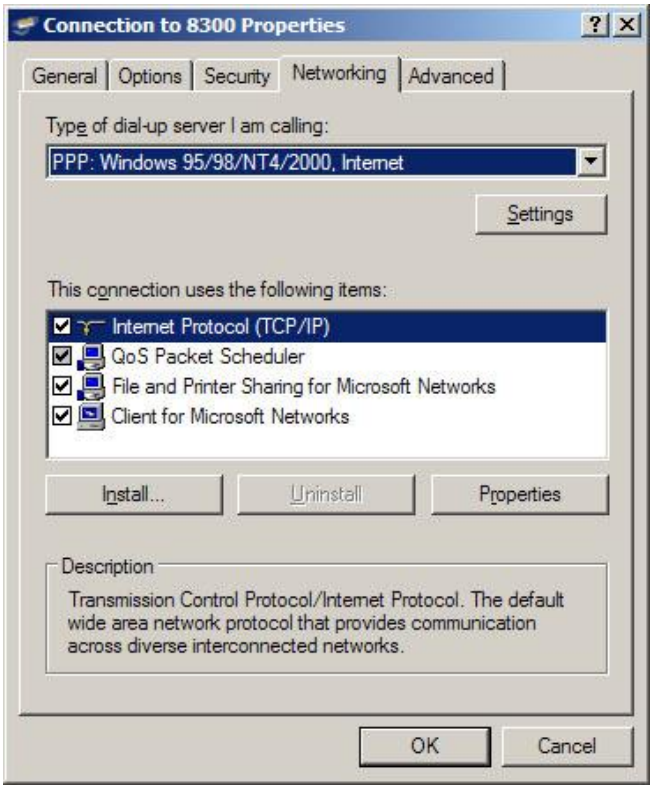
j) Make sure that "PPP: Windows 95 / 98 / NT 4 / 2000, Internet" appears in the "Type of dial-up server I am calling" field.

k) Make sure that "Internet Protocol (TCP/IP) is checked.

You may leave "File and Printer Sharing for Microsoft Networks" and "Client for Microsoft Networks" checked if you like

l) Click "OK."

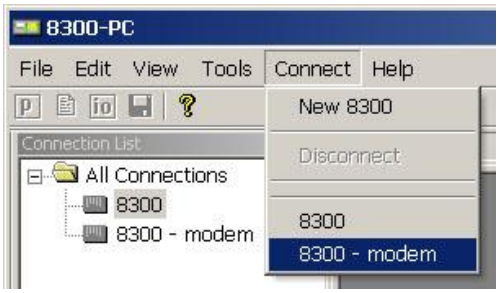
m) When the "Connection properties" window appears, click "OK."



2. Launch an existing Windows XP Direct connection.

Once you have set up a "connection" specifying Direct Connect in the 8600 PC Remote application (see *To set up a new connection* on page 3-89), choosing this

connection from 8600 PC Remote automatically opens a Windows Direct Connection to your 8600.



You can connect by selecting the desired connection from the drop-down list in the CONNECT menu.

You can also connect by double-clicking the connection in the "Connection List" window.

A dialog bubble will appear on the bottom right hand corner of the screen verifying your connection if the connection is successful.

If you have trouble making a connection, refer to *Troubleshooting Windows XP Direct Connect* on page 5-10. If you have trouble the first time after creating a connection according to the instructions above, try restarting your computer to clear its serial port.

3. **To change the properties of an existing connection:**

   Right-click the connection in the "connection List" window and choose "Properties." The "Connection properties" window opens (see page 2-76).

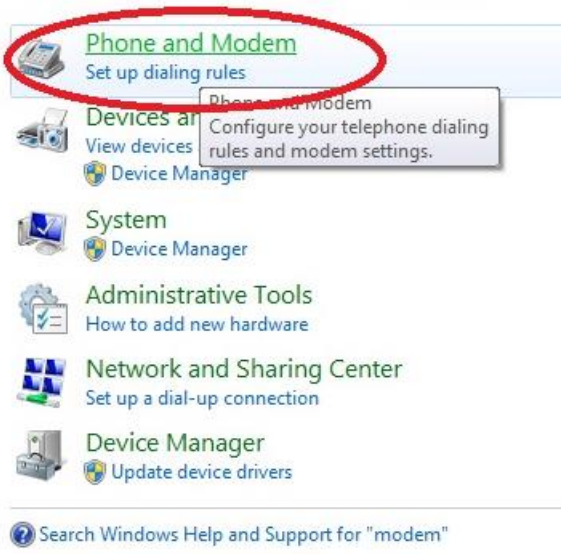
Connecting Using Windows 7 Direct Serial Connection:

You must install the Windows 7 direct serial connection as a modem device using the Modem setup procedures as shown in the steps below.

1. **Add and configure a Direct Connection for Windows 7.**

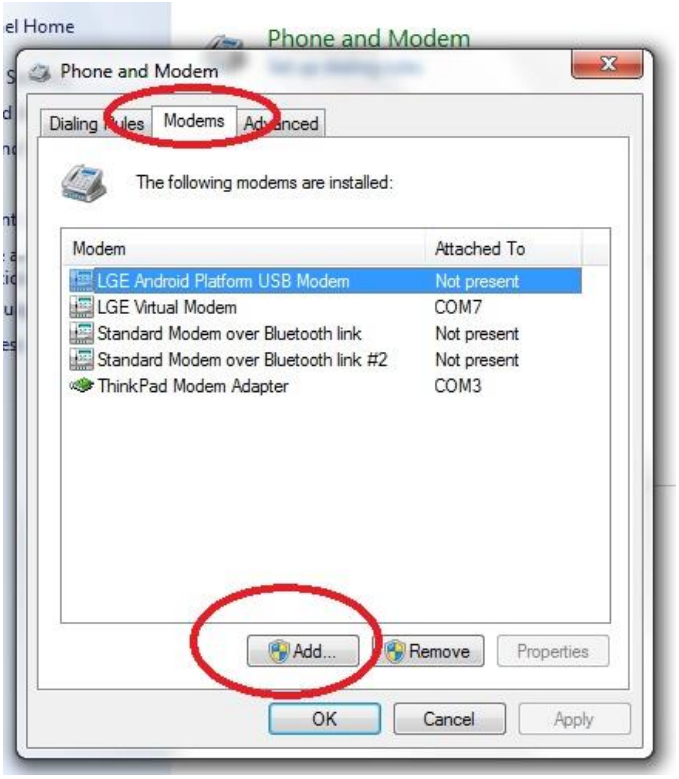
   A) Go to the Control Panel. Find the Modem applet. This is normally under the Phone and Modem section.

   Click on it to start the Phone and Modem applet.



B) In the Phone and Modem applet, click on the Modems tab and click "Add."

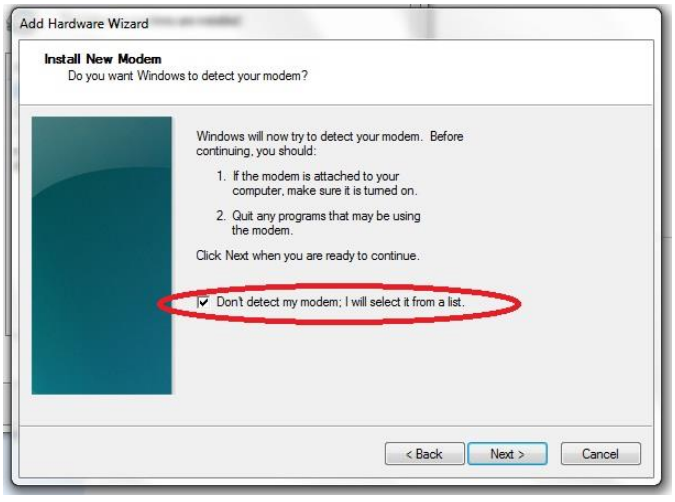
You need administrator's rights to do this. If UAC comes up, provide the relevant credentials and proceed.



C) The Add Hardware Wizard will appear.

a) Tick "Don't detect my modem; I will select it from a list."

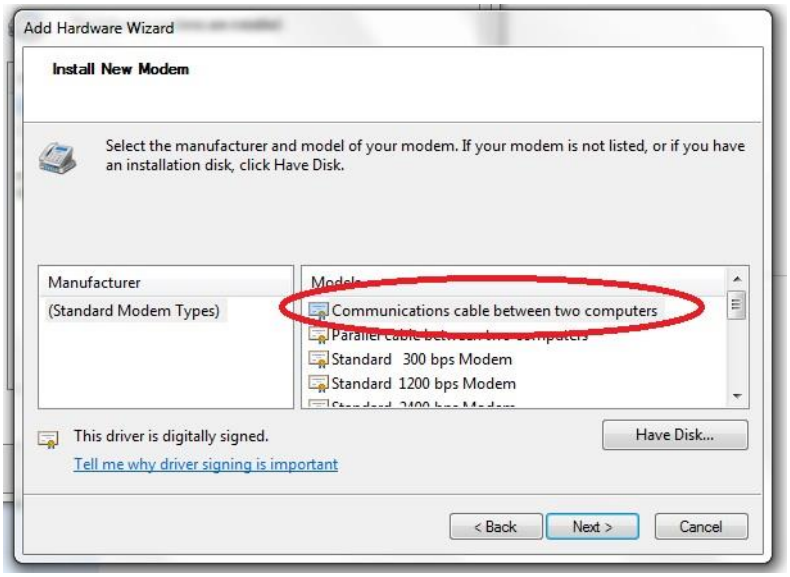
b) Proceed to next step by clicking on the NEXT button.



D) The *Install New Modem* window will appear.

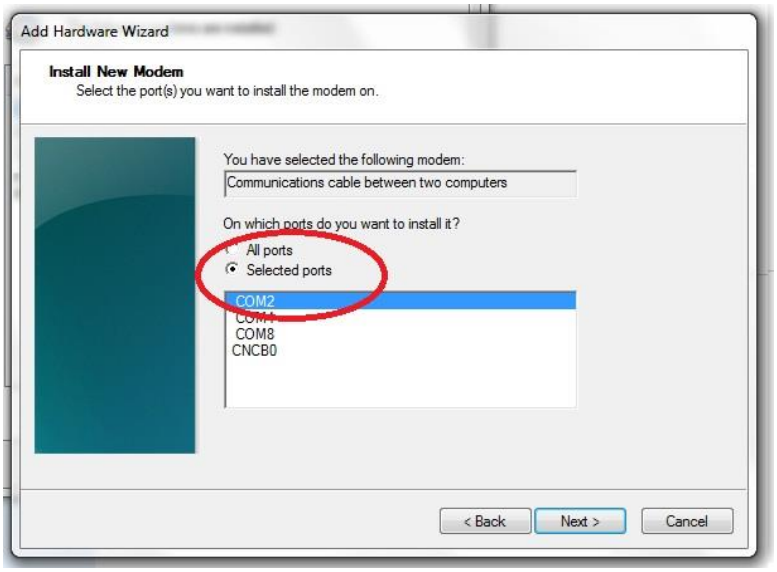
a) Select Communications cable between two computers.

b) Proceed to next step by clicking on the NEXT button.

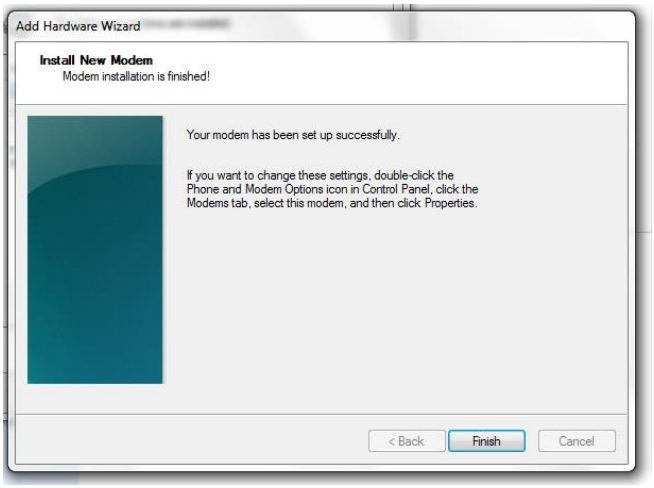


E) Select the Serial com port to which the NULL cable is connected.

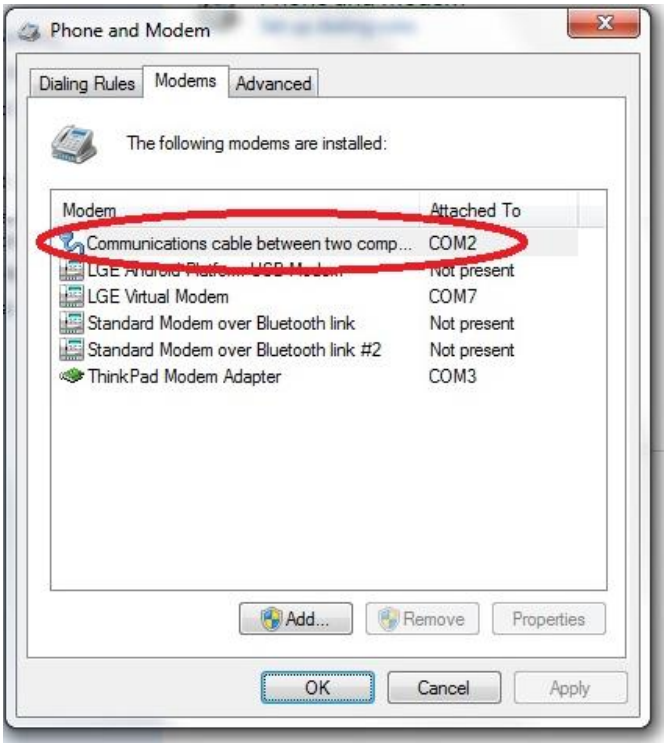
Proceed to next step by clicking on the NEXT button.



F) At the *Modem installation is finished* window, click FINISH to complete the installation.



G) Once you are back at the *Phone and Modem* window, you will see your newly installed communication cable attached to the serial com port that you specified earlier.



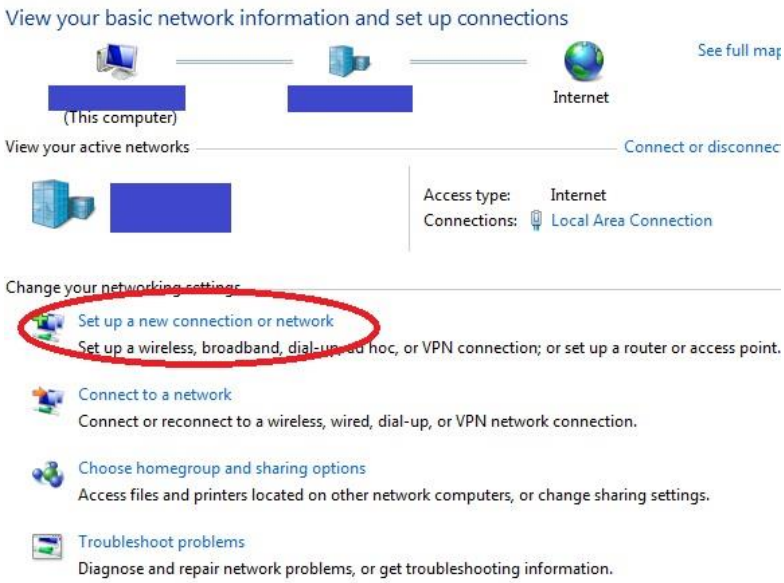
This completes the installation of the device driver for the device "Communications cable between two computers." This configuration is also commonly known as "Direct Cable Communications."

2. Set up the Network for the Direct Cable Connection.

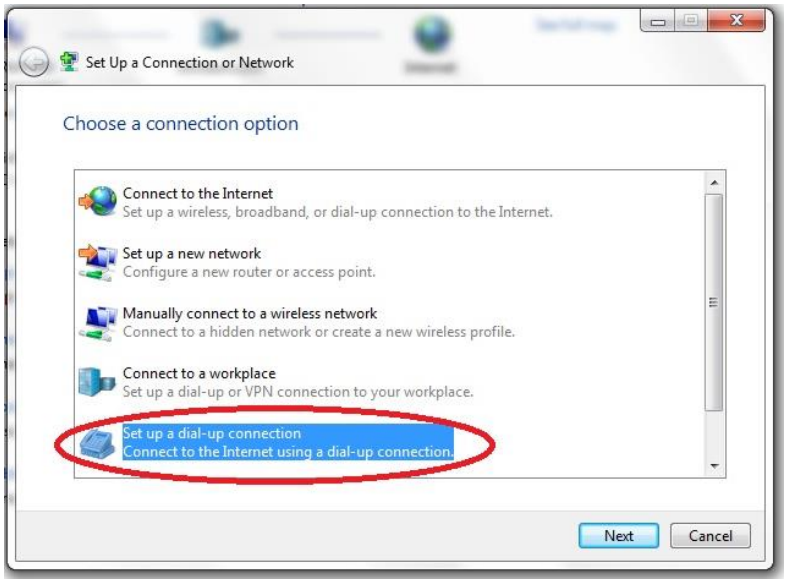
You must next bind the Direct Cable Connection driver to a particular Network. This is where Microsoft will make this connection using the PPP protocol.

It will do this for you automatically. Previous Windows version requires you to install the PPP protocol separately.

A) Go to the *Network and Sharing Center* and click on *Set up a new connection or network link*.

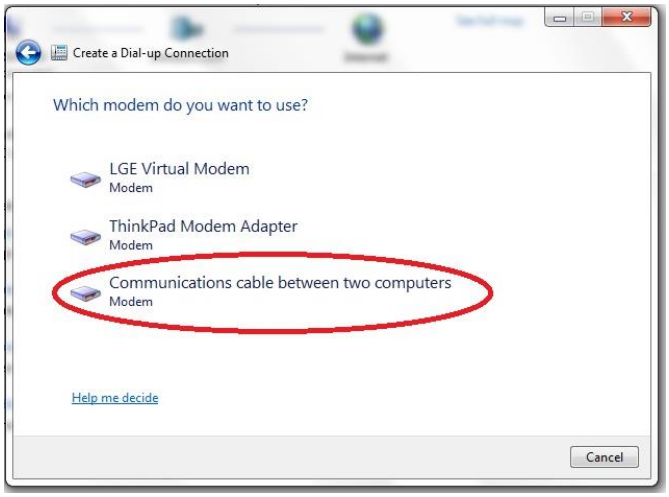


B) In the *Choose a connection option* window, select *Set up a dial-up connection*.



C) If you are asked *Which modem do you want to use?*, select *Communications cable between two computers/modem*.

This only query will open appear if you have configured more than one modem device.

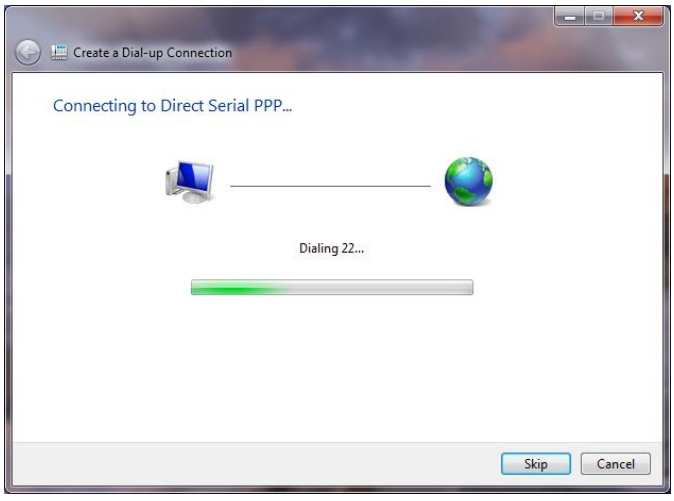


D) When prompted to "Type the information from your Internet Service Provider," enter a dummy number to the phone number field; Windows will not use it.

E) Choose a name for your connection and enter it into the *Connection name:* box. Choose a name that will remind you that this is a PPP connection, such as "Direct Serial PPP."

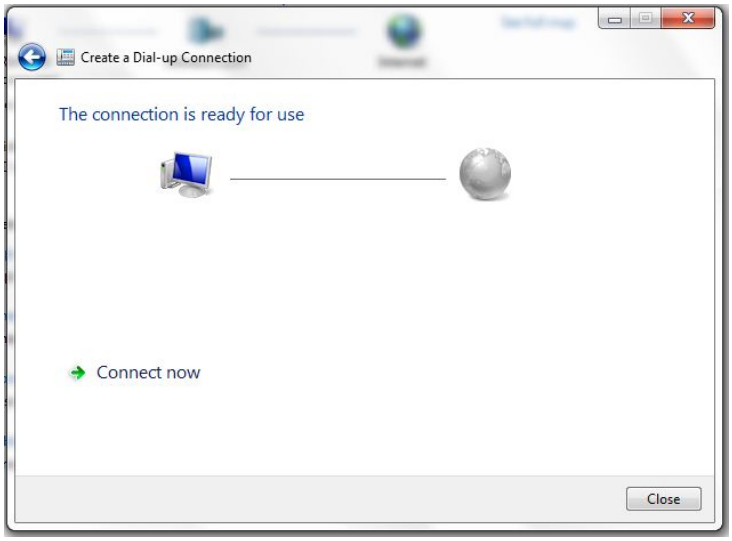
F) Proceed to next step by clicking on the CONNECT button.

G) Click the SKIP button.



Windows will emit a message stating *The connection is ready for use*. However, you must to configure some of the PPP settings before you can make a connection to your Optimod

> Although you did not specifically install anything that states "PPP," the PPP protocol has nevertheless been installed.



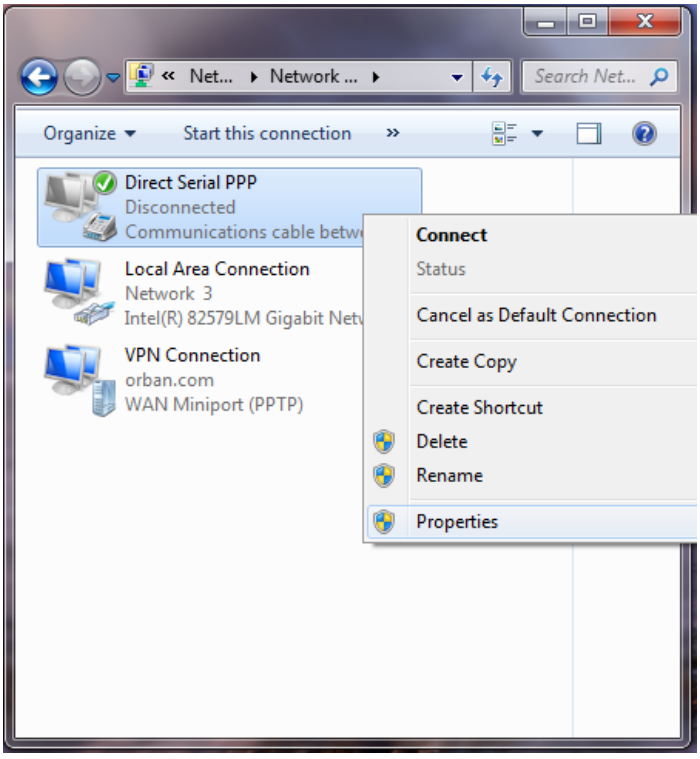
3. Configure the Direct Cable Connection adapter.

In this step, you will customize the PPP settings on Windows 7 so that it can talk to your Optimod.

A) Go to the Windows 7 Network and Sharing Center and click the *change adapter settings* link. This is on the left side of the window.

B) In the *Network adapter* window, right-click the *Direct Serial PPP* icon and click on the properties.

You need administrator rights to proceed from here. If UAC comes up, provide the relevant credentials and continue.



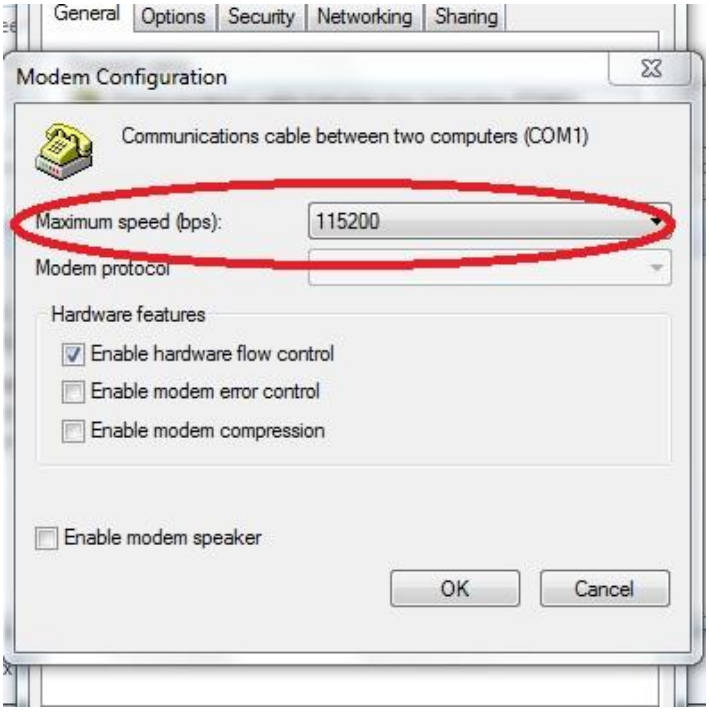
C) In the Direct Serial PPP Properties window, select the General tab.

D) Click the Configure button and select 115200 bps.

E) Click OK to close the window.

F) Make sure your bps settings saves correctly:

a) Dismiss the Direct Serial PPP Properties window by selecting OK.



b) Reopen the Properties window and select the General tab again. If your bps setting is correct, the value has saved and you may skip to step (G) below.

c) If it is not correct, reset it to 115200 bps.

d) Click OK to close the window.

e) Click OK to dismiss the Properties window.

f) Restart your computer.

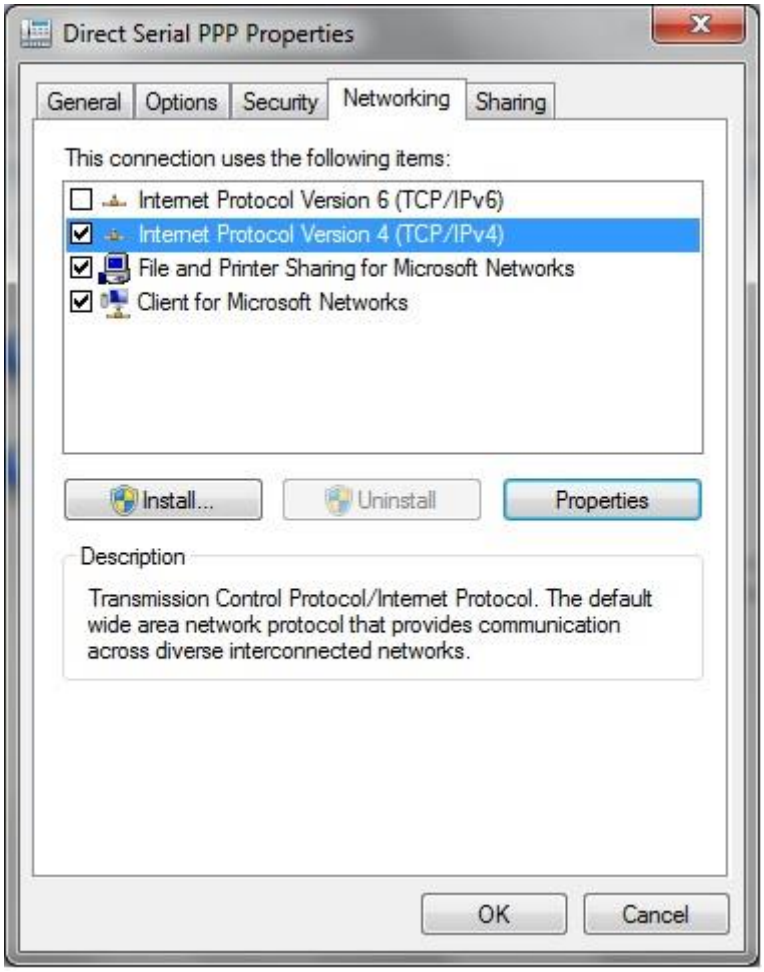
Restarting should ensure that the bps setting is saved.

G) Select the Networking tab.

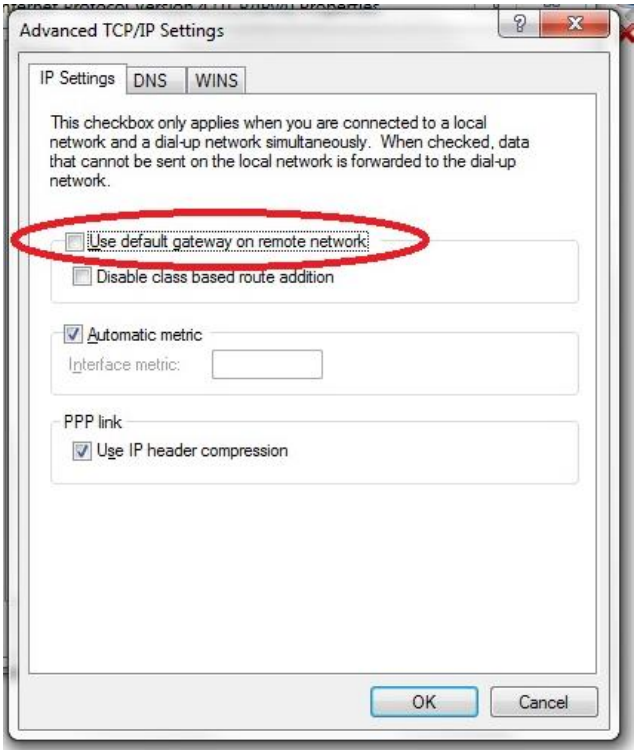
a) Unselect the Internet Protocol Version 6.

b) Click on the Internet Protocol Version 4.

c) *Click the Properties button.*



H) In the *Advance TCP/IP Settings* window, click on the *Advance* button.



a) *Unselect Use default gateway on remote network.*

b) Click OK to close this window.

This prevents Windows 7 from routing all networking requests to your Optimod.

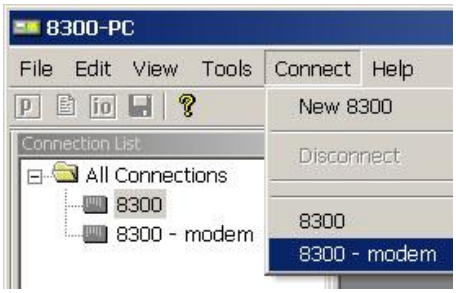
c) *Click OK to close the Internet Protocol version 4 Properties window.*

d) *Click OK at the Direct Serial PPP Properties window to save the Direct Serial PPP settings.*

4. Launch an existing Windows 7 Direct connection.

Once you have set up a "connection" specifying Direct Connect in the 8600 PC Remote application (see *To set up a new connection* on page 3-89), choosing this connection from 8600 PC Remote automatically opens a Windows Direct Connection to your 8600.

You can connect by selecting the desired connection from the drop-down list in the CONNECT menu.



You can also connect by double-clicking the connection in the "Connection List" window.

A dialog bubble will appear on the bottom right hand corner of the screen verifying your connection if the connection is successful.

If you have trouble making a connection, double-check step (3.F) on page 2-92. If you have trouble the first time after creating a connection according to the instructions above, try restarting your computer to clear its serial port.

5. To change the properties of an existing connection:

Right-click the connection in the "connection List" window and choose "Properties." The "Connection properties" window opens (see page 2-76).

Preparing for Communication through Modems

1. **Prepare your 8600 for a modem connection through the serial port.**

   See step 3 on page 2-64.

2. **If you have not already done so, create an 8600 passcode.**

   See *Security and Passcode Programming* on page 2-*43*.

3. **Modem setup:**

You will need two modems and two available phone lines, one of each for your PC and your 8600.

> Reminder: Orban supports only the 3Com / U.S. Robotics® 56kbps fax modem EXT on the 8600 side (although other 56kbps modems will often work OK).

Connect the modem to the 8600's serial port with a standard (not null) modem cable. The cable provided with your 8600 is a null modem cable and will not work.

You can use either an internal or an external modem with your PC.

A) Connect the telephone line from the wall phone jack to the wall connection icon on the back of the modem (modem in).

B) Connect the modem cable from the modem to the serial port of the 8600.

C) Set the modem to AUTO ANSWER and turn it on.

> For 3Com / U.S. Robotics® 56kbps fax modem EXT, set dipswitches 3, 5, and 8 in the down position to activate the AUTO ANSWER setting. All other dipswitches should be set to the up position.

Connecting Using Windows 2000 Modem Connection

This connection is used both for upgrading your 8600 and for connecting the 8600 PC Remote application to your 8600.

1. **Add and configure modem for Windows 2000:**

If your modem is already installed, skip to Launch a Windows 2000 Modem connection *on page 2-100.*

A) Install Windows 2000 modem:

Use either an internal modem or external modem with your computer.

a) If you are using an external modem, connect the modem to a serial port on your PC and make sure the modem is connected to a working phone line.

b) On your PC, click "Start / Settings / Control Panel / Phone and Modem Options."

c) Click the "Modems" tab.

d) Verify that your modem appears in the list available under "The following Modems are installed."

e) Verify that your modem is "Attached to" the correct port.

If your modem is unavailable or not attached to the correct port, you will need to Add it. See your Windows documentation.

f) If your modem is available in the list available under "The following Modems are installed" and it is attached to the correct port, then click "Properties" for that modem.

g) Make sure the port speed is set at 115200.

h) Click "OK."

B) Create a New Windows 2000 Dial-Up Connection:

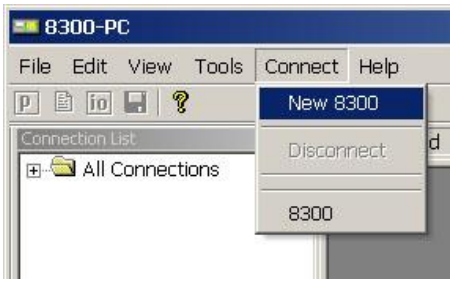
a) Click "Start / Settings / Network and Dial-up Connections / Make New Connection."

b) Once the New Connection Wizard has opened, Click "Next."

C) Create a New Windows 2000 Direct Connection:

a) Launch 8600 PC Remote.

b) Choose "Connect / New 8600"

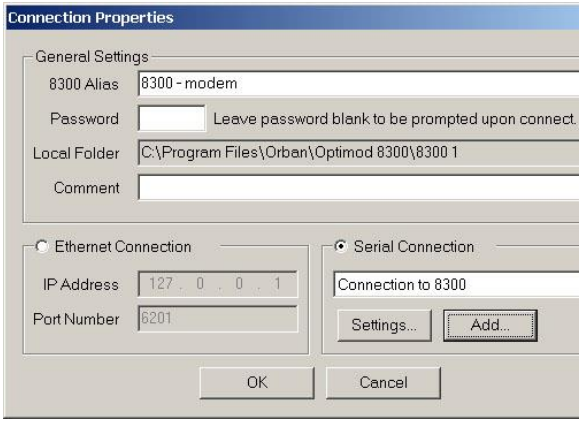


c) Give your 8600 a name (e.g., "KABC") by entering this name in the "8600 Alias" field.

d) If you wish to have 8600 PC Remote remember the password for this Optimod, enter the password in the "Password" field.

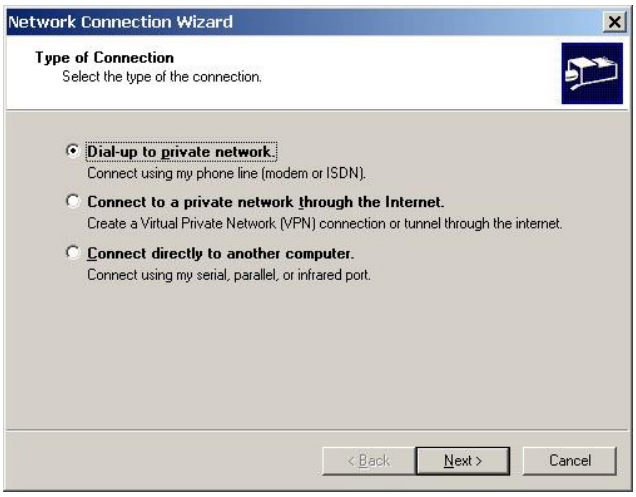
e) Select "Serial Connection."

f) Click the "Add" button.



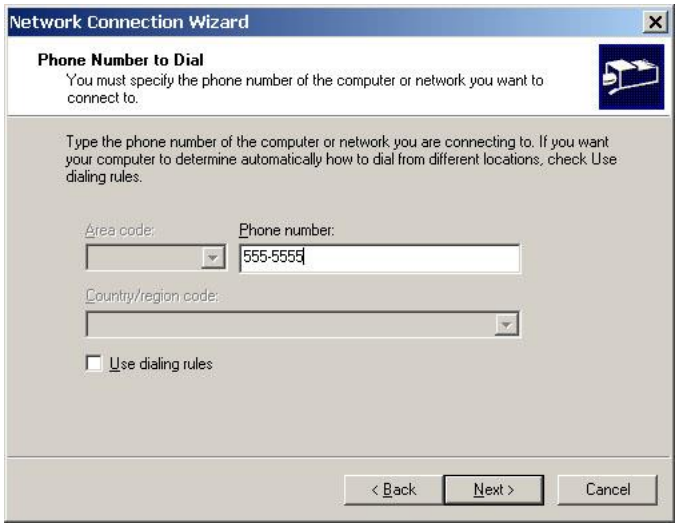
g) Select "Dial-up to private network."

h) Click "Next."



i) Enter the phone number of the modem connected to the 8600 that you are setting up.

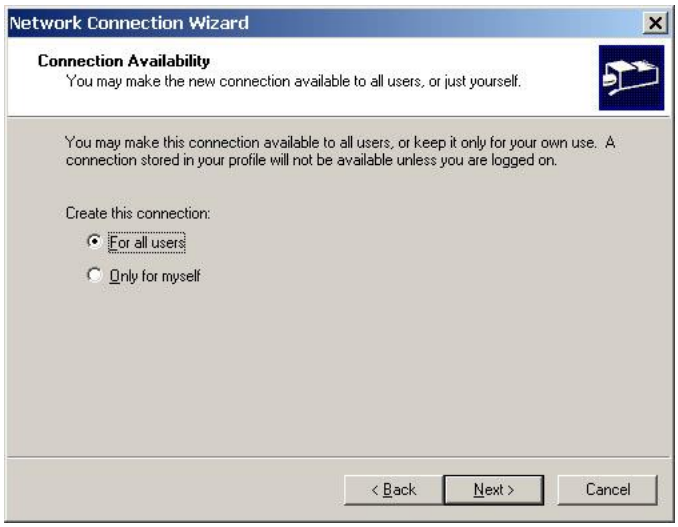
j) Click the "Next" button.



k) Select either "For all users" or "Only for myself."

The correct setting depends on how your network and security are configured.

l) This screen may not appear in computers set up for single users.



m) Click the "Next" button.

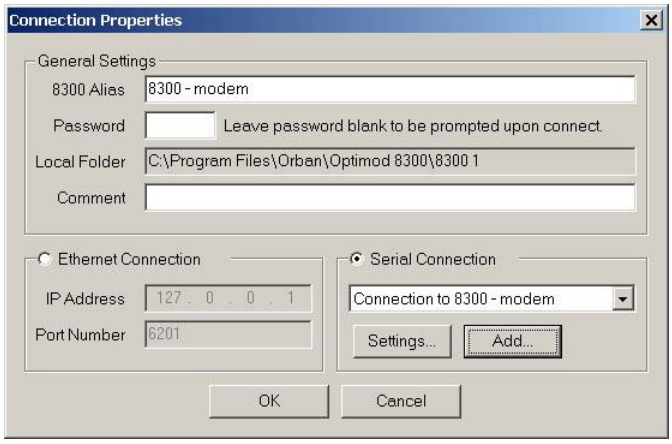
n) Type in a name for your Connection such as: "Connection to 8600 – Modem."

o) Click the "Finish" button.

p) Click "Yes."

D) Edit your new Direct Connection properties:

a) Click "Settings."



b) Click the "General" tab.

c) In the "Connect using" field, select the modem you will be using to make the connection on the PC side.

d) Click "Configure."

e) Set "Maximum speed (bps)" to "115200."

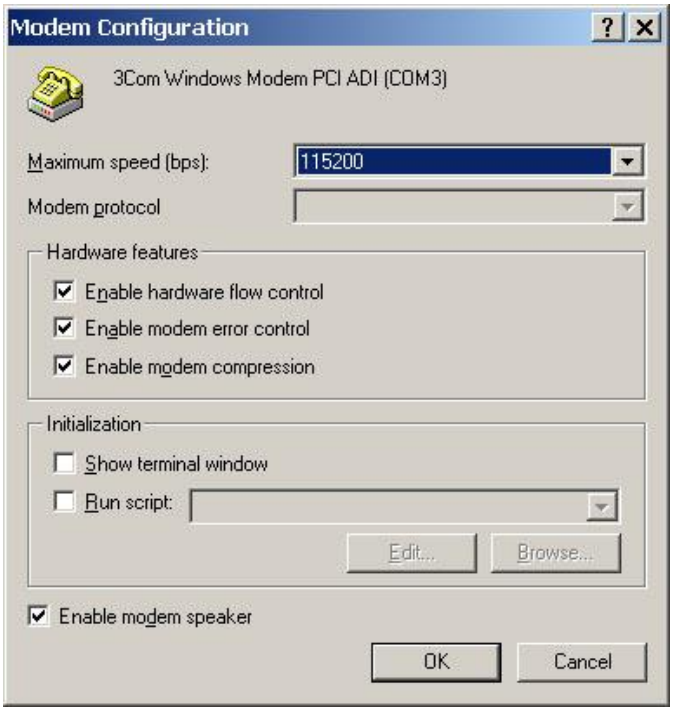
f) Check "Enable hard-ware flow control."

g) Check "Enable modem error control."

h) Check "Enable modem compression."

i) Make sure that all other boxes are not checked.

j) Click "OK."



k) Select the Networking tab.

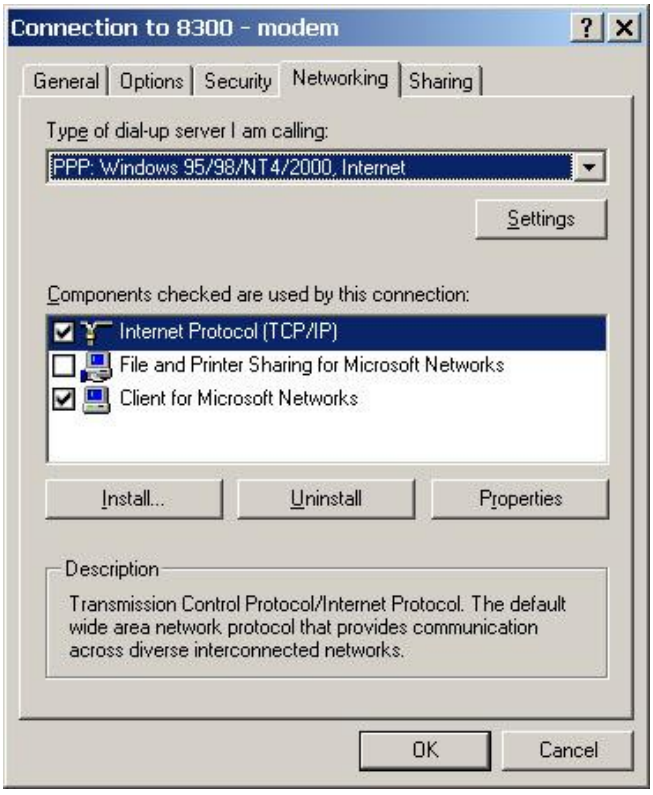
l) Make sure that "PPP: Windows 95 / 98 / NT 4 / 2000, Internet" appears in the "Type of dial-up server I am calling" field.

m) Make sure that "Internet Protocol (TCP/IP) is checked.

You may leave "Client for Microsoft Networks" checked if you like.

n) Click "OK."

o) When the "Connection properties" window appears, click "OK."



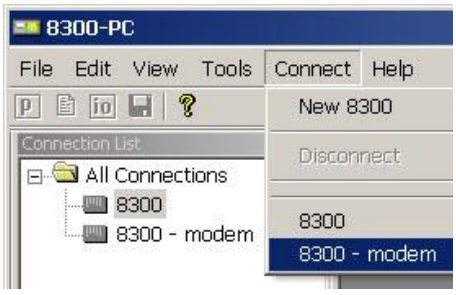
2. **Launch a Windows 2000 Modem connection.**

Once you have set up a "connection" specifying a modem connection in the 8600 PC Remote application (see *To set up a new connection* on page 3-89), choosing this connection from 8600 PC Remote automatically opens a Windows modem connection to your 8600.

You can connect by selecting the desired connection from the drop-down list in the CONNECT menu.

You can also connect by double-clicking the connection in the "Connection List" window.

If the connection is successful, a dialog bubble will appear on the bottom right hand corner of the screen verifying your connection.



3. **To change the properties of an existing connection:**

Right-click the connection in the "connection List" window and choose "Properties." The "Connection properties" window opens (see page 2-96).

Connecting using Windows XP Modem Connection

1. **Add and configure modem for Windows XP:**

Skip this step if your modem is already configured and working.

A) Configure the Windows XP PC ports:

Use either an internal modem or external modem with your computer.

a) If you are using an external modem, connect the modem to a serial port on your PC.

b) Make sure the modem is connected to a working phone line.

c) Click "Start / Control Panel / Systems."

d) Go to the "Hardware" tab and click "Device Manager."

e) In the Device Manager dialog box click the "+" next to the "Ports (COM and LPT)" icon.

A list will branch off, showing your available ports.

f) Double-click "Communications Port (COM1) or (COM2)," depending on how you set up your system.

The "Communications Port (Comx) Properties" dialog box opens.

Not all PCs have a COM2.

IMPORTANT: The COM port you choose at this point *must* match the COM port to which you connected your modem.

g) From the tabs at the top, choose "Port Settings" and configure the settings to match your PC modem.

If you are using a U.S. Robotics® external modem, the settings will be: Bits per second= 115200, Data bits = 8, Parity = None, Stop bits = 1, Flow Control = None.

h) When you are finished, click the OK button to close the "Communications Port (Comx) Properties" dialog box.

i) Click the OK button in the "Systems Properties" dialog window.

j) Close the "Control Panel" window.

If your modem is already installed, skip to Launch an existing Windows XP modem connection *on page 2-105.*

B) Install the Windows XP modem:

a) Use either an internal modem or external modem with your computer.

If you are using an external modem, connect the modem to a serial port on your PC and make sure the modem is connected to a working phone line.

b) On your PC, click "Start / Settings / Control Panel / Phone and Modem Options."

c) Click the "Modems" tab.

d) Verify that your modem appears in the list available under "The following Modems are installed."

e) Verify that your modem is "Attached to" the correct port.

If your modem is unavailable or not attached to the correct port, you will need to Add it. See your Windows documentation.

f) If your modem is available in the list available under "The following Modems are installed" and it is attached to the correct port, then click "Properties" for that modem.

g) Make sure the port speed is set at 115200.

h) Click "OK."

C) Create a new Windows XP modem connection:

a) Launch 8600 PC Remote.

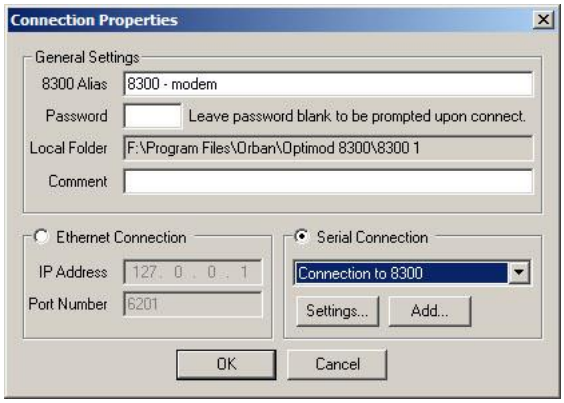
b) Choose "Connect / New 8600."

The Connection Properties window opens.

c) Give your 8600 a name (e.g., "KABC") by entering this name in the "8600 Alias" field.

d) If you wish to have 8600 PC Remote remember the password for this Optimod, enter the password in the "Password" field.

You must enter a valid password to connect. This means that at least one 8600 passcode must have been assigned via the 8600's front panel. (See *Security and Passcode Programming* on page 2-*43*.)



e) Click "Add."

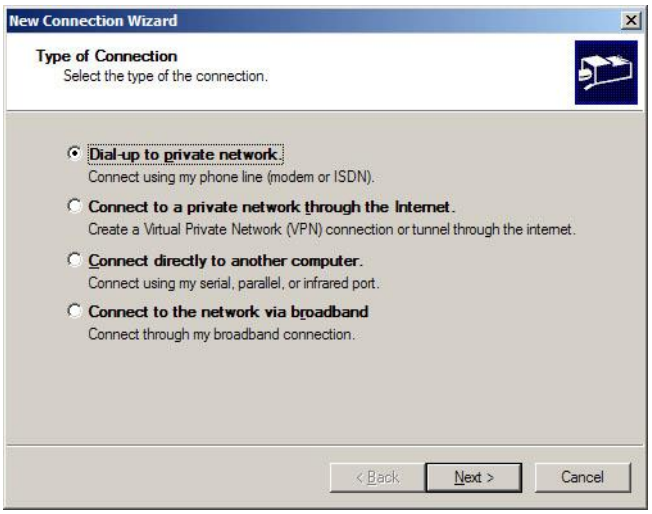
The Windows New Connection Wizard starts up.

f) Select "Serial Connection."

g) Click the "Add" button.

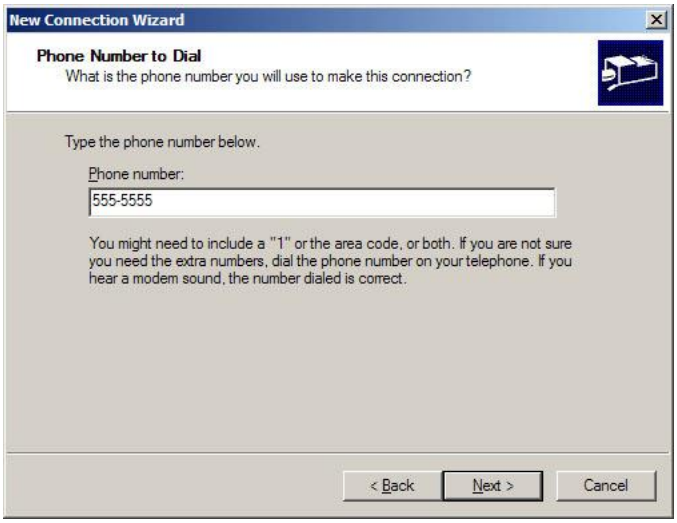
h) Select "Dial-up to private network."

i) Click "Next."



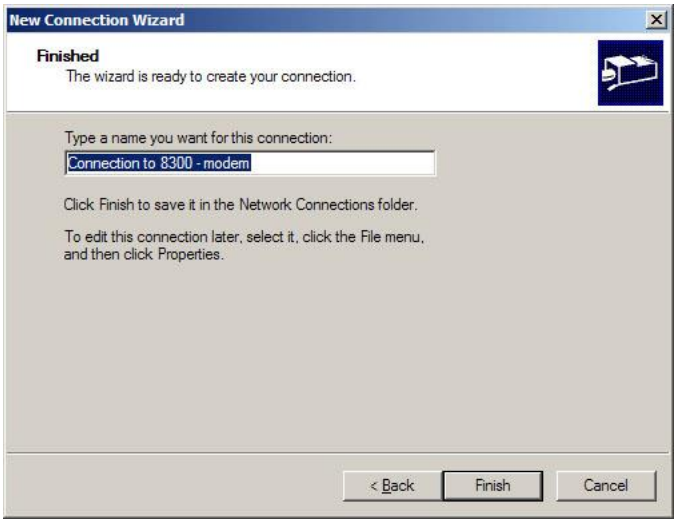
j) Enter the phone number of the modem connected to the 8600 you are setting up.

k) Click "Next."



l) Type in a name for your Connection such as: "Connection to 8600 – Modem"

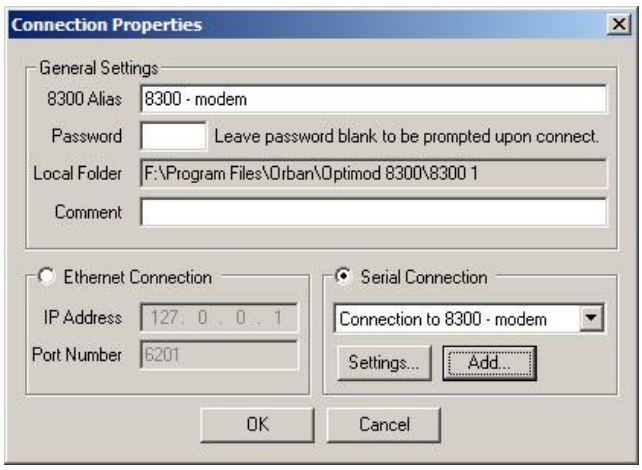
m)Click the "Finish" button.



n) Click "Yes."

D) Edit your new Direct Connection properties:

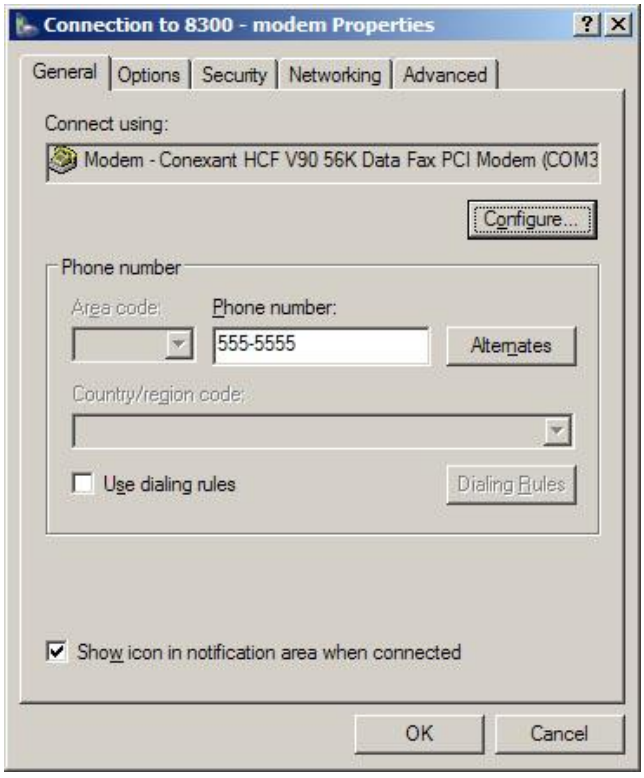
a) Click "Settings."



b) Click the "General" tab.

c) Select the modem you will be using to make the connection on the PC side.

d) Click "Configure."



e) Set "Maximum speed (bps)" to "115200."

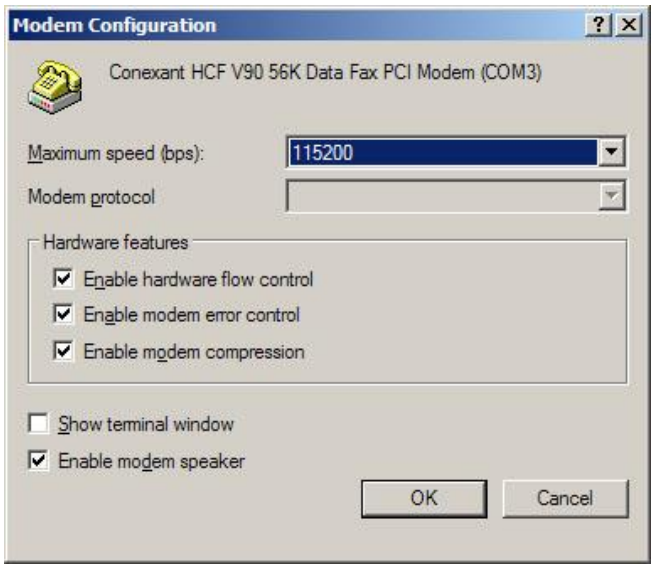
f) Check "Enable hardware flow control."

g) Check "Enable modem error control."

h) Check "Enable modem compression."

i) Make sure that no other box is checked.

j) Click "OK."



k) Select the Networking tab.

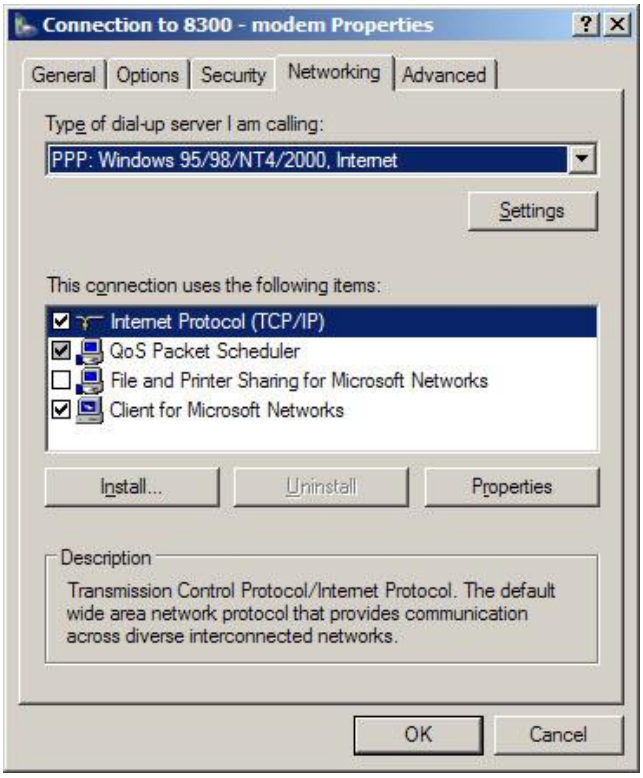
l) Make sure that "PPP: Windows 95 / 98 / NT4 / 2000, Internet" ap–pears in the "Type of dial-up server I am calling" field.

m) Make sure that "Internet Protocol (TCP/IP) is checked.

You may leave "Client for Microsoft Networks" checked if you like.

n) Click "OK."

o) When the "Connection properties" window ap-pears, click "OK."



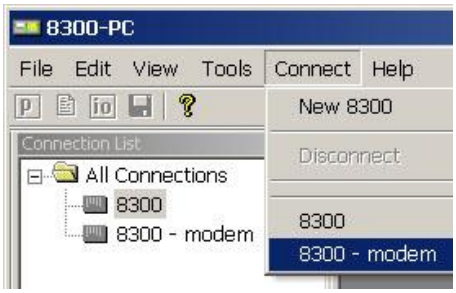
2. Launch an existing Windows XP modem connection.

Once you have set up a "connection" specifying a modem connection in the 8600 PC Remote application (see *To set up a new connection* on page 3-89), choosing this connection from 8600 PC Remote automatically opens a Windows modem connection to your 8600.

You can connect by selecting the desired connection from the drop-down list in the CONNECT menu.

You can also connect by double-clicking the connection in the "Connection List" window.

If the connection is successful, a dialog bubble will appear on the bottom right hand corner of the screen verifying your connection.



If you have trouble making a connection, refer to *Troubleshooting Windows XP Modem Connect* on page 5-11. If you have trouble the first time after creating a connection according to the instructions above, try restarting your computer to clear its serial port.

3. **To change the properties of an existing connection:**

   Right-click the connection in the "connection List" window and choose "Properties." The "Connection properties" window opens (see page 2-96).

SNMP Support

The SNMP (Simple Network Management Protocol) features allow you to monitor your Optimod's status and to send Alarm notifications via your Optimod's Ethernet connection to your network. It is beyond the scope of this manual to provide a general explanation of how SNMP works. The text below provides sufficient information to use your Optimod in your specific SNMP setup if you are already familiar with the general principles of setting up SNMP.

SNMP Network Setup

These controls are only available from Optimod PC Remote.

In the I/O menu, select the NETWORK tab to access the SNMP configuration controls.

- **SNMP** (Enable/Disable): enables or disables the SNMP feature. Select "enable" and disconnect from the PC remote to update the unit and allow SNMP access.

  If you wish to disable SNMP access after it has been enabled, select "disable" and then reboot your Optimod.

- **Primary Manager Address**: (255.255.255.255) sets the address of the Primary SNMP Manager.

- **Primary Manager Port**: (162) sets the port of the Primary SNMP Manager.

- **Secondary Manger Address**: (255.255.255.255) sets the address of a Secondary SNMP Manager.

- **Secondary Manger Port**: (162) sets the address of a Secondary SNMP Port.

SNMP Mib file

The 8600.mib file is in the location where you installed your PC Remote application.

The default 8600 install location is:

```
Program Files\Orban\Optimod 8600 PC Remote
or
Program Files(x86)\Orban\Optimod 8600 PC Remote
```

SNMP Default Settings

- **SNMP Agent**: Disabled
- **Primary Manager(Alarm) Address**: 255.255.255.255
- **Primary Manager (Alarm) Port**: 162
- **Secondary Manger (Alarm) Address**: 255.255.255.255
- **Secondary Manger (Alarm) Port**: 162

SNMP Features

The Get/Query functions return information when you request it from an SNMP client. When triggered, the Traps/Alert functions push information to the SNMP mailbox automatically.

Get/Query:

- **AES Input Silent**
- **AES Error**
- **Analog Input Silent**
- **Analog Input Active** (The analog input is selected as the input source.)
- **Digital Input Active** (The digital input is selected as the input source.)
- **Diversity Delay**
- **Primary and Secondary Manager IP**
- **Primary and Secondary Manager Port**
- **Station Name**
- **System Diagnostics Orban** (walks through all of the “get” commands and displays their status.)

- **Power Supply 1 is Inactive**: Indicates that Power Supply 1 in the dual-redundant power supply is not operating, either because it has failed or because it is not connected to the AC line.
- **Power Supply 2 is Inactive**: (see above)

Set/Control:

- **Diversity Delay**
- **Primary and Secondary Manager IP**
- **Primary and Secondary Manager Port**
- **Station Name**

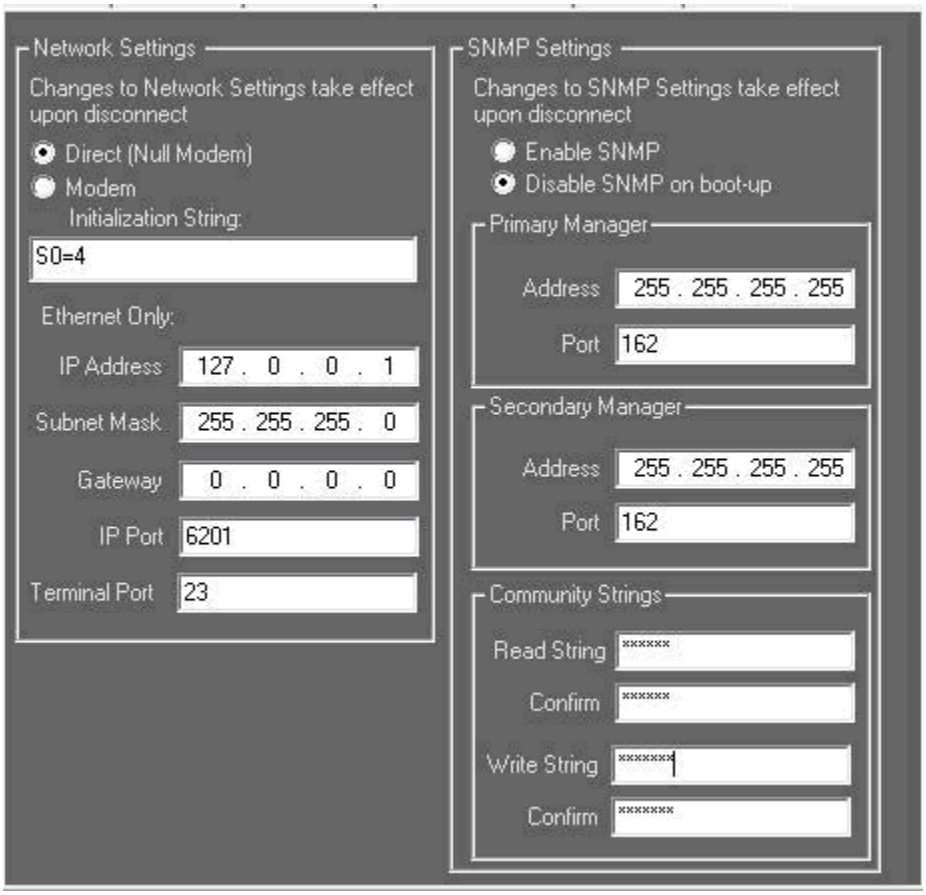
Traps/Alert:

- **Analog Input Silent**
- **Analog Input Active (selected as input source)**
- **AES Input Silent**
- **AES Error**
- **Digital Input Active** (The digital input is selected as the input source)
- **Power Supply 1 is Inactive**
- **Power Supply 2 is Inactive**

SNMP Community String:

The "SNMP Community string" is like a user id or password that allows access to a router's or other device's statistics. It is set at Optimod PC Remote to implement SNMP security. PRTG sends the community string along with all SNMP requests. If the community string is correct, the device responds with the requested information. If the community string is incorrect, the device simply discards the request and does not respond.

- **Read String** allows users to set a password for SNMP to retrieve information from the Optimod. Default is PUBLIC.
- **Write String** allows users to set a password for SNMP to write (set) information from the Optimod. Default is PRIVATE.



RDS/RBDS Generator

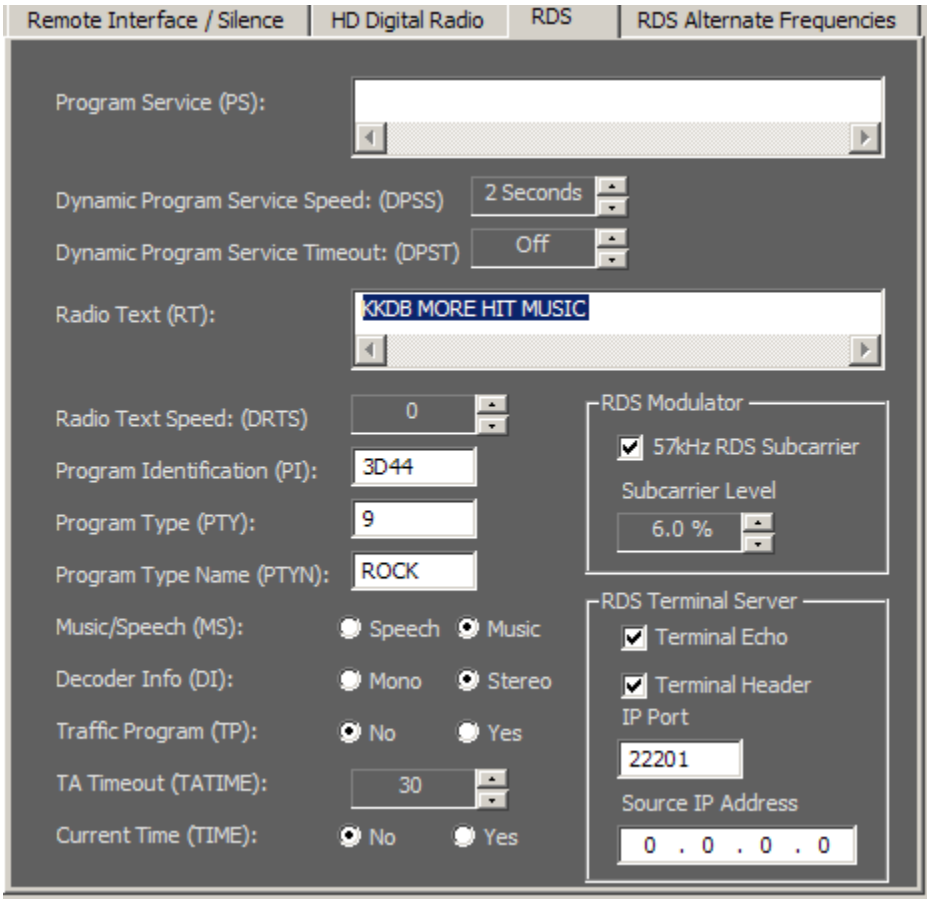
Your Optimod includes a full-featured RDS/RBDS generator that supports dynamic PS. We presume that you are already familiar with the basics of RDS and you wish to implement RDS via your Optimod. See the *References* on page 2-120 for more about RBS and RBDS.

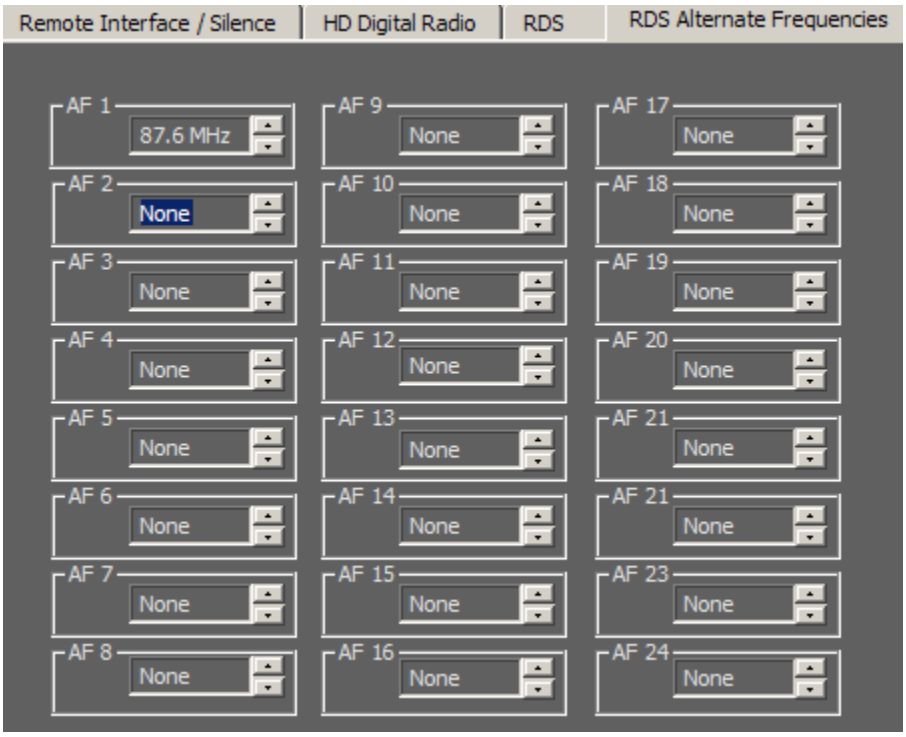
There are three ways to control RDS generation from the 8600:

- Use the **System I/O** [the default] to set up the RDS to generate static RDS.
- Change the RDS data dynamically by using **processing presets**.
- Change the RDS data dynamically by using **terminal control**.

You can use both User Presets and the RDS terminal control to change the RDS dynamically. For example, if you are running a mixed format where part of the broadcast day is devoted to news and part to jazz, you may want the RDS Program Type to change when you recall a preset appropriate for the format that is currently on-air.

Using System I/O





RDS	
Program Service (PS= / DPS=)	Default: undefined
Radio Text (RT= / TEXT=)	Default: undefined
Radio Text Speed (DRTS=)	Default: Off
Program Identification (PI=)	Default: undefined
Program Type (PTY=)	Default: undefined
Program Name (PTYN=)	Default: undefined
Music/Speech (MS=)	Default: Music
Decoder Info (DI=)	Default: Stereo
Traffic Program (TP=)	Default: 0
TA Timeout (TATIME)	Default: 30
57kHz RDS Subcarrier (RDS=)	Default: Off
Subcarrier Level (RDSLEVEL=)	Default: 6%
Terminal Echo (ECHO=)	Default: yes
IP Address	Default: 0.0.0.0
RDS Terminal Port (PORT1=)	Default: 22201
Current Time (TIME=)	Default: No
Alternate Frequency 1-24 (AF=)	Default: None

Table 2-1: System I/O RDS controls and defaults

In the **System I/O** RDS tab, turn on the RDS generator by setting 57 kHz RDS control to YES. You can also set up the Modulation level and all other RDS I/O controls in the System I/O RDS screens per the screenshot above. Table 2-1 shows a list of RDS controls available in System I/O.

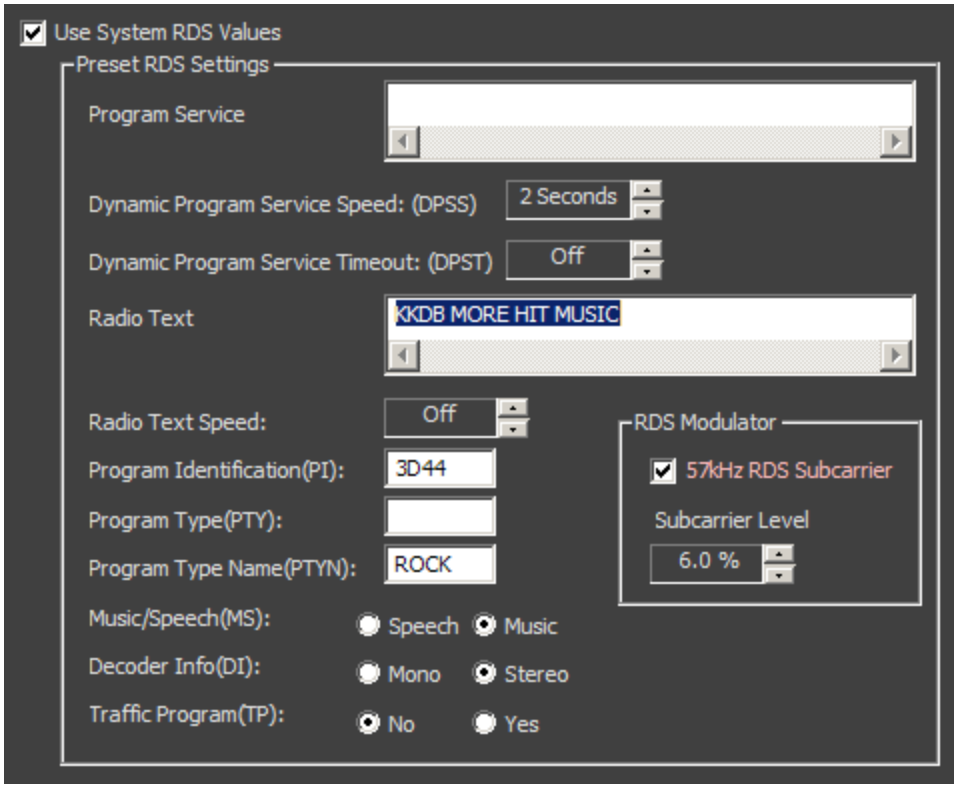
When you open I/O RDS screen in the PC remote when the System RDS controls are active, the Optimod will update the RDS output when a control is changed or if you select the DONE button.

Using Processing Presets

RDS adjustment via processing presets are available only from PC Remote.

To use **processing presets**, edit your on-air preset with the RDS checkbox USE SYSTEM RDS VALUES deselected. Then set the RDS preset controls as required and save the result as a new user preset. It is possible to make as many user presets as you want whose audio processing controls are set identically, but the RDS controls are set differently.

These controls are located in the processing preset's RDS tab in PC remote.



Note that because of buffering to ensure reliable RDS encoding, there is a delay of about four seconds between when you change a control and when your Optimod puts it on-air.

Using the Terminal Server

The **terminal server** is another way to change RDS dynamically. Your Optimod will update the outgoing RDS to whatever RDS control value it receives via the terminal, overwriting the system and preset RDS values. The terminal connects via TCP/IP, using the IP address and port specified in SETUP > RDS.

The values received via the terminal only persist until overwritten by a change in the system or processing preset. When you change a value in a System I/O RDS control at the Optimod front panel or via PC Remote, or if you recall a preset that has RDS activated, the new system/preset will overwrite whatever the Optimod last received from the terminal.

Although the terminal RDS will override the RDS settings in the active system or processing preset, it will not overwrite the RDS setting for the I/O or preset. For example, if you are using a processing preset to generate RDS with the ***PS 1234,*** then you send the terminal command ***PS ABCD***, the RDS will generate **ABCD,** but if you recall the processing preset again, the RDS generator will revert to transmitting ***1234*** because the terminal settings are not saved.

The `INIT` command clears any values previously sent from the Terminal connection and causes the RDS to revert to the System values.

By using the SAVE command, you can rewrite the SYSTEM RDS settings from the terminal server. For security, this command requires you to include a currently active password (see *Security and Passcode Programming* on page 2-43).

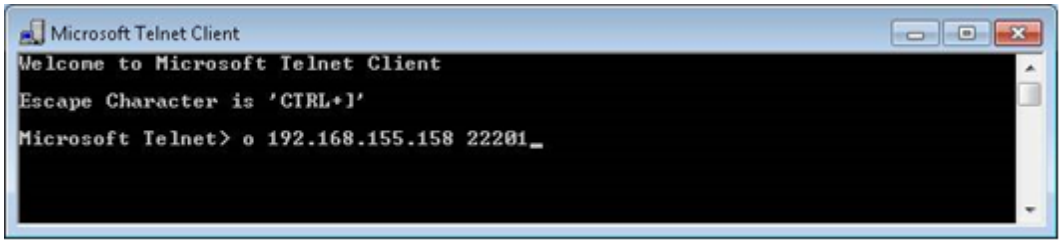
Using Telnet to Control the Terminal Server

Although the terminal server is usually used to accept commands from an automation system, batch file, or application like Modulation Index's PAD::bridge, you can also control RDS from Windows directly via the Windows Telnet command line utility or the free utility PuTTY.

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

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

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



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

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

The returned information will look similar to the following:

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

```
AF 20=0
AF 21=0
AF 22=0
AF 23=0
AF 24=0
```

Security

In the System I/O RDS control screen, you can set the RDS *Terminal control security* by specifying an RDS IP address from which to accept commands. Once set, this IP will be the only IP that can connect to the unit to update RDS. The 8500S 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 (as shown in Table 2-1) 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

Table 2-2 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 in Table 2-2

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=3
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↵
MS=	Music/Speech Switch	0 = Music 1 = Speech	MS=1↵ (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%)

[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
TIMEOUT=	RDS terminal connection timeout	Timeout (in minutes) between last transmitted command and when the Optimod disconnects automatically. Use it to allow the Optimod to reconnect automatically if the terminal connection is lost temporarily. *This command only affects the current connection; you must reissue it each time you connect.*	TIMEOUT=15↵ (current connection stays up for 15 minutes) TIMEOUT=0↵ (no auto-disconnect occurs)
VER↵	RDS Welcome Header	{ 0=No, 1=Yes } Sets whether the RSD welcome header is sent to the Network client upon connection..	VER 0↵
INIT↵	Use System RDS parameters	Use RDS parameters from System.	INIT↵
SAVE↵	Save RDS parameters	saves the current RDS parameters to the currently active RDS control set (either the System Settings or the User Preset group) NOTE: Any changes will not appear in the PC application if it is open. You must reconnect to see the saved values.	SAVE↵

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

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 2-2: Preset/Terminal RDS controls and defaults

Alternative Frequency Channel Numbers

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

MHz	CHAN	MHz	CHAN	MHz	CHAN	MHz	CHAN
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 2-3: Alternative Frequency Channel Numbers:

Program Type (PTY)

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 2-4: 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 following:

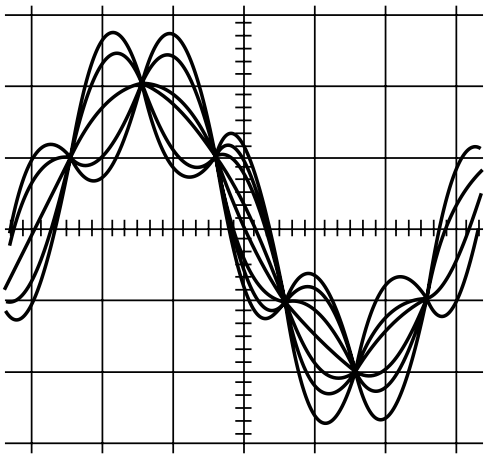


Figure 2-5: Pilot/SCA Phasing Scope Trace

To minimize the amount of peak level that the RDS subcarrier adds to the composite baseband, your Optimod's RDS generator is locked in quadrature with respect to the pilot tone, so Figure 2-5 is for reference only. If your transmission system is broadcasting stereo correctly, it will also correctly pass the phasing built into your Optimod's SCA generator.

References

- IEC 62016 Edition 2.0, 2009-07 (European RDS and most of North American RBDS)
- NRSC-4-B (ways in which the RBDS standard deviates from IEC 62016)
- NRSC-G300-B (guidelines for usage of RBDS)

Section 3 Operation

8600 Front Panel

Headphone Jack allows you to monitor the output of the processing through headphones. The headphones carry the same output as the rear-panel analog output: either analog-channel processing, HD processing, or low-delay monitor processing. Headphone impedance should be 75Ω or higher.

Headphone Level Control (the blue control knob to the right of the jack) adjusts headphone volume.

The red **Enter** button allows you to choose pop-up menu items, icons, and buttons. If you are in the Preset screen, it allows you to put a Factory or User Preset on-air once you have selected it.

If you edit a Factory Preset, you must save it as a new User Preset to retain your edit.

The green joystick, labeled **Locate**, is a pointing device that allows you to navigate to settings and controls on each screen. If multiple screens are available, pressing and holding the knob left or right moves you to the previous and next function screens.

The yellow **Escape** button allows you to navigate quickly to underlying screens, higher-level screens or the Meters screen, or displays the pop-up menu.

When a pop-up item, like Menu, is onscreen, *Escape* always returns you to the underlying screen.

Pressing *Escape* from a secondary screen page, like System Setup > Place / Date / Time 1 takes you back to the top level; in this case, the System Setup screen.

Escape from top-level screens (like the System Setup screen), brings you back to the Meters screen. If you are already in the Meters screen, *Escape* displays the pop-up Menu.

The **Control Knob** is the large blue knob on the front panel. Turning the knob scrolls through displayed lists (like the Preset screen list) or changes a setting that is highlighted onscreen (e.g., the setting last selected by the *Locate* joystick). Pushing the knob in (towards the front panel) displays the pop-up Menu over the previous screen.

Screen Display supplies control setting information and screen help, and displays the gain reduction and level meters (described directly below).

The 8600's color LCD displays the following meters and indicators:

In meters show the peak input level applied to the 8600's analog or digital inputs with reference to 0 dB = digital full-scale. If the meter reads at the top of the scale and the analog input is active, this indicates clipping in the A/D converter.

AGC meters show the gain reduction of the slow AGC processing that precedes the multiband compressor. Full-scale is 25 dB gain reduction.

> The AGC is a two-band unit with Orban's patented bass coupling system. The two meters indicate the gain reduction of the AGC Master and Bass bands.

Gate indicators show gate activity. They light when the input audio falls below the threshold set by the gate threshold controls. (There are two gating circuits—one for the AGC and one for the multiband limiter—each with its own gate threshold control.) When gating occurs, the AGC and compressor's recovery times slow drastically to prevent noise rush-up during low-level passages.

Multiband gain reduction meters show the gain reduction in the multiband compressor. Full-scale is 25 dB gain reduction. The MB GR METER switch (in INPUT/OUTPUT > UTILITIES) determines what signals the 2-Band and 5-Band Compressor gain reduction meters indicate. The switch can be set to FM, HD (digital radio), or SPLIT. In SPLIT mode, the 8600's front panel display shows the gain reduction of the FM analog multiband compressors on the left side of the split meters and the gain reduction of the digital radio compressors on the right. If the left and right channel gain reductions are not identical in a given band of the 2.0 processing, that band's meter displays the larger of the left or right channel gain reductions.

Many editing modes display small gain reduction meters in a window on the left side of the 8600's front-panel screen. In SPLIT mode, these meters are context-sensitive. They show the FM channel gain reduction when you are in an FM processing screen and show the HD gain reduction if you are in an HD processing screen. If the screen is not clearly FM or HD, the meters show the HD gain reduction. A "GR" label below the meters indicates whether the meters are displaying FM or HD gain reduction. A field below the meters indicates whether the meters are showing FM or HD gain reduction and whether the 8600's speech/music detector is detecting speech or music.

8600 PC Remote displays the gain reduction of the analog FM and HD multiband compressors on two metering groups, each containing either two and five meters. All meters are split and show the left and right channel gain reductions. A horizontal scroll bar below the meters allows you to drag the display left or right to expose the meters you wish to see.

> If the Five-Band structure is active, all the meters on the main meter screen display gain reduction (G/R) activity. If the Two-Band structure is active, only the two leftmost meters display G/R activity.
>
> The MB GR METER switch is not present in 8600FM units.

2B HF meters display the gain reductions in dB of the independent left and right channel high frequency limiters in the 8600's Two-Band structure. These meters appear only when the 8600 is in Two-Band mode.

Out meters display the 8600's instantaneous peak output level.

Comp meter displays the stereo encoder's output level before the COMP 1 or COMP 2 attenuators, using a linear percent scale over a 0% to 125% modulation range.

Multiplex Power / BS.1770 gain reduction uses a two-color display to indicate the action of the ITU Multiplex Power controller and the BS.1770 safety limiter in the analog radio processing chain. It shows how much the Multiplex Power Controller and BS.1770 Safety Limiter have reduced the clipper drive, reducing the average power in the processed audio. The BS.1770 gain reduction is shown in cyan and the multiplex power gain reduction is shown in dark blue.

> This meter, labeled LOUDNESS, is displayed on the 8600's screen. It always appears when the Two-Band Structure is active. If the Five-Band Structure is active, it only appears if the Multiplex Power Controller or BS.1770 meter is turned on.

Multiplex Power Level meter indicates the amount of multiplex power produced by the 8600. The meter complies with the ITU-R BS.412 standard and is calibrated and with reference to the 8600's 100% peak modulation level.

HD (HD Digital Radio) G/R meters show the gain reductions in the left and right HD look-ahead limiters.

> These meters only appear when the METER SEL Switch (on the HD DIGITAL RADIO page of the 8600's INPUT/OUTPUT menu) is set to HD GR. These meters will not appear on 8600FM units.

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-to-average 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 im-

portant 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 8600 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 look-ahead 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 8600, the tradeoff between brightness and the other two parameters has been considerably improved (by 2.5 – 3 dB above 6 kHz) when an "MX" preset is active compared to when an "8500-style" preset is active. Nevertheless, all intelligent processor designers must acknowledge and work within the laws of physics as they apply to this trade-off.

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

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

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.

OPTIMOD-FM—from Bach to Rock

You can adjust OPTIMOD-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 OPTIMOD-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 8600 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 preemphasis / deemphasis process. Although the 8600 loses less high frequency energy than any previous Orban processor (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[1], 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 8600 processing is to use slow multiband release times and considerable band 4→band 5 coupling, which will not further exaggerate distortion already on the CD.

It is also wise to use a mic processor on talent microphones so that the audio density of speech material will more closely match that of the music. If you do not use a mic processor, you can alternatively set up the 8600's speech-mode processing parameters to create more density than do the music-mode parameters. However, the external mic processor's performance will be more predictable because the 8600 will detect speech with substantial background music or effects as "music" and will not apply "speech" mode processing to this material.

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[1] As of this writing, two audio restoration programs that offer this feature are Diamond Cut DC8 and iZotope Rx.

About the 8600's Signal Processing Features

Dual-Mono Architecture

The 8600 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 8600 would always process stereo program material.

Signal Flow

The signal flows through the 8600 through the following blocks:

- Input Conditioning, including sample rate conversion, defeatable 30 Hz highpass filtering, and defeatable phase rotation
- Stereo Enhancement
- Two-Band Gated AGC, with target-zone window gating and silence gating
- Equalization, including high-frequency enhancement
- 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 8600 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 8600's output spectral control is immaculate, ensuring maximum stereo and RDS coverage. Because there is very little energy above 16 kHz, the 8600's digital output will pass through any uncompressed digital STL without adding noticeable over-

shoot 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 8600 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.

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 (step 8 on page 2-32). If you are not using it and do not need to activate it smoothly "on-air," bypass it.

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 8600 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 8600'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 8600 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.

Multiband Compression: The multiband compressor/limiter can be operated in five-band or two-band mode. The 8600 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 full anti-aliased.

Ordinarily, 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 8600'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 8600's speech/music detector can activate it automatically.

> The Ultra-low Latency structure does not offer compressor lookahead.

"Intelligent" Clipping: In non–MX presets, the 8600 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.

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

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

Speech Mode: You can set many of the processing parameters separately for speech signals, as detected by the 8600's speech/music detector. This allows you to tune the processing separately for speech and music.

A SPEECH DETECT control allow you to force the 8600 into Music mode, overriding the Speech/Music detector. This control is contained in the processing preset. In five-band 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."
>
> By using the SPEECH DETECT control, you can override the speech/music detector. See *To Override the Speech/Music Detector* on page 3-71.

DSP-derived Stereo Encoder: The 8600's stereo encoder is derived from algorithms first developed for the high-performance Orban 8218 stand-alone encoder. The 8600's stereo encoder operates at 512 kHz-sample rate to ease the performance re-

quirements 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 8600 has two independent composite outputs, whose levels are both software-settable. For convenience, two SCA inputs sum into the 8600's analog composite output amplifier. The second SCA input can be configured to provide a 19 kHz-reference output for subcarrier generators that need it.

The 8600 does not digitize SCAs.

SSB Stereo Encoder Operation: Starting with version 1.1, the 8600 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 8600'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.

In SSB/VSB mode, the bandwidth of the 8600's composite output signal extends to 38,150 Hz when the 8600'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 8600's Multipath Mitigator, which removes interchannel 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 8600'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 8600'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 8600's composite output, the 8600 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 8600'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 8600 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 8600'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 8600's main clipping system to do the vast majority of the work because of its sophisticated distortion-controlling mechanisms. This means that the 8600 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 8600's processing—the 8600 gets competitively loud without composite clipping. However, it is also possible to reduce the drive level to the 8600'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 8600 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 8600 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 8600, preventing excessive reduction of the multiplex power.

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

See *ITU-R Multiplex Power Controller* on page 3-81.

Two-Band Purist Processing

In addition to five-band processing, suitable for pop music and talk formats, the 8600 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 8600'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. (See *FM→HD Control Coupling* on page 3-79.)

The peak limiter in the digital radio processing chain is a mastering-quality look-ahead 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 preemphasis 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 8600'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 iBiquity 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.

See *About the 8600's HD* / Digital Radio Processing (starting on page 3-71) for a complete description of the HD setup and subjective adjustment controls.

BS.1770 Compliance for Digital Radio

The 8600 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. See step 10 on page 2-32.

> Because it only offers processing for analog radio, model 8600FM does not include a BS.1770 Safety Limiter.

Input/Output Delay

The sophisticated look-ahead algorithms in the 8600 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 8600 needs to look ahead at the next part of the program waveform. (Slowly changing bass waveforms require particularly long look-ahead delays.) As digital on-air processing advances further and further from its analog roots, this is the inevitable price of progress.

The amount of delay depends on several things. Because of their unprecedentedly sophisticated processing, "MX" presets introduce delays of approximately 265 ms, which makes real-time monitoring through headphones impossible for talent. The 8600 offers a low-delay headphone monitoring output to solve this problem, which also occurs in HD Radio installation where the diversity delay is typically 8-10 seconds.

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

For a thorough discussion of delay issues and solutions, see *Monitoring on Loudspeakers and Headphones* on page 1-26.

Customizing the 8600's Sound

The subjective setup controls on the 8600 give you the flexibility to customize your station's sound. Nevertheless, as with any audio processing system, proper adjustment of these controls requires that you choose the desired audio texture for your station while managing the trade-offs between loudness, density, and audible distortion. The following pages provide the information you need to adjust the 8600 controls to suit your format, taste, and competitive situation.

When you start with one of our Factory Presets, there are three levels of subjective adjustment available to you to let you customize the Factory Preset to your requirements: Basic Modify, Intermediate Modify, and Advanced Modify.

Basic Modify

Basic Modify allows you to control three important elements of 8600 processing: the stereo enhancer, the equalizer, and the dynamics section (multiband compression, limiting, and clipping). At this level, there is only one control for the dynamics section: LESS-MORE, which changes several different subjective setup control settings simultaneously according to a table that we have created in the 8600's permanent ROM (Read-Only Memory). In this table are sets of subjective setup control settings that provide, in our opinion, the most favorable trade-off between loudness, density, and audible distortion for a given amount of dynamics processing. We believe that most 8600 users will never need to go beyond the Basic level of control. The combinations of subjective setup control settings produced by this control have been optimized by Orban's audio processing experts on the basis of years of experience designing audio processing and upon hundreds of hours of listening tests.

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.

In the 8600, LESS-MORE affects only the dynamics processing (compression, limiting, and clipping). Unlike the 8200, the 8600 has equalization and stereo enhancement that are decoupled from LESS-MORE. You can therefore change EQ or stereo en-

hancement and not lose the ability to use LESS-MORE. When you create a user preset, the 8600 will automatically save your EQ and stereo enhancement settings along with your LESS-MORE setting. When you recall the user preset, you will still be able to edit your LESS-MORE setting if you wish.

Intermediate Modify

Intermediate Modify is a compromise between Basic Modify and Advanced Modify. It allows adjusting the dynamics section at approximately the level of control available in Orban's 8200 processor. The controls are not extremely dangerous (although you can still get into trouble if you try hard enough). Most people will never have any reason to go beyond Intermediate Modify, even if they want to create a "signature sound" for their station.

Note: Intermediate Modify does not provide LESS-MORE control. Furthermore, once you have edited a preset's dynamics parameters in Intermediate Modify, LESS-MORE control is no longer available in Basic Modify and will be grayed-out if you access its screen. As noted above, we recommend using the Basic Modify LESS-MORE control to achieve a sound as close as possible to your desired sound before you make further modifications at the Intermediate Modify level.

Advanced Modify

If you want to create a signature sound for your station that is far out of the ordinary, or if your taste differs from the people who programmed the LESS-MORE tables, Advanced Modify is available to you. At this level, you can customize or modify any subjective setup control setting to create a sound exactly to your taste. You can then save the settings in a User Preset and recall it whenever you wish.

Compressor attack times and thresholds are available, along with settings affecting the automatic clipping distortion control. These controls can be exceedingly dangerous in inexperienced hands, leading you to create presets that sound great on some program material, yet fall apart embarrassingly on other material. We therefore recommend that you create custom presets at the Advanced Modify level only if you are experienced with on-air sound design and are willing to take the time to double-check your work on many different types of program material.

Important Note: Once you have edited a preset's dynamics parameters in Advanced Modify, LESS-MORE control is no longer available in Basic Modify and will be grayed out if you access its screen. As noted above, we strongly recommend using the Basic Modify 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.

A subtle side effect of this is the 8600's behavior when you switch preemphasis. All factory presets actually have two variations, one for 50µs and one for 75µs. The 8600 uses the appropriate variation automatically. However, once you have created a user preset, it will no longer automatically switch its parameters when you change preemphasis. As a rule of thumb, for similar high frequency texture, a 75µs presets should have its B4 COMPRESSION THRESHOLD control set 3 dB higher than an equivalent 50µs preset.

Gain Reduction Metering

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

To accommodate the FM preemphasis 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 (using Basic, Intermediate, or Advanced Modify), you can save it as a user preset. The 8600 can store an indefinite number of user presets, limited only by available memory.

The 8600 preserves any edited, unsaved preset until you recall another preset. This is true even if the 8600 is powered down or reboots. However, to ensure that you do not accidentally lose your work, it is wise to save as a User Preset any edited preset you want to keep. To save a preset:

A) Press the *Escape* button repeatedly until you see the main menu.

B) Locate to SAVE/SAVE AS and press *Enter*.

C) The SAVE PRESET screen appears.

D) Choose a name for your preset.

> Some non-alphanumeric characters (such as < and >) are reserved and cannot be used in preset names.

E) Use the "virtual keyboard" to create a preset name.

Use the *Locate* button to navigate to each character. Then press *Enter* to accept that character.

> The *Shift* key on the virtual keyboard changes it between upper and lower case.

F) *Locate* to the SAVE button and press *Enter*.

- 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 8600 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 8600 will give you an opportunity to choose a new name for the preset you are saving.

You can save user presets from the 8600 PC Remote application. (See *Using the 8600 PC Remote Control Software* on page 3-88.) Please note that when you save presets from the PC Remote application, you save them in the 8600's memory (as if you had saved them from the 8600's front panel).

The PC Remote application also allows you to *archive* presets to your computer's hard drive (or other storage device) and to restore them. However, archiving a preset is not the same as saving it. Archived presets reside on a storage medium supported by your computer (like its hard drive), while saved presets reside in the 8600's local non-volatile memory. You cannot archive a user preset until you have saved it. (See *To back up user presets, system files, and automation files onto your computer's hard drive* on page 3-91.)

To Delete a User Preset

A) From the 8600 pop-up (main) MENU display, toggle the LOCATE button to select SYSTEM SETUP, and then press the ENTER button.

> If the pop-up Menu isn't onscreen, press the blue Control Knob or the ESCAPE button.

B) LOCATE to SYSTEM SETUP > MEMORY icon and then press the ENTER button.

C) *Locate* to highlight the user preset you wish to delete.

> Factory presets cannot be deleted.

D) LOCATE to DELETE and then press the ENTER button to delete the highlighted preset.

E) Repeat steps (C) and (D) until you are finished deleting presets.

F) When you have finished deleting user presets, LOCATE to DONE and press the ENTER button.

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 8600.

In the 8600, 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 com-

ponents (like equalizers, compressors, limiters, and clippers), 8600'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 8600 realizes its processing structures as a series of high-speed mathematical computations made by Digital Signal Processing (DSP) integrated circuit chips. In the 8600, 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, phase-linear 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 8600 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 Optimod-FM 8200 version 3.0 except that the clippers run at 256 kHz sample rate and are anti-aliased, and it offers the same stereo enhancement, equalization section, advanced-technology AGC, composite limiter, and multiplex power controller as the other 8600 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		
Preset Names	**Source Preset**	**Normal Less-More**
PROTECT-0DB	PROTECTION-0DB	2.0
CLASSICAL-2 BAND	CLASSICAL-2 BAND	5.0
CLASSICAL-2B+AGC	CLASSICAL-2B+AGC	5.0
CLASSICAL-5 BAND	CLASSICAL-5 BAND	7.0
CLASSICAL-5B+AGC	CLASSICAL-5B+AGC	5.0
CLASSICAL-2B MX	CLASSICAL-2B MX	7.0
CLASSIC-2B+AGC MX	CLASSIC-2B+AGC MX	7.0
CLASSICAL-5B MX	CLASSICAL-5B MX	5.0
CLASSIC-5B+AGC MX	CLASSIC-5B+AGC MX	7.0
COUNTRY-MEDIUM	ROCK-SMOOTH	7.0
COUNTRY-LIGHT	ROCK-LIGHT	7.0
COUNTRY UL	COUNTRY UL	7.0
CRISP	CRISP	9.5
CRISP+LOWBASS MX	CRISP+LOWBASS MX	10.0
CRISP+MIDBASS MX	CRISP+MIDBASS MX	10.0
DANCE ENERGY	DANCE ENERGY	9.0
EDGE	EDGE	10.0
EDGE MX	EDGE MX	9.0
FOLK-TRADITIONAL	ROCK-SOFT	7.0
GOLD	GOLD	9.5
GOLD HEAVY MX	GOLD HEAVY MX	7.0
GOLD HEAVY BASS MX	GOLD HEAVY BASS MX	7.0
GOLD OPEN MX	GOLD OPEN MX	7.0
GREGG	GREGG	9.5
GREGG MX	GREGG MX	7.0
GREGG OPEN	GREGG OPEN	9.5
GREGG OPEN MX	GREGG OPEN MX	7.0
GREGG LL	GREGG LL	9.5
IMPACT	IMPACT	9.5
IMPACT LL	IMPACT LL	9.5
IMPACT MX	IMPACT MX	9.0
INSTRUMENTAL	JAZZ	7.0
JAZZ	JAZZ	7.0
MELLOW JAZZ MX	MELLOW JAZZ MX	9.0
SMOOTH JAZZ	SMOOTH JAZZ	9.0
SMOOTH JAZZ MX	SMOOTH JAZZ MX	9.0
LOUD-BIG	LOUD-BIG	9.0
LOUD-FAT	LOUD-FAT	7.0
LOUD-COMPRESSED	LOUD-COMPRESSED	9.5
LOUD-HOT	LOUD-HOT	8.5

FACTORY PROGRAMMING PRESETS		
LOUD-HOT LL	LOUD-HOT LL	9.5
LOUD-HOT+BASS	LOUD-HOT+BASS	9.5
LOUD-HOT+BASS LL	LOUD-HOT+BASS LL	9.5
LOUD-PUNCHY	LOUD-PUNCHY	9.0
LOUD+SLAM	LOUD+SLAM	9.0
LOUD-WIDE	LOUD-WIDE	9.5
NEWS-TALK	NEWS-TALK	7.0
ROCK-DENSE	ROCK-DENSE	7.0
ROCK GEN UL	ROCK GENERAL UL	7.0
ROCK-LIGHT	ROCK-LIGHT	7.0
ROCK-MEDIUM	ROCK-MEDIUM	7.0
ROCK-MEDIUM MX	ROCK-MEDIUM MX	7.0
ROCK-MEDIUM+MIDBASS	ROCK-MEDIUM+MID-BASS	7.0
ROCK-MEDIUM+LOW BASS	ROCK-MEDIUM+LOW BASS	7.0
ROCK-OPEN	ROCK-OPEN	7.0
ROCK-OPEN UL	ROCK-OPEN UL	7.0
ROCK-SOFT	ROCK-SOFT	8.5
ROCK-SOFT MX	ROCK-SOFT MX	7.0
ROCK-SMOOTH	ROCK-SMOOTH	7.0
ROCK-SMOOTH MX	ROCK-SMOOTH MX	7.0
SPORTS	SPORTS	7.0
URBAN-LIGHT	URBAN-LIGHT	7.0
URBAN-HEAVY	URBAN-HEAVY	7.0
URBAN UL	URBAN UL	7.0

Table 3-1: 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 8600'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 8600'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.

- Presets with "MX" in their names use the 8600's new-technology FM peak limiter to achieve the highest performance available in the 8600. MX presets offer greater transient impact, less preemphasis-induced high frequency loss, and lower perceived distortion than their 8500-technology counterparts. Achieving this level of performance requires the MX presets to have much longer delays (typically 265 milliseconds) than the 8500-style presets. This is too long to permit live off-air monitoring into talent headphones. This is the main reason why the 8600 offers 8500-style presets too—these presets have much lower delays than do their MX counterparts.

The remaining three preset types (listed below) are identical to their counterparts in Optimod-FM 8500:

- 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 MX, 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.

CLASSICAL, CLASSICAL MX: The 8600 offers 8500-style and MX variants of all classical presets. The MX versions provide less preemphasis limiting, lower distortion, and higher transient impact than the non-MX versions and are therefore recommended if their longer delay is acceptable.

As their names imply, the CLASSICAL 5-BAND and CLASSICAL 2-BAND presets are optimized for classical music, gracefully handling recordings with very wide dynamic range and sudden shifts in dynamics. The Five-Band version uses heavy inter-band coupling to prevent large amounts of automatic re-equalization, which could otherwise cause unnatural stridency and brightness in strings and horns and which could pump up very low frequency rumble in live recording venues.

The Five-Band preset defeats the AGC, using only the multiband compressor for gain reduction. It also defeats phase rotation to ensure the most transparent Five-Band sound available.

Even more transparent, "purist" classical processing is available from the CLASSICAL 2-BAND preset, which is phase-linear and which preserves the spectral balance of the original material as much as possible. However, if you need a bit more automatic re-equalization than the CLASSICAL 2-BAND preset provides, use the CLASSICAL 5-BAND preset.

CLASSICAL-5B+AGC uses the AGC set for 2:1 compression ratio. Because of the AGC, it affects more of the total dynamic range of the recording than does the CLASSICAL-5 BAND preset. However, the AGC provides extremely smooth and unobtrusive compression because of the gentle ratio and window gating. This preset uses the Five-Band compressor very lightly with a fast release time as a peak limiter. The AGC does almost all of the compression.

There is also a corresponding two-band preset called CLASSICAL-2B+AGC.

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.

While there are no specifically named Country MX presets, we suggest that modern country stations experiment with ROCK-MEDIUM MX, ROCK-SMOOTH MX, ROCK-

SOFT MX, and GREGG MX presets. Classic country stations should try the GOLD OPEN MX preset.

CRISP, CRISP MX: CRISP is an 8500-style preset that provides a bright upper mid-range sound by emphasizing frequencies around 6 kHz. It is a loud preset that is appropriate for mass-appeal music formats.

There are two MX CRISP presets:

- CRISP+MIDBASS MX has the same basic bass texture as the IMPACT MX preset and is targeted more toward smaller table radios with substantial bass response to 100 Hz but little below that.
- CRISP+LOWBASS MX has the same basic bass texture as the GREGG MX preset. It puts more energy in the frequency range below 120 Hz and is targeted more towards premium auto radios and other radios with substantial bass response below 100 Hz.

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.

EDGE MX is also appropriate for dance-oriented stations and will outperform DANCE ENERGY by producing more transient impact, more HF energy, and lower distortion. Because of its lower distortion, EDGE MX is appropriate for both 75 µs and 50 µs preemphasis.

EDGE, EDGE MX: This preset is designed for hit music, urban, and dance-oriented stations that prefer extremely punchy bass to fastidious distortion control. The 8500-style version uses HARD bass clipping, is loud, and has a bright high frequency texture. The MX version uses MEDIUM bass clipping and produces lower distortion. It gets its punch more from transient definition (snare drums, kick drum beaters, and other percussion are significantly more audible than in EDGE) than from brute-force bass power.

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 or ROCK-LIGHT presets.

ROCK-SMOOTH MX and ROCK-SOFT MX are also appropriate for this format.

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, distorted, 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 [ ] MX: There are three GOLD presets using MX technology: GOLD HEAVY, GOLD HEAVY BASS, and GOLD OPEN.

- 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 intended for formats playing music from the earliest rock-and-roll era through the mid 1970's. It is a relaxed, clean, easy-sounding 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.

  > This preset was designed by Greg Ogonowski and was inspired by early homebrew multiband processing found on some popular AM radio stations in the late 1960s and early 1970s, particularly CKLW.

GREGG, GREGG MX: GREGG, GREGG MX, GREGG OPEN, GREGG OPEN MX, 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 Orban's Vice President of New Product Development, 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.

In terms of loudness, midrange texture, and HF texture, the non-"MX" versions of these presets are similar to the LOUD-HOT+BASS presets. The MX versions fully exploit the 8600's ability to increase high frequency "air" and transient impact, and are excellent general-purpose presets for stations playing post-1980 recordings of various genres of popular music.

IMPACT MX: There are three IMPACT presets.

IMPACT MX is intended for CHR and similar formats where attracting a large audience (maximizing cume) is more important than ensuring long time-spent-listening. This very loud, bright, "major-market" preset has a great deal of presence energy that cuts through on lower-quality radios.

Its sound changes substantially as the LESS-MORE control is turned down—distortion decreases while bass punch and transparency improve. Therefore, exploring various LESS-MORE settings is very worthwhile with IMPACT, because, for many markets, this preset will be "over the top" if it is not turned down with LESS-MORE.

> GREGG MX may be preferable to IMPACT MX at IMPACT MX'S lower LESS-MORE settings.

IMPACT Is the 8500 version of this preset and has about 18ms of latency. Its audible performance is inferior to IMPACT MX, so it should only be used in circumstances where the delay produced by IMPACT MX is unacceptable.

IMPACT LL is the 13 ms-latency version of IMPACT. The same suggestions about exploring Less-More settings apply to this preset too.

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. Stations programming classic jazz should also investigate the MELLOW JAZZ MX preset.

Stations programming "smooth jazz" should investigate the SMOOTH JAZZ MX preset, which is louder and more "commercial"-sounding. There is also an 8500-style SMOOTH JAZZ preset available.

LOUD: There are several LOUD presets, all of which use "8500-style" technology and typically have 18 ms of latency. We have not made MX versions of the LOUD presets; we retained these presets in the 8600 mainly for historical reasons. Most were first introduced with Optimod-FM 8400 and some 8600 users may have custom presets based on them.

Before deciding to use a preset in the LOUD family, broadcasters wanting high loudness and dial impact should investigate the CRISP MX, EXTREME MX, IMPACT MX, and GREGG MX presets. GREGG MX's loudness is comparable to the LOUD presets; CRISP MX, EXTREME MX, and IMPACT MX are somewhat louder.

LOUD-HOT is very bright and present, with up-front vocals. Release time is medium.

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

LOUD-HOT+BASS is based on LOUD-HOT. 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-WIDE provides a large amount of stereo enhancement. "Wide" refers to the stereo enhancer setting, which is more extreme than the other LOUD presets.

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.

LOUD-BIG compromises between LOUD-HOT and LOUD-HOT+BASS. It uses a 12 dB/octave bass equalizer slope to achieve punchy bass that still has enough mid-bass boost to help smaller radios.

LOUD-FAT has dramatic punch on percussive material and a very fat-sounding low end, plus outstandingly effective distortion control. It avoids overt bass distortion despite the full bass sound. It is slightly quieter than the loudest of the "loud" preset family.

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 on-field sounds, so the preset does not pump this up as the NEWS-TALK preset would tend to do.

ROCK, ROCK MX: We have included all of the 8500-style ROCK presets and have made MX versions of ROCK-MEDIUM, ROCK-SOFT, and ROCK-SMOOTH, which are cleaner and punchier than their 8500-style counterparts. ROCK-SOFT MX and ROCK-SMOOTH MX are intentionally tuned 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 LESS-MORE to ensure clean speech. At high LESS-MORE levels the distortion control may be relaxed somewhat to increase bass punch.

ROCK-LIGHT has an open sound with little audible compression and less brightness than the first three presets. It is a compromise between ROCK-OPEN and ROCK-SOFT.

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.

Our first choice for Contemporary Hit Radio (CHR) is GREGG MX, which provides an excellent balance between loudness and distortion. However, ROCK-MEDIUM MX produces a cleaner, less processed sound and may therefore be appropriate if you want to encourage listeners to stay tuned in for very long times.

For Album-Oriented Rock (AOR) we again recommend GREGG MX, although you might prefer the more conservative ROCK-SOFT MX or ROCK-SMOOTH MX versions.

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.

SMOOTH JAZZ, SMOOTH JAZZ MX: These presets are designed for commercial stations playing smooth jazz. They are loud presets designed to prevent stridency with saxes and other horns. SMOOTH JAZZ is based on a custom 8500 preset that has been used successfully by a major-market smooth jazz station with very good ratings. The SMOOTH JAZZ MX preset exploits the 8600's ability to produce more transient impact and lower distortion. For either preset, if the loudness/distortion tradeoff is not to your taste use LESS-MORE to turn it down, producing lower loudness with less distortion.

URBAN: There are two URBAN (Rap) presets: HEAVY and LIGHT. These are 8500-style presets that have been retained in the 8600 mainly for historical reasons; GREGG MX, IMPACT MX, and EDGE MX are more contemporary sounding.

The URBAN presets are similar to ROCK-MEDIUM and ROCK-OPEN but with a different bass sound. They use the 3-pole (18 dB/octave) shape on the bass equalizer. URBAN-HEAVY is appropriate for Urban, Rap, Hip-Hop, Black, R&B, Dance and other similar formats. URBAN-LIGHT is appropriate for light R&B formats.

Equalizer Controls

Table 3-2 on page 3-31 summarizes the equalization controls available for the Five-Band structure.

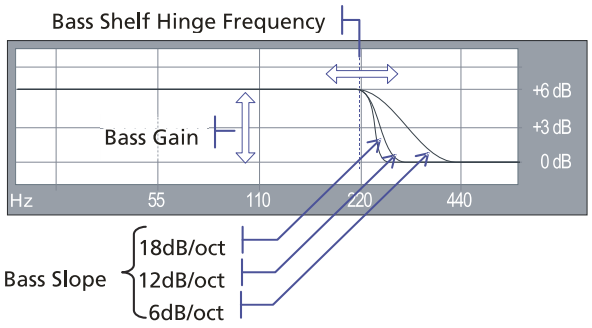
> These EQ controls are available in Basic Modify, Intermediate Modify and Advanced Modify screens. However, some of the control names have abbreviated names in the Advanced Modify screen, as noted in the table.

Except for BRILLIANCE and DJ BASS BOOST, these equalization controls are common to both the Two-Band and Five-Band structures.

> The equalizer is located between the AGC and multiband compressor sections of both structures.

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 LESS-MORE, Intermediate Modify, or Advanced Modify) before you create your User Preset.

In general, you should be conservative when equalizing modern, well-recorded program material. Except for Bass Shelf Gain, most of the factory presets use less than 3 dB of equalization.



Bass Shelf Controls, the 5-band structure's low bass equalization controls, are designed to add punch and slam to rock and urban music. They provide a parametric shelving equalizer with control over gain, hinge frequency, and slope (in dB/octave).

Bass Shelf Hinge Frequency sets the frequency where shelving starts to take effect.

Bass Gain sets the amount of bass boost (dB) at the top of the shelf.

Bass Slope sets the slope (dB/octave) of the transition between the top and bottom of the shelf.

Because the Five-Band structure often increases the brightness of program material, some bass boost is usually desirable to keep the sound

Equalizer Controls			
Group	**Basic / Full Modify Name**	**Advanced Name**	**Range**
Bass Shelf	BASS FREQ	Bass Frequency	80, 85, 90, 95, 100, 105, 110, 115, 120, 125, 130, 135, 140, 145, 150, 160, 170, 180, 190, 200, 210, 220, 230, 240, 250, 270, 290, 310, 330, 350, 380, 410, 440, 470, 500Hz
	BASS GAIN	Bass Gain	0 … 12 dB
	BASS SLOPE	Bass Slope	6,12,18 dB / Oct
Low	LF FREQ	Low Frequency	20 ... 500 Hz
	LF GAIN	Low Gain	–10.0 … +10.0 dB
	LF WIDT	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
Brilliance	BRILLNCE	Brilliance	0.0 … +6.0 dB
HF Enhancer	HF ENH	High Frequency Enhancer	0 … 15
DJ Bass (5B only)	DJ BASS	DJ Bass Boost	Off, 1… +10 dB
	Rumble Filter/30 Hz HPF	30 HZ HPF	Off/On
	Phase Rotator	Phase Rotate	Off/On
Phase Corrector /	Phase Corr	Phase Corrector	On…Off
Multipath Mitigator	Phase Corr Xover	Phase Corrector Crossover	62.5…796 Hz

Table 3-2: Equalization Controls

spectrally well balanced. Adjustment of bass equalization must be determined by individual taste and by the requirements of your format. Be sure to listen on a wide variety of radios—it is possible to create severe distortion on poor quality speakers by over-equalizing the bass. Be careful!

The moderate-slope (12 dB/octave) shelving boost achieves a bass boost that is more audible on smaller radios, but which can sound boomier on high-quality receivers. The steep-slope (18 dB/octave) shelving boost creates a solid, punchy bass from the better consumer radios with decent bass response. The 6 dB/octave shelving boost is like a conventional tone control and creates the most mid-bass boost, yielding a "warmer" sound. Because it affects the mid-bass frequency range, where the ear is more sensitive than it is to very low bass, the 6 dB/octave slope can create more apparent bass level at the cost of bass "punch."

There are no easy choices here; you must choose the characteristic you want by identifying your target audience and the receivers they are most likely to be using. Regardless of which curve you use, we recommend a +2 to +5 dB boost for most formats. Larger amounts of boost will increase the gain reduction in the lowest band of the multiband compressor, which may have the effect of reducing some frequencies. So be aware the large fixed bass boosts may have a different effect than you expect because of the way that they interact with the multiband compressor. (The GREGG presets use this effect purposely to create a dynamic cut in the mid-bass.)

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.

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

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

Low 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.

The LF parametric can be used in the mid-bass region (100-300 Hz) to add "warmth" and "mellowness" to the sound when boosting. When cutting, it can remove a "woody" or "boxy" sound. In our presets, we tend to use it very sparingly (in the order of 1 dB boost) to add a bit of extra bass warmth.

One formula for producing a very "big" bass sound on the air is to use a peaking boost at 100 Hz in combination with a Bass Shelf boost at 6 dB/octave.

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 8600'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 8600's curves accurately emulate an analog equalizer, even 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 Frequency determines the center frequency of the equalization, in Hertz. Range is 250-6000 Hz.

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 MB 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. Because the 3.7 kHz band compressor is partially coupled to the gain reduction in the 6.2 kHz band in most presets, tuning MID FREQ to 2-4 kHz and turning up the MID GAIN control will decrease energy in the 6.2 kHz band—you will be increasing the gain reduction in both the 3.7 kHz and 6.2 kHz bands. You may wish to compensate for this effect by turning up the BRILLIANCE control.

Use the mid frequency equalizer with caution. Excessive presence boost tends to be audibly strident and fatiguing. Moreover, the sound quality, although loud, can be very irritating. We suggest a maximum of 3 dB boost, although 10 dB is achievable. In some of our factory presets, we use 3 dB boost at 2.6 kHz to bring vocals more up-front.

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 Frequency 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 tape 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.

Brilliance controls the drive to Band 5 in the Five-band structure only. (This control is nonfunctional in the 2-Band structure.) The high frequency limiter and Band 5 clipper dynamically control these boosts, protecting the final clipper from excessive HF drive. We recommend a maximum of 4 dB of BRILLIANCE boost and most people will prefer substantially less.

DJ Bass (5-Band only) 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 activates 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 prefer 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 were 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 100 Hz, 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.

> This control is only available in the Five-Band structure.
>
> If the GATE THRESH (Gate Threshold) control is turned OFF, the DJ BASS boost setting is disabled.

HF Enhance 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.

Rumble Filter 30 Hz HPF determines if a 30 Hz 18 dB/octave highpass filter will be placed in-circuit before other processing. This control affects both the HD and FM processing chains regardless of the setting of the FM→HD CONTROL COUPLING control.

Phase Rotator determines if the phase rotator will be in-circuit. The purpose of the phase rotator is to make voice waveforms more symmetrical. This can substantially reduce distortion when they are peak limited by the 8600's back end processing. This control affects both the HD and FM processing chains regardless of the setting of the FM→HD CONTROL COUPLING control.

> In most cases, we recommend that the phase rotator be left active. However, because it can slightly reduce the clarity and definition of program material, you can defeat it if you are operating the 8600 conservatively and not attempting to achieve very high on-air loudness.
>
> You have somewhat more leeway than you do in older Orban FM processors like the 8200 because the 8600's new distortion control will work to help prevent audible speech distortion even when the phase rotator is switched out.

Multipath Mitigator / Phase Corrector See *"Multipath Mitigator" Left/Right Phase Skew Correction* on page 3-8.

Stereo Enhancer Controls		
Basic / Intermediate Name	**Advanced Name**	**Range**
Amount	Amount	0.0 ... 10.0
Enhancer	In / Out	Out / In
Ratio Limit	Ratio Lim	70 … 100%
Diffusion	Diffusion	Off, 0.3 ... 10.0
Style	Style	222 / Delay
Depth	Depth	0 … 10

Table 3-3: Stereo Enhancer Controls

Stereo Enhancer Controls

The stereo enhancer is common to the FM and HD processing chains. You You can operate it in one of two modes or "styles." The first emulates the Orban 222 analog stereo enhancer, while the second mode, called Delay, emulates a popular enhancer from another manufacturer that adds a delayed version of the L–R signal to the original L–R to create stereo enhancement. (See **Stereo Enhancement** on page 3-8 for more information.)

Both modes have gating that operates under two conditions.

- The two stereo channels are close to identical in magnitude and phase.

  In this case, the enhancer assumes that the program material is actually mono and thus suppresses enhancement to prevent the enhancement from exaggerating the undesired channel imbalance.

- The ratio of L–R / L+R of the enhanced signal tries to exceed the threshold set by the L-R / L+R Ratio Limit control.

  In this case, the enhancer prevents further enhancement in order to prevent excess L–R energy, which can increase multipath distortion.

The stereo enhancer has the following controls:

Amount sets the maximum spatial enhancement.

Enhancer In / Out bypasses the stereo enhancer. OUT is equivalent to setting the AMOUNT to 0.

L-R / L+R Ratio Limit sets the maximum amount of enhancement to limit the amount of multipath distortion that the stereo enhancement can add. However, if the original program material exceeds this limit with no enhancement, the enhancer will not reduce it.

Diffusion applies only to the DELAY enhancer. This control determines the amount of delayed L–R added to the original signal.

Style sets one of two stereo enhancer types: 222 or DELAY.

Depth sets the delay in the delay line. It applies only to the DELAY enhancer.

AGC Controls

The AGC is common to the two-band and Five-Band structures.

Five of the AGC controls are common to the Intermediate Modify and Advanced Modify screens, with additional AGC controls available in the Advance Modify screen, as noted in the following table.

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 Modify parameters to your own taste. Always start with LESS-MORE to get as close to your desired sound as possible. Then edit the Advanced Modify parameters using the Advanced Modify screen and save those edits to a User Preset.

AGC On/Off control activates or defeats the AGC.

This control is usually used to defeat the AGC when you want to create a preset with minimal processing (such as a CLASSICAL preset). The AGC is also ordinarily defeated if you are using an external AGC (like Orban's 6300). However, in this case it is better to defeat the AGC globally in the System Setup screen.

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.)

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

AGC Controls		
Intermediate Names	**Advanced Names**	**Range**
AGC On/off	AGC	Off / On
Drive	Drive	–10 ... 25 dB
Release	Release	0.5, 1.0, 1.5, 2 … 20 dB / S
Gate Thresh	Gate Thresh	Off, –44 ... –15 dB
Bass Coupling	Bass Coupl	Off, 12 … 0 dB
—	MaxDeltaGR	0 … 24 dB, Off
—	Window Size	–25 … 0 dB
—	Window Rel	0.5 … 20 dB
—	AGC Matrix	L / R, sum / dif
—	AGC Ratio	∞1, 4:1, 3:1, 2:1
—	Bass Thresh	–12.0 … 2.5 dB
—	Idle Gain dB	–10 … +10 dB
—	AGC Bass Attack	1 … 10
—	AGC Master Attack	0.2 .. 6
—	AGC Bass Release	1 … 10 dB / sec
—	Master Delta Thresh	–6 … +6 dB
—	Bass Delta Thresh	–6 … +6 dB
—	AGC Crossover	Allpass, LiNoDly, Linear

Table 3-4: AGC Controlsa

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.

Master AGC 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's MB RELEASE control to FAST, and you can trade the two off to produce different effects.

Unless it is purposely speeded-up (with the MB 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 Five-Band structure's controls.

AGC Gate Thresh (Threshold) control determines the lowest input level that will be recognized as program by OPTIMOD-FM; 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 8600. The first affects the *AGC* and the second affects the *multiband compressor*. Each 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 in-

troducing 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 MULTIBAND GATE THRESH (Gate Threshold) control is turned OFF, the DJ BASS control (in the Equalization screen) is disabled.

AGC Bass Coupling control clamps the amount of dynamic bass boost (in units of dB) that the AGC can provide. (In V1.x, the unit of measure was percent.)

The AGC processes audio in a master band for all audio above approximately 200 Hz and a bass band for audio below approximately 200 Hz. Starting with V2.0 software, 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 cannot decrease below 6 dB even if the gain reduction signal from the Bass compressor sidechain is lower. However, the audio path 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 dynamic bass boost.

Note that the operation of this control was changed in 8600 V2 software to work as explained above. You may have to tweak this control to achieve the same bass balance that you had previously with V1.x software.

Advanced AGC Controls

The following AGC controls are found only in the Advanced Modify screen.

AGC Max Delta GR determines the maximum gain difference permitted between the two channels of the AGC. Set it to "0" for perfect stereo coupling.

> This control works the same regardless of whether the AGC operates in left/right or sum / difference MATRIX modes, in both cases controlling the maximum gain difference between the "channels." Depending on the MATRIX mode setting, the "channels" will handle left and right signals or will handle sum and difference signals. When the AGC operates in sum / difference MATRIX mode, this control determines the maximum amount of width change in the stereo sound field.

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 WINDOW SIZE. The window has 2:1 asymmetry around the current AGC gain reduction. For example, if the 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 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 WINDOW SIZE is 3 dB.

Window Release (see WINDOW SIZE above.)

AGC Matrix allows you to operate the AGC in left/right mode or in sum / difference mode. Usually you will operate in left/right mode. However, sum / difference mode can give a type of stereo enhancement that is different from the enhancement modes offered in the 8600's built-in stereo enhancer. This will only work if you allow the two channels of the AGC to have different gains. To do this, set the AGC MAXDELTGR control greater than zero.

> It is unwise to set this control beyond 3 dB. Multipath distortion could increase because the amount of L–R energy builds up excessively. We prefer using the 8600's stereo enhancer because its built-in gating circuits prevent over-enhancement.

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.

Previous Orban AGCs had compression ratios very close to ∞:1, which produces the most consistent and uniform sound. However, the 8600 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.

Bass Thresh 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 BASS COUPLING control is moved towards 0 dB, the AGC BASS THRESH control affects the sound less and less.

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

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

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

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

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

Master Delta Threshold allows you to set the difference between the compression thresholds of the sum and difference channels. (This control is only useful when you set the AGC MATRIX to SUM / DIF.) By setting the threshold of the difference channel lower than the sum channel, you can have the AGC automatically produce more gain reduction in the difference channel. This will reduce the separation of material with an excessively wide stereo image (like old Beatles records). To make this work, you must set the MAX DELTA GR control away from zero. For example, to limit an excessively wide image while preventing more than 3 dB difference in gain between the sum and difference channels, set the MAX DELTA GR control to 3.0 and the MASTER DELTA THRESHOLD control to some positive number, depending on how much automatic width control you want the 8600 to perform.

Bass Delta Threshold works the same as MASTER DELTA THRESHOLD, but applies to the bass band. You will usually set it the same as MASTER DELTA THRESHOLD.

AGC Crossover allows you to choose ALLPASS, LINODLY, or LINEAR modes.

ALLPASS is a phase-rotating crossover like the one used in the 8200's two-band AGC. It introduces one pole of phase rotation at 200 Hz. The overall frequency response remains smooth as the two bands take different degrees of gain reduction—the response is a smooth shelf without extra peaks or dips around the crossover frequency. The two bands are down 3 dB at the crossover frequency.

> All Five-Band factory presets automatically use ALLPASS because of its smooth, shelving behavior and low delay. Its allpass characteristic complements the existing phase rotator that reduces voice distortion. Because the Five-Band structure uses phase-rotating crossovers in the five-band compressor / limiter, there is little or nothing to be gained by using a phase-linear crossover in the Five-Band structure's AGC.

LINODLY (Linear-Phase; no delay) is a phase-linear crossover whose upper band is derived by subtracting its lower band from the crossover's input. When the upper and lower bands have the same gain, their sum is perfectly flat with no phase rotation.

However, when the upper and lower bands have different gains, peaks and dips appear in the frequency response close to the crossover frequency.

> LINODLY is useful if you need a crossover with low delay and no phase distortion when flat. Its downside is the possibility of coloration when the gains of the two bands are widely disparate.

LINEAR is a phase-linear crossover whose upper band is derived by subtracting its lower band from the crossover's input, as passed through a delay equal to the group delay of the lowpass crossover filter. The overall frequency response remains smooth as the two bands take different degrees of gain reduction—the response is a smooth shelf without extra peaks or dips around the crossover frequency. The two bands are each down 6 dB at the crossover frequency. This crossover has constant delay even when the two bands have unequal gains.

> While LINEAR has the ideal combination of no phase distortion (even when non-flat) and smooth shelving behavior, it adds about 4 ms to the overall delay (compared to ALLPASS and LINODLY), so it is not a good choice if you need to drive talent headphones.

Peak Limiter Controls

There are two sets of peak limiter controls, one of which is active when an 8500-style preset is active and the other of which is active when an "MX" preset (which uses the 8600's new peak limiter technology) is active. For a given limiter algorithm (8500-style or "MX"), the peak limiter controls are common to the Two-Band and Five-Band structures, except as noted in the control descriptions on the following pages. In the Five-Band structure, the peak limiter is located after bands 1 and 2 are summed. In the Two-Band structure, the peak limiter is located after the Bass band.

Peak Limiter Controls Common to 8500-style and MX Presets

Except for COMP MODE and MULTIPLEX OFFSET, which are only available in the Advanced Modify, all of the peak limiter controls common to 8500-style and MX Presets appear in both the Intermediate Modify and Advanced Modify screens.

Monitor Drive sets the drive level into the FM peak limiter in the low-delay monitoring processor, which is a separate instance of the Ultra-Low Latency structure. Turning it up produces more FM peak limiter artifacts in the monitor output. Turning it

Peak limiter Controls Common to 8500-Style and MX Presets		
Intermediate Name	**Advanced Name**	**Range**
(Composite) Limit Drive	Comp Drive	Off, 0 ... 3 dB
—	PilotProtect	Off, On
—	Multiplex Offset	0 … –10 dB
—	Comp Mode	Hard, Half-Cos
---	Monitor Drive	–5. 0… 0.0 dB

Table 3-5: Peak Limiter Controls Common to 8500-Style and MX Presets

down makes the monitor sound cleaner.

> Because the ULL structure's peak limiter has lower performance than the normal-latency and MX structures, you will not be able to match the sound of the on-air signal exactly, but it should be close enough to facilitate good performances by on-air talent.

Composite Limit Mode determines if the composite limiter will operate as a hard clipper or as Orban's patented "Half-Cosine Interpolation" composite limiter (which is not a clipper).

> When operating as a hard clipper, the composite limiter will produce maximum brightness in the frequency range from 5 to 15 kHz because, unlike the 8600's left/right audio-domain peak limiters, it can produce square waves in this frequency range. The downside compared to Half-Cosine mode is that it can noticeably compromise stereo imaging.
>
> When operating in Half-Cosine mode, the composite limiter produces somewhat less brightness but does not compromise stereo imaging.
>
> In either mode, the baseband output is overshoot compensated and bandlimited to 53 kHz (*Figure 3-1*), unlike conventional composite clippers.

Composite Limit Drive sets the drive level, in dB, into the composite limiter.

This control has no effect on the 8600's left and right analog or digital outputs.

> The COMPOSITE LIMIT DRIVE control is set to "0 dB" for most factory presets. At this setting, it removes a few tenths of a dB of residual overshoot from the audio processing without affecting audio quality. We prefer to use the audio-domain overshoot compensation to do most of the work because it operates at a 256 kHz sample rate and is fully anti-aliased, whereas the composite limiter will inevitably introduce aliasing around 38 kHz upon demodulation in the receiver. This is because it introduces

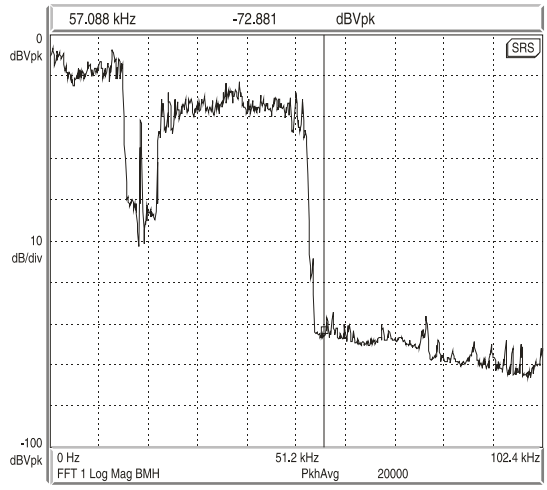


Figure 3-1: 0-100 kHz Baseband Spectrum (Loud-Hot preset)

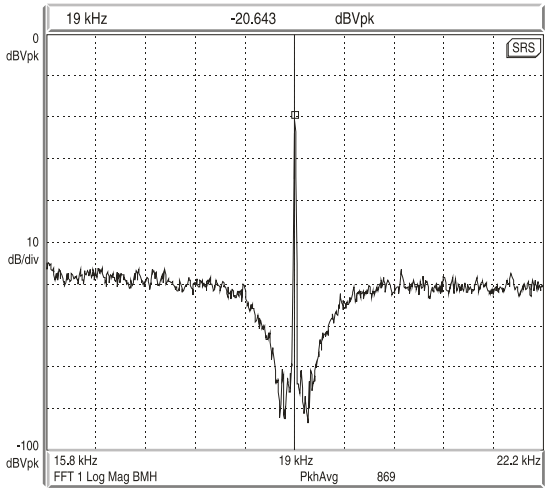


Figure 3-2: 19 kHz Pilot Notch Filter Spectrum (Loud-Hot preset; detail)

spectrum in the stereo subchannel area when it clips material in the 0 to 15 kHz area. The receiver will "see" this as stereo material and will demodulate it as if it were part of the stereo subchannel. Accordingly, harmonics of L+R material will be frequency-shifted upon demodulation and will no longer bear a harmonic relationship to the material that produced them.

If you want to use the composite limiter more heavily, one option is to trade off composite limiting against left/right domain overshoot compensation. To do this, back off the OSCOMP DR (Overshoot Compensation) control and increase the COMPOSITE LIMIT DRIVE control setting proportionately.

Because of the improved distortion control in the MX presets, a very small amount of composite limiting can significantly degrade the cleanliness of the audio. If your competitive situation permits, we recommend that you turn off the composite limiter, which will reduce the average modulation by less than 3% compared to leaving it set to 0 dB, which is where it is set in the MX factory presets.

Pilot Protect (Pilot Protection Filter) turns the 19 kHz notch filter on or off. It affects the composite output only.

The 8600's composite limiter always protects frequencies above 53 kHz. However, the 19 kHz notch filter can introduce substantial overshoot with certain program material when the composite limiter is driven hard. For example, if the composite limiter limits energy at 6.33 kHz, the 19 kHz notch filter will remove the third harmonic produced by the limiting. This will cause the output level to increase. For this reason, we offer the option to use the filter to provide excellent pilot protection at the cost of a slight potential overshoot, or to defeat the filter.

If the composite limiter is operated lightly (as it is in the factory presets) to remove a few percent residual overshoot, then the 19 kHz notch filter should have no observable effect on output overshoot and should remain in-circuit. In fact, there is a very good reason to tolerate a slight bit of overshoot for the sake of protecting the pilot, even if you are using the composite limiter more heavily. The loss of stereo coverage area (in fringe areas and in heavy multipath) due to pilot modulation will be much more obvious to the listeners than the loss of a few tenths of a dB of loudness.

If you are looking at the entire baseband on a spectrum analyzer with a 0-100 kHz sweep, you may be unable to see the effect of the pilot filter. This is because the filter protects the pilot ±250 Hz from 19 kHz and the spectrum analyzer will not resolve this when looking at the entire stereo baseband. To see the filter's effect, zoom the spectrum analyzer in to examine only the area immediately around 19 kHz. (See *Figure 3-2* on page 3-43).

We believe that ±250 Hz is a good compromise between excessive width (which would cause overshoot) and insufficient protection. ±250 Hz is sufficient to protect the phase-locked loops used in most stereo decoders. There is actually considerable protection ±1 kHz from the pilot, but the full 60 dB of protection is limited to ±250 Hz.

In all cases, the composite limiter protects the baseband to –80 dB from 55 to 100 kHz. This provides a 2 kHz guard band to protect the RDS/RBDS subcarrier at 57 kHz.

We have noted that the Belar "Wizard" FM stereo monitor indicates some pilot modulation even when the pilot protection filter is turned on. This is because the Belar demodulates a bandwidth wider than ±250 Hz around the pilot. A spectrum analyzer will reveal that, in fact, the pilot is protected by at least 60 dB in this 500 Hz wide area of the spectrum.

Multiplex Offset (Multiplex Power Offset) operates only when the ITU-412 multiplex power controller is activated (and is thus irrelevant to users in countries that do not enforce this standard). The control introduces a fixed loss before the FM analog peak limiting chain. If the MULTIPLEX POWER THRESHOLD control (in the INPUT/OUTPUT / UTILITIES screen) is set to 0, the MULTIPLEX POWER OFFSET control produces the same amount of loss (in dB) as this control's setting. Resetting the MULTIPLEX POWER THRESHOLD control away from 0 will change the loss. (For example, setting the MULTIPLEX POWER THRESHOLD control to +3 will cause the loss to decrease by 3 dB.)

The MULTIPLEX POWER THRESHOLD control can only introduce loss, never gain.

Regardless of the setting of the MULTIPLEX POWER THRESHOLD and MULTIPLEX POWER OFFSET controls, the resulting gain offset can never be larger than 0 dB.

The MULTIPLEX POWER OFFSET control's purpose is to reduce unnatural loudness variations that the multiplex power controller might produce. These can occur because the ITU specification does not call for psychoacoustic weighting. The Optimod does not force the multiplex power controller to dynamically produce all of the required gain reduction (which could vary widely, depending on the program material). Instead, the MULTIPLEX POWER OFFSET control produces most of the gain reduction. The gain reduction produced by the control is, of course, unchanging and cannot introduce audible artifacts.

The ideal dynamic gain reduction for the multiplex power controller is 2 to 3 dB with typical program material. However, the actual gain reduction will vary widely depending on whether the underlying processing preset is "loud" or "quiet." Therefore, the appropriate setting of the MULTIPLEX POWER OFFSET control depends strongly on what preset is in use. Accordingly, each preset has its own setting of the MULTIPLEX POWER OFFSET, which is a processing parameter like any other in a given preset. Hence, adjustments that affect the multiplex power controller appear in two independent places in the Optimod: The MULTIPLEX POWER THRESHOLD control is a *system setup* control, while the MULTIPLEX POWER OFFSET is *part of the on-air preset*.

Depending on the preset, the MULTIPLEX POWER OFFSET control's setting can vary from 0 dB (no effect) to as much as –16 dB. If you customize a preset in any way (including using LESS-MORE), you may wish to trim the MULTIPLEX POWER OFFSET for that preset so that the multiplex power controller produces 2-3 dB of indicated gain reduction with typical program material. This will achieve the maximum on-air loudness that complies with the ITU standard while minimizing the potential for unnatural and audibly disturbing loudness inconsistencies caused by the operation of the multiplex power controller.

For more information on the multiplex power controller, refer to *ITU-R Multiplex Power Controller* on page 3-81.

Bass Clip Mode sets the operation of the bass clipper to HARD, LL HARD, MEDIUM, or SOFT. LL HARD IS not available in MX presets because reducing the ~260 ms delay of these presets by 4 ms would have no practical advantage.

- HARD operates the clipper like the clipper in the 8200. It produces the most harmonic distortion.

  This can be useful if you want maximum bass punch, because this setting allows bass transients (like kick drums) to make square waves. The peak level of the fundamental component of a square wave is 2.1 dB *higher* than the peak level of the flat top in the square wave. Therefore, this allows you to get low bass that is actually higher than 100% modulation—the harmonics produced by the clipping work to hold down the peak level.

  The square waves produced by this clipper are filtered through a 6 dB/octave lowpass filter that is down 3 dB at 400 Hz. This greatly reduces the audibility of the higher clipper-generated harmonics. Nevertheless, the downside is that material with sustained bass (including speech) will sound substantially less clean than it will with the Medium or Soft settings.

  Note that the HARD CLIP SHAPE control determines how squared-off the clipped bass waveforms become. (See *Hard Clip Shape* on page 3-49.)

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

  - LLHARD automatically defeats the compressor lookahead. This action is functionally equivalent to setting the LOOKAHEAD control to OUT, except that it reduces input/output delay by 5 ms).

  - 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.

  To minimize speech distortion, the speech/music detector automatically switches the bass clipper to MEDIUM when speech is detected, provided that the Five-Band structure is active, LATENCY is HIGH, and the BASS CLIP

8500-Style Peak limiter Controls		
Intermediate Name	**Advanced Name**	**Range**
—	OSComp Dr	–2.0 … +2.0
Bass Clip Thresh	BassClpThre	–6.0 … 0.00
—	SpeechBCTh	–6.0 … 0.00
Bass Clip Mode	BassClpMode	Soft, Medium, Hard, LLHard
—	Hard Clip Shap	0 … 10
HF Clip Thr	HF Clipping	0 ... 6.0 dB

Table 3-6: 8500-Style Peak Limiter Controls

MODE is set to HARD. (See "Lookahead" on page 3-70 for more about the speech/music detector.) If the bass clipper is set to LLHARD, the speech/music detector will reset the clipper threshold to the setting specified by the SPEECHBCTHR control. The default setting is "0 dB," which results in very little bass clipper action during speech. This prevents audible speech distortion that this clipper might otherwise introduce.

Switching the BASSCLIPMODE to LLHARD (from any other mode) removes five milliseconds of delay from the signal path. Switching can cause audible clicks, pops, or thumps (due to waveform discontinuity) if it occurs during program material. 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.

The HARD CLIP SHAPE control (in Advanced Control) offers further control over the sound of the HARD and LLHARD modes. See page 3-49.

- MEDIUM uses more sophisticated signal processing than HARD to reduce distortion substantially.
- SOFT uses the most sophisticated look-ahead signal processing to reduce distortion further. Using SOFT adds an additional 18 ms of delay to the processing (so that the total is approximately 36 ms).

MEDIUM and SOFT are not available in Low Latency mode. The bass clipping is always HARD, but the HARD CLIP SHAPE control is still available to "soften" the clipping.

8500-Style Peak Limiter Controls

All of the 8500-style peak limiter controls are common to the Intermediate Modify and Advanced Modify screens except for OSCOMP DR and HARD CLIP SHAPE, which are only available in the Advanced Modify screen.

Bass Clip Thresh sets the clipping threshold of Orban's patented embedded bass clipper, which is used only in the FM processing chain and not in the HD chain. The clipper is embedded in the multiband crossover so that any distortion created by clipping is rolled off by part of the crossover filters. The threshold of this clipper is usually set between 2 dB and 5 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 do not smash against the back-end clipping system, causing overt intermodulation distortion between the bass and higher frequency program material.

Some 8600 users feel that the bass clipper unnecessarily reduces bass punch at its factory settings. To accommodate these users, the threshold of the bass clipper is user-adjustable. The range (with reference to the final clipper threshold) is 0 to –6 dB. As you raise the threshold of the clipper, you will get more bass but also more dis-

tortion 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.

HF Clipping determines the amount of protection provided by the 8600's high frequency multiband clipper. This control was first introduced in the 8200 and allowed users to trade off distortion against brightness. Because of subsequent improvements in the clipping system, this control is much less useful than it was in the 8200 and we recommend always setting it to "0."

Final Clip Drive control adjusts the level of the audio driving the back end clipping system that OPTIMOD-FM uses to control fast peaks. The FINAL CLIP DRIVE control primarily determines the loudness/distortion trade-off.

Turning up the FINAL CLIP DRIVE 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 is increased as well. Lower settings of the FINAL CLIP DRIVE control reduce loudness, of course, but result in a cleaner sound.

If the MULTIBAND 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 MB LIMIT THR (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 MB LIMIT THR 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 8600'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 DRIVE control over a very narrow range (typically –0.5 dB to –1.0 dB) and to determine almost all of the loudness/distortion trade-off by the setting of the MULTIBAND CLIPPING control.

The final clipper operates at 256 kHz sample rate and is fully anti-aliased.

Advanced 8500-Style Peak Limiter Controls

The following Peak limiter control is found only in the Advanced Modify screen.

OSComp Dr (Overshoot Compensation Drive) sets the drive into the overshoot compensator with reference to the final clip threshold, in units of dB. The normal setting is "0 dB."

The overshoot compensator can produce audible distortion on material with strong high frequency content (like bell trees) and this control lets you trade off this distortion against loudness. (Such material can cause strong overshoots, forcing the overshoot compensator to work hard to eliminate them.) We do not recommend operating this control above "0" because this would reduce the effectiveness of the distortion cancellation used in earlier processing. However, you can reduce it below "0" if you value the last bit of high frequency cleanliness over loudness.

The overshoot compensator works at 256 kHz sample rate and is fully anti-aliased.

Hard Clip Shape (Hard Clip Shape) allows you to change the knee of the input/output gain curve of the bass clipper when BASS CLIP MODE is set to HARD. It allows you to control the shape of the "knee"—the transition between no clipping and flat-topping. "0" provides the hardest knee, where the transition between linear operation and flat-topping occurs abruptly as the clipper's input level is changed. "10" is the softest knee, where the transition starts 6 dB below BASSCLIPTHRESH setting and occurs gradually. The factory default setting is "7.6." See Figure 3-3.

The MED and SOFT bass clipper characteristics use sophisticated look-ahead algorithms that produce lower distortion than HARD mode, regardless of where the HARDCLIPSHAPE control is set. However, operating in HARD mode with the HARDCLIPSHAPE control set beyond "0" may produce a tradeoff between punch and bass distortion that is more appropriate for pop music requiring substantial bass punch to make its musical point. In any event, the HARDCLIPSHAPE control adds a useful color to the 8600's sound palette—one that is not duplicated by the existing

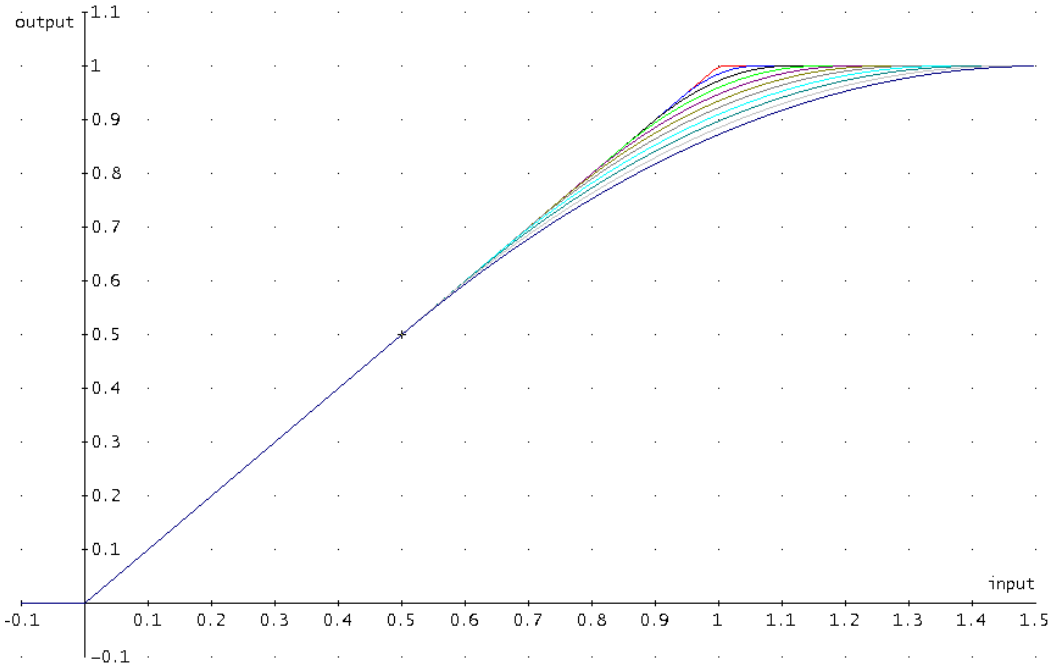


Figure 3-3: Bass Clipper Input/Output Transfer Curves as Bass Clip Shape Control is Varied from 0.0 (Hard) to 10.0 (Soft)

MED and SOFT bass clipper characteristics.

This control does nothing if the BASS CLIP MODE is set to MED or SOFT.

Speech Bass Clip Threshold ("SPEECHBCTHR")

(See *Bass Clip Mode* on page 3-46.)

8600 MX Technology Peak Limiter Controls

The MX peak limiter is designed to make FM analog broadcasts more competitive with the cleanliness, punch, and open high frequencies of the digital media against which FM analog transmissions now battle. To achieve this, it offers decreased audible distortion, more transient punch and a crisper, more open high frequency texture. Compared to the 8500-style peak limiter, the MX peak limiter typically provides 2.5 to 3 dB more power at high frequencies, which minimizes audible HF loss caused by preemphasis limiting. Drums and percussion cut through the mix. Highs are airy. See *Measuring the Improvements in Optimod-FM 8600's FM Peak Limiting Technology* starting on page 3-99 for a more detailed discussion.

The MX technology peak limiter consists of two main stages—first, a pre-limiter stage to control low frequency and high frequency peaks ahead of the main peak limiter and then the main peak limiter, which uses various proprietary techniques to control distortion, minimize transient loss, and minimize high frequency loss.

The controls affecting this section allow you to trade off loudness, distortion, and bass energy. Because the laws of physics apply to MX technology, increasing bass increases the likelihood that audible IM distortion between bass and other program elements will occur. Depending on your station's format, you may prefer to have a cleaner sound (with less bass) or a sound with more distortion but punchier bass. We offer controls, explained below, that allow you to make this tradeoff.

Peak Limiter Drive determines the level applied to the MX peak limiter and thus how much peak reduction the peak limiter must do. The MX peak limiter uses a variety of tactics to adapt its operation intelligently to the program material applied to it. Compared to the 8500-style peak limiter, the MX peak limiter produces a different set of artifacts when overdriven. Depending on how you set the other controls, these artifacts may include loss of bass punch, harsh clipping distortion, "soft" IM distortion, and/or excessive density that can cause fatigue. If you intend to make adjustments at the intermediate or advanced levels, it is wise to familiarize yourself with these artifacts by purposely overdriving the peak limiter via the PEAK LIMITER DRIVE CONTROL and listening to what happens with different types of program material.

This control's range is ±6 dB relative to an arbitrary 0 dB setting that provides a competitively loud sound without objectionable distortion on most program material. Settings above 0 dB are likely to produce objectionable distortion with some program material, while settings below 0 dB trade off lower loudness for a cleaner sound.

Bass PreLimit Mode allows you to operate the bass pre-limiter in MEDIUM mode, which produces some low-order harmonic distortion below 100 Hz, SOFT mode, which produces essentially no harmonic distortion but may reduce bass punch com-

pared to MEDIUM mode, or HARD mode, which produces the most distortion but also the most punch. The modes are very similar to the SOFT and MEDIUM bass clipper modes in the 8500-style presets. See BASS CLIP MODE on page 3-46.

Bass Clip Threshold sets the threshold of the bass clipper with respect to 100% modulation, which corresponds to 0 dB. Starting with v4.3.0 software, it works in SOFT and MEDIUM bass clipper modes in addition to HARD mode. See BASS CLIP MODE in page 3-46 for more detail.

Bass PreLimiting When the BASS PRELIMIT MODE is set to MEDIUM or SOFT, the MX bass pre-limiter can intelligently reduce the bass applied to the main peak limiter to reduce or prevent audible IM distortion. It does so when the pre-limiter's analysis of the program material indicates that this action is needed to prevent or minimize audible IM distortion between the bass (125 Hz and below) and other program elements in the main peak limiter. The BASS PRELIMITING control allows you to specify the maximum amount of bass reduction that can occur. Lower settings increase bass punch but do not protect against IM distortion as effectively as higher settings do.

There are two controls, one for Speech mode and one for Music mode, allowing you to have separate settings depending on whether the 8600 automatically detects speech or music input.

Bass Limiting Like the bass pre-limiter, the main peak limiter can automatically reduce bass when it detects potentially audible IM distortion. The BASS LIMITING control allows you to limit the amount of potential bass reduction at the expense of a possible increase in IM distortion.

Distortion Control This control determines the amount of audible distortion that the main peak limiter is permitted to create. Higher settings can increase loudness and punch at the expense of audible clipping distortion. Lower settings are cleaner but may reduce punch and loudness. We prefer it at 0, which is its cleanest setting. All MX factory presets use this setting.

The best way to familiarize yourself with the effects of this control is by listening extensively to different types of program material within your station's format while experimenting with different settings of the control. Because the MX peak limiter uses newly developed algorithms that differ from those used by past Orban proces-

MX Peak limiter Controls		
Intermediate Name	**Advanced Name**	**Range**
Peak Limiter Drive	Limiter Dr	–6.0…+6.0 dB
Bass Pre-Limiting	BassPreLimit	0 … 1
Bass Pre-Limit Mode	BassPreLimMode	Soft, Medium, Hard
Bass Clip Threshold	BassClpThre	–6…+6 dB
Bass Limiting	Bass Limting	0…–6
Distortion Control	Dist Cntrl	0 … 1
—	Multiband Clipping	–4.0…+5.0 dB
—	Speech BPLim	0 … 1

Table 3-7: MX Technology Peak Limiter Controls

sors, the loudness/distortion/brightness/punch tradeoffs are also different and it is worthwhile to take the time to get a feel for the MX limiter's capabilities.

Multiband Clipping This control is included to make it easier to transform existing 8500 user presets into MX presets. Like the MB CLIPPING control in the 8500-style presets, this control simultaneously and equally changes the thresholds of all the multiband compressor/limiters. (Changing the setting of each BX THRESHOLD control by an equal amount would produce the same effect as changing the MULTIBAND CLIPPING control by this amount.)

The MULTIBAND CLIPPING control is partially redundant to the PEAK LIMITER DRIVE control because when the multiband compressors are producing a considerable amount of gain reduction, both controls affect the level applied to the MX peak limiter. In this case, these controls are effectively cascaded and their effect is additive. For example, turning down the MULTIBAND CLIPPING control by 2 dB and turning down the PEAK LIMITER DRIVE by 3 dB reduces the drive level to the MX peak limiter by 5 dB when the multiband compressors are producing a considerable amount of gain reduction. However, when the multiband compressors are producing little or no gain reduction, the MB LIMIT THR control will have little or no effect. In our example, the drive level to the MX peak limiter might be reduced by as little as 3 dB.

Another important difference between the MULTIBAND CLIPPING and PEAK LIMITER DRIVE controls is that the MULTIBAND CLIPPING control setting is tracked by a hidden MULTIBAND CLIPPING control in the digital radio processing chain. Therefore, you can adjust the MULTIBAND CLIPPING control in the FM chain without needing to reset the HD LIMITER DRIVE control to ensure that the FM and HD channels are loudness-matched. There is no such "hidden" HD control corresponding to the PEAK LIMITER DRIVE control, so adjusting the PEAK LIMITER DRIVE control requires you to manually adjust the HD LIMITER DRIVE control by the same amount to preserve the loudness match in the two channels.

The Two-Band Structure

The Two-Band structure consists of a slow two-band gated AGC (Automatic Gain Control) for gain riding, followed by an equalization section, a gated two-band compressor, a high-frequency limiter, and a complex peak limiting system similar to the one used in the Five-Band structure. The Two-Band structure can be used with the MX-style peak limiter (to achieve highest performance) or with an 8500-style peak limiter (to achieve lower delay). Like the "Two-Band Purist" structure in Orban's OPTIMOD-FM 8200, the 8600's Two-Band Structure is phase-linear throughout to maximize sonic transparency.

The Two-Band structure has an open, easy-to-listen-to sound that is similar to the source material if the source material is of good quality. However, if the spectral balance between the bass and high frequency energy of the program material is incorrect, the Two-Band structure (when its Band Coupling 2 > 1 control is operated toward 0%) can gently correct it without introducing obvious coloration.

The Two-Band structure is mainly useful for classical or "fine arts" programming that demands high fidelity to the original program source.

Setting Up the Two-Band Structure for Classical Music

To set up the Two-Band structure, recall preset CLASSICAL-2B, CLASSIC-2B MX, CLASSICAL-2B+AGC, or CLASSIC-2B+AGC MX presets.

> These are the only two-band presets. To achieve the equivalent of an 8500 PROTECTION preset, use the CLASSICAL-2B preset. To achieve higher performance in protection applications, use the CLASSIC-2B MX preset instead.

Classical music is traditionally broadcast with a wide dynamic range. However, with many recordings and live performances, the dynamic range is so great that the quiet passages disappear into the noise on most car, portable, and table radios. Consequently, the listener either hears nothing, or must turn up the volume control to hear all the music. Then, when the music gets loud, the radio blasts and distorts, which makes listening unpleasant.

The Two-Band structure is well suited for classical formats during daytime hours when most people in the audience are likely to be listening in autos or to be using the station for background music. This audience is best served when the dynamic range of the program material is reduced by 10-15 dB so that quiet passages in the music never fade into inaudibility under these less-favorable listening conditions. OPTIMOD-FM controls the level of the music in ways that are, for all practical purposes, inaudible to the listener. Low-level passages are increased in level by up to 10 dB, while the dynamics of crescendos are maintained.

The CLASSICAL-2 BAND and CLASSIC-2B MX presets are two-band presets with the AGC turned off. They use considerable bass coupling to preserve the spectral balance of the input as well as possible. Their LESS-MORE controls primarily affect the amount of compression, not the maximum loudness.

CLASSICAL-2B+AGC AND CLASSIC-2B+AGC MX use the AGC set for 2:1 compression ratio. Because of the AGC, these presets affect more of the total dynamic range of the recording that does the CLASSICAL-2 BAND preset. However, the AGC provides extremely smooth and unobtrusive compression because of the gentle ratio and window gating. In this preset, the Two-Band compressor is used very lightly with a fast release time as a peak limiter. The AGC does almost all of the compression.

During the evening hours when the audience is more likely to listen critically, a classical station may wish to switch to a custom preset (derived from the CLASSIC-2B MX or CLASSICAL-2 BAND presets) that performs less gain reduction. You can create such a preset by modifying the CLASSICAL-2 BAND preset with the LESS-MORE control—turn it down to taste.

> There are also four five-band classical presets—two presets and their MX variants. The MX variants of the CLASSICAL presets achieve the lowest distortion and most open high frequencies. The CLASSICAL-5 BAND preset uses the five-band structure with AGC defeated. It uses substantial interband coupling to retain much of the frequency balance of the original source, but is capable of somewhat more "automatic equalization" than is CLASSICAL-2 BAND. It can therefore re-equalize older program material, but there is also more risk that it will cause coloration that might offend the classical purist.

CLASSICAL-5B+AGC, like its two-band counterpart, uses the AGC set for 2:1 compression ratio.

The five-band structure is not phase-linear, so the CLASSICAL-5 BAND preset is likely to have slightly less audible transparency than the CLASSICAL TWO-BAND structure.

Customizing the Settings

Each Two-Band Factory Preset has a LESS-MORE control (located in the Basic Modify screen) that adjusts on-air loudness. LESS-MORE simultaneously adjusts all of the processing controls to optimize the trade-offs between unwanted side effects as processing levels are decreased or increased.

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

The Two-Band Structure's Full Setup Controls

The tables below show a summary of the Two-Band controls in the dynamics section.

AGC, Equalizer, Stereo Enhancer, and Clipper controls are common to both Two-Band and Five-Band structures and are described in their own sections earlier in Section 3.

Some of the Two-Band controls are common to the Intermediate Modify and Advanced Modify screens. Additional Two-Band controls are available in the Advanced

Two-Band Controls		
Intermediate Name	**Advanced Name**	**Range**
Drive	2B Drive	–10 … 25 dB
Release	2B Release	0.5 … 20 dB / S
Gate Thresh	Gate Thresh	Off, –44 … –15 dB
Bass Coupling	Bass Couple	0 … 100 %
—	Lookahead	0 … 5 milliseconds
—	Master Compression Thresh	–15 … 0, Off
—	Bass Compression Thresh	–10.0 … 5.0 dB, Off
—	Master Attack	4 … 50, Off
—	Bass Attack	4 … 50, Off
—	2B Clipping	–4 … +5
—	HF Limiting (located on Clippers screen)	–4.0 … +2.0
—	HF Clip Thresh	–16.0 … 0.0, Off
—	2B Crossover	LiNoDly, Linear
—	MB Limit Thr	–3.0 +6.0, Off
—	Speech Thr	–4.0 ... +5.0 dB
—	Max Dist Ctrl	0 … 18 dB

Table 3-8: Two-Band Controls

Modify screen.

Except as noted, these controls affect both the analog and HD processing chains.

2B Drive control adjusts signal level going into the two-band compressor and therefore controls the density of output audio by determining the amount of gain reduction in the two-band compressor. The resulting sound texture can be open and transparent, solid and dense, or somewhere in between. The range is –10 to 25 dB.

Regardless of the release time setting, we feel that the optimum amount of gain reduction in the two-band compressor for popular music and talk formats is 10-15 dB. If less gain reduction is used, loudness can be lost. For classical formats, operating with 0-10 dB of gain reduction (with the gain riding AGC set to OFF) maintains a sense of dynamic range while still controlling levels effectively. Because OPTIMOD-FM's density gently increases between 0 and 10 dB of compression, 10 dB of compression sounds very natural, even on classical music.

2B Release control determines how fast the two-band compressor releases (and therefore how quickly loudness increases) when the level of the program material decreases. This release time only applies when the Two-Band Compressor is not gated by the silence gate.

The release time can be adjusted from 0.5 dB / second (slow) to 20 dB / second (fast). Settings toward 20 dB / second result in a more consistently loud output, while settings toward 0.5 dB / second allow a wider variation of dynamic range. Both the setting of the 2B RELEASE control and the dynamics and level of the program material determine the actual release time of the compressor.

The action of the 2B RELEASE control has been optimized for resolution and adjustability. However, its setting is critical to sound quality—listen carefully as you adjust it. There is a point beyond which increasing density (with faster settings of the 2B RELEASE control) will no longer yield more loudness and will simply degrade the punch and definition of the sound.

When the 2B RELEASE control is set between 8 and 1 dB / second (the slowest settings), the amount of gain reduction is surprisingly non-critical. Gating prevents noise from being brought up during short pauses and pumping does not occur at high levels of gain reduction. Therefore, the primary danger of using large amounts of gain reduction is that the level of quiet passages in input material with wide dynamic range may eventually be increased unnaturally. Accordingly, when you operate the 2B RELEASE control between 8 and 2 dB / second, it may be wise to defeat the gain-riding AGC and to permit the two-band compressor to perform all of the gain riding. This will prevent excessive reduction of dynamic range and will produce the most natural sound achievable from the Two-Band structures.

With faster 2B RELEASE control settings (above 8 dB / second), the sound will change substantially with the amount of gain reduction in the two-band compressor. This means that you should activate the gain-riding AGC to ensure that the two-band compressor is always being driven at the level that produces the desired amount of gain reduction. Decide based on listening tests how much gain reduction gives you

the density that you prefer while not creating a feeling of over-compression and fatigue.

> Release in the two-band compressor automatically becomes faster as more gain reduction is applied (up to about 10 dB). This makes the program progressively denser, creating a sense of increasing loudness although peaks are not actually increasing. If the gain-riding AGC is defeated (with the AGC on/off control), you can use this characteristic to preserve some feeling of dynamic range. Once 10 dB of gain reduction is exceeded, full loudness is achieved—no further increase in short-term density occurs as more gain reduction is applied. This avoids the unnatural, fatiguing sound often produced by processors at high gain reduction levels and makes OPTIMOD-FM remarkably resistant to operator gain-riding errors.

2B Gate Thresh (Threshold) control determines the lowest input level that OPTIMOD-FM recognizes as program; lower levels are considered to be noise or background sounds and will cause the AGC or two-band compressor to gate, effectively freezing gain to prevent noise breathing.

There are two independent gating circuits in the 8600. The first affects the AGC and the second affects the two-band compressor. Each has its own threshold control.

The two-band gain reduction will eventually recover to 0 dB and the AGC gain reduction will eventually recover to –10 dB even when their compressor gates are gated. However, recovery is slow enough to be imperceptible. This avoids OPTIMOD-FM's getting stuck with a large amount of gain reduction on a long, low-level musical passage immediately following a loud passage.

It is common to set the 2B GATE THRESH control to –40. Higher settings are primarily useful for radio drama, outside sports broadcasts, and other non-musical programming that contain ambiance, low-level crowd noise, and the like. Slightly higher settings may increase the musicality of the compression by slowing down recovery on moderate-level to low-level musical passages. When such passages cause the gate to cycle on and off, recovery time will be slowed down by the ratio of the "on-time" to the "off time." This effectively slows down the release time as the input gets progressively quieter, thus preserving musical values in material with wide dynamic range (classical music for example).

2B Bass Coupling control is used to set the balance between bass and the rest of the frequency spectrum.

The two-band compressor processes audio in a master band for all audio above approximately 200 Hz and a bass band for audio below approximately 200 Hz. The 2B BASS COUPLING control determines how closely the on-air balance of material below 200 Hz matches that of the program material above 200 Hz.

Settings toward 100% (wideband) make the output sound most like the input. Because setting the 2B BASS COUPLING control at 100% will sometimes cause bass loss, the most accurate frequency balance will often be obtained with this control set between 70% and 90%. The optimum setting depends on the amount of gain reduc-

tion applied. Adjust the 2B BASS COUPLING control until the band 1 and band 2 Gain Reduction meters track as closely as possible.

With the 2B RELEASE (Two-Band Release) control set to 2 dB / second, setting the 2B BASS COUPLING control toward 0% (independent) will produce a sound that is very open, natural, and non-fatiguing, even with large amounts of gain reduction. Such settings will provide a bass boost on some program material that lacks bass.

With fast release times, settings of the 2B BASS COUPLING toward 100% (wideband) do not sound good. Instead, set the 2B BASS COUPLING control toward 0% (independent). This combination of fast release and independent operation of the bands provides the maximum loudness and density on small radios achievable by the Two-Band structure. But such processing may fatigue listeners with high-quality receivers and also requires you to activate the AGC to control the average drive level into the two-band compressor, preventing uncontrolled build-up of program density. Instead of operating the Two-Band structure like this, you should usually choose a Five-Band preset instead.

Advanced Two-Band Controls

The following Two-Band controls are found only in the Advanced Modify screen.

2B Master Comp Threshold sets the level where gain reduction starts to occur in the Master (above 200 Hz) band of the Two-Band Compressor.

2B Bass Comp Threshold determines the compression threshold of the bass band in the Two-Band Compressor. It can be used to set the target spectral balance of the Two-Band Compressor.

> As the Two-Band Compressor BASS COUPLING control is moved towards "100%," the Two-Band Compressor BASS THRESH control affects the sound less and less.

2B Master Attack sets the attack time of the Two-Band Compressor master compressor (above 200 Hz).

2B Bass Attack sets the attack time of the Two-Band Compressor bass compressor (below 200 Hz).

Lookahead determines the lookahead time (in milliseconds) in the two-band compressor. 3.6 milliseconds give minimum overshoot for the factory preset attack time of 11.0. If you adjust LOOKAHEAD for less delay, you will get progressively more overshoot. This will cause more voice distortion but will create more transient impact on percussion because transients will hit the clippers harder instead of being controlled inaudibly by the lookahead compressor.

> This control does not affect the HD processing chain.

2B Crossover sets the structure of the 2-Band crossover to Linear or Linear with No Delay. See page 3-41 for more detail about these modes.

The relationship and interaction of the next four controls is complicated and is best appreciated by listening and experimenting:

2B Clipping is a compression threshold control that affects the bass and master bands equally. It sets the drive level to the high frequency limiting and multiband distortion-controlling processing that precedes the final clipping section. The distortion-controlling section uses a combination of distortion-cancelled clipping and look-ahead processing to anticipate and prevent excessive clipping distortion in the final clipper.

This control does not affect the HD processing chain.

MB Limit Threshold (not in MX presets) determines the threshold of the lookahead clipping distortion controller (measured in dB) with reference to the final clipper. In general, it should be set at "0" so that it matches the threshold of the final clipper. Setting it higher than "0" allows more punch (due to clipping) at the expense of higher clipping distortion, which may be particularly annoying on voice.

This control does not affect the HD processing chain.

2B HF Limiting (not in MX presets) sets the threshold of the high frequency limiter in the Two-Band structure. When this control is set lower, gain reduction does more high frequency limiting. When this control is set higher, distortion-cancelled clipping does more high frequency limiting. It controls the tradeoff between loss of high frequencies (due to high frequency limiting) and excessive distortion (due to clipping).

This control only affects the analog processing chain.

The HF Limiting control is in the clippers screen in the unit.

HF Clip Threshold (not in MX presets) sets the threshold of the multiband, distortion-cancelled clipper in the Two-Band structure's high frequency limiter. Higher numbers yield more brightness, but also cause more high frequency distortion.

This control does not affect the HD processing chain.

Max DistCntr (Maximum Distortion Control ; not in MX presets). See *Max Dist Ctrl;* page 3-68.

Speech Thresh (Speech Threshold) control. See *MB Speech Threshold* on page 3-71.

The Two-Band MX presets have the same peak limiting controls as the Five-Band MX presets. See *8600 MX Technology Peak Limiter Controls* starting on page 3-50.

The Five-Band Structure

The Five-Band structure consists of a stereo enhancer, a slow gain-riding two-band AGC, an equalization section, a five-band compressor, a dynamic single-ended noise reduction system, an output mixer (for the five bands), and a complex peak limiting system. The peak limiting system depends on the preset name. Presets with "MX" in their names invoke the 8600's MX limiter to achieve the highest performance avail-

able in the 8700. MX presets offer greater transient impact, less preemphasis-induced high frequency loss, and lower perceived distortion than their 8500-technology counterparts. Achieving this level of performance requires the MX presets to have much longer delays (typically 265 milliseconds) than the 8500-style presets. This is too long to permit live off-air monitoring into talent headphones. This is the main reason why the 8600 offers 8500-style presets too—these presets have much lower delays than do their MX counterparts.

Unlike the Two-Band structure, whose two-band compressor has a continuously variable release time, the release time of the Five-Band compressor is switchable to seven increments between slow and fast. Each setting makes a significant difference in sonic texture.

When the input is noisy, you can sometimes reduce the noise by activating the single-ended noise reduction system. Functionally, the single-ended noise reduction system combines a broadband downward expander with a program-dependent low-pass filter. This noise reduction can be valuable in reducing audible hiss, rumble, or ambient studio noise on-air.

Putting the Five-Band Structure on the Air

The Five-Band structure is very flexible, enabling you to fine-tune your on-air sound to complement your programming. There are numerous Factory Programming Presets whose names are the same as common programming formats. They offer considerable variety, with various combinations of release time, equalization, low frequency coupling, and high frequency coupling.

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 "Customizing the Settings," which follows.

Customizing the Settings

The LESS-MORE control can edit each of these presets to optimize the trade-off between loudness and distortion according to the needs of the format. To fine-tune them, presets can be further edited with Basic, Intermediate or Advanced Modify.

The controls in the Five-Band structure give you the flexibility to customize your station's sound. Nevertheless, as with any audio processing system, proper adjustment of these controls requires appropriately balancing the trade-offs between loudness, density, and audible distortion. The following provides the information you need to adjust the Five-Band structure controls to suit your format, taste, and competitive situation.

The Five-Band Structure's Full Setup Controls

The tables below show the Multiband and Band Mix controls in the dynamics section. The AGC, Equalizer, Stereo Enhancer, and Clipper controls are common to both the Two-Band and Five-Band structures and are discussed in their own sections above.

Except as noted, these controls affect both the analog and HD processing chains.

Multiband Controls		
Intermediate Name	**Advanced Name**	**Range**
Drive	MB Drive	0 ... 25
Release	(see MB ATTACK / REL screen)	Slow, Slow2, Med, Med2, MFast, MFast2, Fast
—	Speech MB Release	See above
Gate Threshold	Gate Thresh	Off, –44 ... –15 dB
Clipping	MB Clipping	–4.0 ... +5.0 dB
—	Speech Thr	–4.0 ... +5.0 dB
Down Expander	Down Expand	Off, –18.0 … 12.0 dB
B5 Down Expander	B5 Down Expand	Off, –18.0 … 12.0 dB
Band 2 > 1 Coupling	(see Band Mix screen)	0 ... 100 %
Band 2 > 3 Coupling	(see Band Mix screen)	0 ... 100 %
Band 3 > 2 Coupling	(see Band Mix screen)	0 … 100 %
Band 3 > 4 Coupling	(see Band Mix screen)	0 ... 100 %
—	Band 4 > 5 Coupling (see Band Mix screen)	0 ... 100 %
—	B1 Compression Threshold	–16.00 … 0.00, Off
—	B2 Compression Threshold	–16.00 … 0.00, Off
—	B3 Compression Threshold	–16.00 … 0.00, Off
—	B4 Compression Threshold	–16.00 … 0.00, Off
—	Speech B1 Comp Thresh	–16.00 … 0.00, Off
—	Speech B2 Comp Thresh	–16.00 … 0.00, Off
—	Speech B3 Comp Thresh	–16.00 … 0.00, Off
—	Speech B4 / 5 Comp Thresh	–16.00 … 0.00, Off
—	MB Limit Thr	–3.0 +6.0, Off
—	Max Dist Ctrl	0 … 18 dB
—	Speech Max Dist Ctrl	0 … 18 dB
—	HF Clip Thr	–16.00 … 0.0, Off
—	B5 Clip Thr (UL presets only)	–16.00 … 0.0, Off
—	B4 Clip Thr (UL presets only)	–16.00 … 0.0, Off
—	HF Limiter	Off, –23.8 ... 0.0 dB
—	DownExpStCpl	Off, On
—	B1/B2 Xover	100. 150, 200 Hz

Table 3-9: Multiband Controls

MB Drive control adjusts the signal level going into the multiband compressor and therefore determines the average amount of gain reduction in the multiband compressor. Range is 25 dB.

Adjust the MB 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 MB DRIVE interacts with the MB RELEASE setting. With slower release time settings, increasing the MB DRIVE setting 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 MB GATE THRESH control to "freeze" the gain when the input gets quiet and/or by activating the single-ended noise reduction.

When the multiband compressor's release time is set to its faster settings, the setting of the MB DRIVE control becomes 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 10 dB, 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 MB RELEASE instead.

MB Release; Speech MB Release control can be switched to any one of seven settings:

> The **Slow** 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. The SLOW and SLOW2 settings produce an unprocessed sound with a nice sense of dynamic range. With these settings, the Five-Band structure provides gentle automatic equalization to keep the frequency balance consistent from record to record (especially those recorded in different eras). 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. It 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 only 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.

> The Factory Presets for this sound are quite different from the other three release time settings. The amount of gain reduction in the multiband compressor is substantially lower (so that it operates more like a limiter than like a compressor) and the release time of the gain-riding AGC is speeded up (so that it provides compression and some increase of density). We made these trade-offs to prevent excessive build-up of density.

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 useful for stations that are ordinarily heard very softly in the background because it improves intelligibility under these quiet listening conditions. Stations that are usually 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.

The SPEECH MB RELEASE control overrides the MB RELEASE control when the 8600 automatically detects speech (page 3-70). You may wish to set the SPEECH MB RELEASE control faster for speech (to maximize smoothness and uniformity) and slower on music (to prevent excessive build-up of density).

MB Gate Thresh (Threshold) control determines the lowest input level that will be recognized as program by OPTIMOD-FM; 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 8600. The first affects the AGC and the second affects the multiband compressor. Each has its own threshold control.

V2.0 software sets the multiband compressor's gating threshold with respect to the signal driving the multiband compressor/limiter, which follows the AGC. Meanwhile, the AGC gating threshold remains set with respect to the AGC's input, as it was in V1.x software. Driving the MB gate detector after the AGC allows the AGC to do its work of slowly correcting the drive level to later processing, including the multiband gate.

When the AGC produces 10 dB of gain reduction, the calibration of the MB GATE

MB Attack / Release		
Intermediate Name	**Advanced Name**	**Range**
Release	MB Release	Slow, Slow2, Med, Med2, MFast, MFast2, Fast
—	Speech MB Release	Slow, Slow2, Med, Med2, MFast, MFast2, Fast
—	B2 Attack	4 … 50 ms, Off
—	B3 Attack	4 … 50 ms, Off
—	B4 Attack	4 … 50 ms, Off
—	Speech B1 Attack	4 … 50 ms, Off
—	Speech B2 Attack	4 … 50 ms, Off
—	Speech B3 Attack	4 … 50 ms, Off
—	Speech B4 / 5 Attack	4 … 50 ms, Off
—	B1 Limiter Attack	0 … 100%
—	B2 Limiter Attack	0 … 100%
—	B3 Limiter Attack	0 … 100%
—	B4/5 Limiter Attack	0 … 100%
—	Lookahead	In, Out, Auto
—	Speech Mode	Speech, Music, Auto
—	Delta Release 1	–6 … 6
—	Delta Release 2	–6 … 6
—	Delta Release 3	–6 … 6
—	Delta Release 4 / 5	–6 … 6

Table 3-10: MB Attack / Release Controls

control in V2.x software is the same as it was in V1.x software. When upgrading from V1.x to V2.x, it should thus be unnecessary to change the MB GATE setting in User Presets unless the AGC's nominal gain reduction is significantly different from 10 dB.

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.

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

MB Clipping sets the drive level to the multiband distortion controlling processing that precedes the final clipping section. The distortion-controlling section uses a combination of distortion-cancelled clipping and look-ahead processing to anticipate and prevent excessive clipping distortion in the final clipper.

Like any other dynamics processing, the distortion-controlling section can produce artifacts of its own when overdriven. These artifacts can include loss of definition, smeared high frequencies, a sound similar to excessive compression, and, when operated at extreme settings, audible intermodulation distortion. You can adjust the MB CLIPPING control to prevent such artifacts or to use them for coloration in "highly processed" formats.

MB Down Expander (Multiband 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. There are two controls: the MB DOWN EXPANDER control sets the expansion threshold in Bands 1-4, while the B5 DOWN EXPANDER DELTA THRESH control (first introduced as part of V2.0 software) allows you to fine-tune the Band 5 downward expander's threshold by adding or subtracting an offset from the setting of the MB DOWN EXPAND control. Activate the single-ended dynamic noise reduction by setting these controls 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 causing 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 and 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 sports.

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 MB DOWN EXPANDER control of the noise reduction system to complement the program material you are processing. The MB DOWN EXPANDER should be set higher when the input is noisy and lower when the input is relatively quiet. The best way to adjust the MB DOWN EXPANDER 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 MB DOWN EXPANDER 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 MB DOWN EXPANDER 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.

Band 5 is particularly critical for noise reduction because much of the Downward Expander's utility lies in hiss reduction. Hiss has most of its energy in band 5, while program material typically has less energy in this band, so the B5 DOWN EXPANDER DELTA THRESHOLD control's setting is critical to removing hiss while minimizing removal of desired program energy.

Starting in V2.0, the Downward Expander's dynamic frequency response is no longer constrained to being strictly lowpass—Band 5 is now uncoupled from the lower bands, so the band 5 downward expander can produce less gain reduction than other bands. This can help prevent loss of desired high frequency material in the program.

Band 3 > 4 Coupling control determines the extent to which the gain of band 4 and 5 (centered at 3.7 kHz) is determined by and follows the gain of band 3 (centered at 1 kHz). Set towards 100% (fully coupled) it reduces the amount of dynamic upper midrange boost, preventing unnatural upper midrange boost in light pop and instrumental formats.

Band 4 > 5 Coupling control determines the extent to which the gain of band 5 (6.2 kHz and above) is determined by and follows the gain of band 4.

> The sum of the high frequency limiter control signal and the output of the BAND 4 > 5 CPL control determines the gain reduction in band 5. The BAND 4 > 5 CPL control receives the independent left and right band 4

gain control signals; this feed is unaffected by the band 4 MAX DELTA G/R control. Range is 0 to 100% coupling.

Band 3 > 2 Coupling and Band 2 > 3 Coupling controls determine the extent to which the gains of bands 2 and 3 track each other.

When combined with the other coupling controls, these controls can adjust the multiband processing to be anything from fully independent operation to quasi-wideband processing.

Band 2 > 1 Coupling control determines the extent to which the gain of band 1 (below 100 Hz) is determined by and follows the gain of band 2 (centered at 400 Hz). 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 100 Hz (the "slam" region) to have maximum impact in modern rock, urban, dance, rap, and other music where bass punch is crucial. The default setting is 30%.

MB Band 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 ±6 dB range.

There are two sets of mixer controls, one for the analog processing chain and one for the HD processing chain.

The range of the band mix controls has been purposely limited because the only gain control element after these controls is the back-end clipping system (including the multiband clipper / distortion controller), which can produce considerable audible distortion if overdriven. The thresholds of the individual compressors have been carefully tuned 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 8600 more vulnerable to unexpected audible distortion with certain program material. Therefore, you should make large changes in EQ with the bass and parametric equalizers and the HF enhancer, because these are located *before* the compressors. The compressors will therefore protect the system from unusual overloads caused the chosen equalization. Use the multiband mix controls only for fine-tuning.

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.

On/Mute 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 8600 presets. The mute control for a given band affects *both the FM and HD processing* because it mutes the input of compressor in that band. This prevents the muted compressor's sidechain from producing a gain reduction signal that could couple into unmuted bands through the various band-coupling controls.

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.

Advanced Multiband and Band Mix Controls

The following Multiband and Band Mix controls are found only in the Advanced Modify screen.

Compression Threshold; Speech Compression Threshold controls set the compression threshold for music and speech in each band (following the 8600's automatic speech/music discriminator), in units of dB below the final clipper threshold. We recommend making 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 significantly affect the amount of distortion produced by the back-end clipping system.

You can use these controls to set independent frequency balances for music and speech (page 3-70).

Band Mix		
Intermediate Name	**Advanced Name**	**Range**
Band Mix 1	B1 Out Mix	–6.0 … +6.0
Band Mix 2	B2 Out Mix	–6.0 … +6.0
Band Mix 3	B3 Out Mix	–6.0 … +6.0
Band Mix 4	B4 Out Mix	–6.0 … +6.0
Band Mix 5	B5 Out Mix	–6.0 … +6.0
Mute Band 1	Band 1 On/Mute	On, Mute
Mute Band 2	Band 2 On/Mute	On, Mute
Mute Band 3	Band 3 On/Mute	On, Mute
Mute Band 4	Band 4 On/Mute	On, Mute
Mute Band 5	Band 5 On/Mute	On, Mute
—	B1MaxDeltGR	0 … 24 dB, Off
—	B2MaxDeltGR	0 … 24 dB, Off
—	B3MaxDeltGR	0 … 24 dB, Off
—	B4MaxDeltGR	0 … 24 dB, Off
—	B5MaxDeltGR	0 … 24 dB, Off
(see Multiband screen)	B2 > B1 Coupl	0 ... 100 %
(see Multiband screen)	B2 > B3 Coupl	0 ... 100 %
(see Multiband screen)	B3 > B2 Coupl	0 … 100 %
(see Multiband screen)	B3 > B4 Coupl	0 ... 100 %
(see Multiband screen)	B4 > B5 Coupl	0 ... 100 %

Table 3-11: MB Band Mix Controls

B1-B5 Attack (Time); Speech B1-B5 Attack 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 for music and speech respectively, following the 8600's automatic speech/music detector. 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 and/or pumping produced in the clipping system (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.

Because there are separate controls for music and speech (page 3-70), you can set attack times faster for speech (to minimize clipping distortion) and slower for music (to maximize punch and transient definition).

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

> The look-ahead delay times in bands 3, 4, and 5 automatically track the setting of the ATTACK time controls to minimize overshoot for any attack time setting.
>
> Only the five-band MX presets have Band 5 ATTACK controls for the FM analog processing. In the 8500-style presets, the band 4 and band 5 compressors share one compression sidechain, although their associated clippers and downward expanders operate independently.

MB Limit Thr (available in 8500-style presets only) sets the threshold of the clipping distortion controller with reference to the threshold of the final clipper, in dB.

> For best clipping distortion control, most effective setting for this control is "0 dB" for almost all program material. However, the loudest and most intense-sounding 8500-style presets rely on considerable clipping to achieve their loudness and brightness. For these presets, we found it necessary to set the MB LIMIT THR control substantially higher than "0" to permit more clipping depth. In some cases, this results in substantially objectionable distortion artifacts with isolated program material. However, this is the price to be paid for this extreme level of on-air loudness.
>
> For the NEWS-TALK and SPORTS presets, we set the MB LIMIT THR control slightly below "0." This ensures the cleanest possible speech quality at the cost of highest loudness. If you want higher loudness in these presets, you can edit them to increase the setting of the MB LIMIT THR control.
>
> If you set this control too low, (and/or set the MB CLIPPING control too high) the first artifact that you are likely to notice is intermodulation distortion between vocals and bass. Be aware of this possibility when you are adjusting this control, because the effect sometimes becomes clearer once you are accustomed to listening for it. Headphone listening will usually increase the audibility of this artifact.
>
> This control does not affect the HD processing chain.

Max Dist Ctrl; Speech Max Dist Ctrl (available in 8500-style presets only) limit the maximum amount of final clipper drive reduction (in dB) that the 8600's clipping distortion controller can apply in Music and Speech modes respectively, preventing

over-control of transient material by the distortion controller. Instead, the final clipper is permitted to control some of the transient material (to increase "punch"), even though, technically, such clipping introduces "distortion." A setting of 4 to 5 dB works best in most cases. Factory default is 5 dB for virtually all presets.

This control does not affect the HD processing chain.

HF Clip Threshold (available in 8500-style presets only) sets the threshold of the multiband clipper in band 5 with reference to the final clipper threshold, in dB. This clipper helps prevent distortion in the final clipper when the input program material contains excessive energy above 6 kHz.

The Band 5 multiband clipper operates at 256 kHz and is fully anti-aliased.

This control does not affect the HD processing chain.

B5 Clip Threshold (UL presets only) has the same functionality as HF CLIP THRESHOLD (above).

B4 Clip Threshold (UL presets only) sets the threshold of the multiband clipper in band 4 with reference to the final clipper threshold, in dB.

High Frequency Limiter sets the amount of additional gain reduction occurring in band 5 when high frequency energy would otherwise cause excessive distortion in the final clipper. It uses an analysis of the activity in the final clipper to make this determination and works in close cooperation with the band-5 multiband clipper. Functionally, this control is a mix control that adds a HF limiter gain reduction signal to the band 4 gain reduction signal to determine the total gain reduction in band 5. Higher settings produce more HF limiting.

This control does not affect the HD processing chain.

MB DownExpStCpl ("Multiband Downward Expander Stereo Coupling") determines whether the multiband downward expander stereo coupler is on or off.

B1 / B2 Xover (Band 1 to Band 2 Crossover Frequency) sets the crossover frequency between bands 1 and 2 to either 100 Hz, 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. The information found in *Getting the Bass Sound You Want* (starting on page 3-86) is also useful.

Band 1-5 MaxDeltGR controls set the maximum permitted gain difference between the left and right channels for each band in the multiband limiter. The 8600 uses a full dual-mono architecture, so the channels can be operated anywhere from fully coupled to independent. We recommend operating the bands fully coupled (BAND 1-4 MAXDELTGR = 0) for best stereo image stability. However, audio processing experts may want to experiment with lesser amounts of coupling to achieve a wider, "fatter" stereo image at the cost of some image instability.

Limiter Attack controls allow you to set the limiter attack anywhere from 0 to 100% of normal in the Five-Band compressor / limiters. Because the limiter and compressor characteristics interact, you will usually get best audible results when you set these

controls in the range of 70% to 100%. Below 70%, you will probably hear pumping because the compressor function is trying to create some of the gain reduction that the faster limiting function would have otherwise achieved. If you hear pumping in a band and you still wish to adjust the limiter attack to a low setting, you can sometimes ameliorate or eliminate the pumping by slowing down the compressor attack time in that band.

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 MB RELEASE setting and the actual release time you achieve in a given band. For example, if you set the MB RELEASE control to medium-fast and the BAND 3 DELTA RELEASE control to –2, then the band 3 release time will be the same as if you had set the MB RELEASE control to medium and set the BAND 3 DELTA RELEASE 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 MB 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 MB RELEASE control), the band in question will instead set its release time at the appropriate end-stop.

Lookahead activates or defeats the look-ahead functionality in the multiband compressor / limiter. Defeating look-ahead improves transient impact at the expense of distortion, particularly on speech. To mitigate this tradeoff, a selectable "auto" mode turns look-ahead on for speech material and off for music, using an automatic speech/music detector. Switching is seamless and click-free because we change the delay in the compressor control sidechains; this is not a way to reduce the 8600's throughput delay.

Choices are LOOKAHEAD IN, OUT, and AUTO.

> Speech is detected if (1) the input is mono, and (2) there are syllabic pauses at least once every 1.5 seconds. Speech with a stereo music background will usually be detected as "music," or the detector may switch back and forth randomly if the stereo content is right at the stereo / mono detector's threshold. Mono music with a "speech-like" envelope may be incorrectly detected as "speech." Music incorrectly detected as "speech" will exhibit a slight loss of loudness and punch, but misdetection will never cause objectionable distortion on music.
>
> Speech that is not located in the center of the stereo sound field will always be detected as "music" because the detector always identifies stereo material as "music." This can increase clipping distortion on such speech.
>
> If the BASS CLIP MODE is set to HARD, the speech/music detector will automatically set it to MEDIUM when speech is detected and HARD otherwise (unless LATENCY is LOW, in which case MEDIUM bass clipping is unavailable and bass clipping will stay HARD).
>
> Speech always sounds cleaner with MEDIUM bass clipping and the increased bass "punch" supplied by HARD is irrelevant to speech.
>
> This control does not affect the HD processing chain.

MB Speech Threshold ("SPEECH THR," located in the Speech page of ADVANCED CONTROL) lets you set the increment (in dB) by which the setting of the MB LIMIT THR control is reduced when speech is detected (see *Lookahead*, above). This control is particularly useful in minimizing speech distortion when you use the LLHARD bass clipper—it allows the main clipping distortion controller to work harder on speech while preserving punch in music.

This control does not affect the HD processing chain.

Speech Detect ("SPEECH DETECT," located in the Speech page of ADVANCED CONTROL) allows you to override the automatic speech/music detector, forcing the processing into SPEECH or MUSIC mode. The three available modes are SPEECH, MUSIC, and AUTO.

To Override the Speech/Music Detector

It is possible to force your Optimod into "speech" or "music" mode via its GPI and API. One can achieve this functionality by creating a "speech" preset and a "music" preset and programming the GPI to recall these presets as desired. To do so:

A) Create a preset with the desired settings for the music-mode and speech-mode controls. Set its SPEECH DETECT control to MUSIC.

B) Save this preset as a User Preset.

C) Set the SPEECH DETECT control to SPEECH.

D) Save the resulting preset as a second User Preset/

You may now program two GPI inputs to recall these two presets as desired. Recall the first preset to force "music" mode and recall the second preset to force "speech" mode. Of course, you can use any other means of recalling these presets including Automation, your Optimod's API, and PC Remote.

About the 8600's HD / Digital Radio Processing

Except for the fact that models 8600HD and 8600FM both offer an analog channel diversity delay, this section applies only to 8600 units, not 8600FM units. Model 8600FM is the same as model 8600 except that model 8600FM does not provide digital radio processing. Model 8600FM can be upgraded to an 8600 in the field by installing the plug-in CPU board contained in the 8600UPG/HD upgrade kit, which can be purchased from your Orban dealer.

The 8600 HD ("HD Radio") output is designed to feed streaming, netcasting, and digital radio channels, which can be DAB, DAB+, DRM, or the iBiquity® HD Radio™ system (formerly known as "IBOC"—"In-Band On-Channel") approved for use in several countries including the United States .

The equalizer and five-band compressor/limiter in the HD processing chain have their own sets of user-adjustable audio controls that are independent of the controls in the FM analog transmission chain's equalizer and five-band compressor/limiter. This allows you to optimize the HD processing for the higher fidelity sound provided by the digital channel while giving you the flexibility to process the analog channel as you wish. The stereo enhancer, phase rotator, and 30 Hz highpass filter are com-

mon to the two processing chains and their controls, which affect both chains, appear only in the FM adjustment screens.

To facilitate matching the FM analog and HD signals to minimize analog/digital crossfades in HD receivers, you can force the HD channel controls to track the FM channel controls by setting the FM→HD CONTROL COUPLING control to FM→HD. When you have finished tuning a preset to optimize the sound of the analog channel, you can then set the FM→HD CONTROL COUPLING control to INDEPENDENT and tweak the HD processing as you wish. This control is located in the HD LIMITING page of ADVANCED CONTROL. You can also toggle this control via two buttons on the button bar in 8600 PC Remote. When FM→HD CONTROL COUPLING is INDEPENDENT, a third button toggles PC Remote focus between corresponding FM and HD tabs. This allows you to compare your FM and HD settings easily.

Setting the FM→HD CONTROL COUPLING control to INDEPENDENT defeats the LESS-MORE control, so if you want to use LESS-MORE, adjust it before you start to make independent adjustments to the FM and HD chains.

If a given preset has certain HD and FM controls set differently, setting the FM>HD CONTROL COUPLING to COUPLED will immediately cause the HD controls to take the same settings as their FM analog counterparts. If you again set FM>HD CONTROL COUPLING to INDEPENDENT, this will restore any edits you made to the HD controls before you set the FM>HD CONTROL COUPLING back to COUPLED.

To control excessive brightness in the HD processing when operating the HD and FM processing independently:

- Use little or no high frequency boost in the HD equalization section.
- Set Band 4>5 coupling to 100%.
- Set the band 5 compression threshold to match the codec that the 8600's HD output is driving. 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 Band 4 compression threshold.

An advanced-design look-ahead limiter controls the peak level of the HD output. The look-ahead limiter (which receives the output of the HD multiband compressor/limiter) is optimized to make the most of the limited bit-rate codec used in the HD Radio system's digital channel. By eschewing any clipping, the HD output prevents the codec from wasting precious bits encoding clipping distortion products, instead allowing the codec to use its entire bit budget to encode the desired program material.

The 8600HD's look-ahead limiter implements "true peak" control by oversampling the HD peak limiter's sidechain at 256 kHz. This allows the 8600 to prevent clipping in a playback device's analog signal path by predicting and controlling the analog

peak level following the playback device's reconstruction filter to an accuracy of better than 0.5 dB. For typical program material, accuracy is 0.2 dB

Thanks to true peak control, sample rate conversion, unless it removes high frequency program energy or introduces group delay distortion, cannot cause sample peaks to increase more than 0.5 dB. For example, sample rate con-version from 48 kHz to 44.1 kHz is highly unlikely to cause sample peak clipping in the 44.1 kHz audio data.

The look-ahead limiter includes a parametric high frequency shelving equalizer that can be placed either before or after gain reduction. You can use it to equalize texture disparities between the FM and HD channels and to reduce codec artifacts at high frequencies. You can also use the HD BAND MIX controls and/or HD compression threshold to achieve this.

The HD output is designed to feed digital channels without preemphasis, which include almost all such channels. The only high-quality digital channels using preemphasis of which we are aware are NICAM channels (which use J.17 preemphasis) and some older CDs (which use EIAJ—50µs/15µs shelving preemphasis). If you use the HD output to feed a digital channel with preemphasis, you must allow extra headroom to compensate for the unpredictable peak level changes that the preemphasis induces.

If the HD output is driving a channel without preemphasis, it will control peak levels with an uncertainty of approximately 1 dB. However, you may want to allow headroom to compensate for data reduction-induced peak overshoots at the receiver, which might otherwise cause clipping. In our experience, 2 dB is typically adequate.

Delay Difference between HD and FM Outputs

In order to make the receiver analog/digital crossfade free from comb filtering, the time delays in the HD Radio's FM and HD channels must have a fixed and predictable offset, correctly implementing the HD Radio receiver's "time diversity" processing.

This "diversity delay" is part of the HD Radio system specification. The availability of the delay in the 8600 eliminates the need to use the delay line built into the HD Radio exciter. In turn, this allows you to connect any output receiving the analog-FM-processed signal to the input of the analog FM exciter. This includes the 8600 composite output, allowing you to use the 8600's stereo encoder and composite limiter during HD Radio broadcasts.

To avoid introducing artifacts in the analog FM path, its delay is held constant and HD delay is varied if the change does not exceed ±1.3 seconds. Larger adjustments change the analog FM delay and re-center the HD delay within the ±1.3 second window. The ±1.3 second window is sufficient for use with hardware that constantly measures the FM/HD delay off-air and automatically adjusts the Optimod's delay to compensate for delay drift in the transmission path. When no output emits the FM+DELAY source and the delay in the composite output is OFF, the extra 1.3 second delay is removed from both the FM and HD paths.

The MX presets have an input-to-output delay of approximately 265 ms, while the 8500-style presets typically have delays below 20 ms. To ensure a smooth transition

between presets of different types (like an 8500-style news preset and a music-oriented MX preset), the following must be true:

- The delay from the 8600's input to its FM analog processed output must not change when a preset is recalled (to prevent a audio segment from being repeated or dropped due to different time delays between presets). This criterion is also relevant to FM analog transmissions without an accompanying HD channel and therefore applies to the model 8600FM processor, which has no HD processing.
- The delay difference between the HD-processed output and the analog FM-processed output must not change. This allows HD radio receivers to crossfade smoothly between the digital and analog channels.

The 8600 will automatically obey these requirements if you do the following:

- Set the 8600's DIVERSITY DELAY TRIM control to 0.37 seconds or longer.
- If you are using the analog left/right or an AES3 digital output to drive the FM analog side of your transmission plant, assign that output's OUT SOURCE control to FM+DELAY.
- If you are using a composite output to drive the FM analog side of your transmission plant, set the stereo encoder's DIVERSITY DELAY control to IN. This control is located in the COMPOSITE screen.

If you need the shortest possible input-to-output time delay (for example, if you are driving talent headphones from the off-air feed), defeat the diversity delay on all outputs (including composite) by setting their OUT SOURCE to FM or DIVERSITY DELAY to OUT as appropriate for the output you are using. In this case, the delay difference between the analog and digital radio outputs will not stay constant when you recall different presets. This is not important for these applications because one gives up proper diversity delay (around eight seconds) for the sake of driving headphones or minimizing the delay in live sports events, which are the two typical applications where minimum delay is required. When using this setup, be aware that analog/digital crossfades in HD Radio receivers will sound very bad. Note too that switching between an MX and non-MX preset will cause about 345 ms of audio to be repeated or dropped.

HD I/O Setup Controls

Input/Output > HD Digital Radio screen:

Meter Sel determines if the Main Meter screen will display the left and right output levels of the FM processing (DISABLED) or the left and right gain reductions of the digital-channel look-ahead limiter (ENABLED).

> The HD look-ahead limiter is not stereo-coupled. This prevents limiting on one channel from causing audible modulation effects on the other channel.

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

Diversity Delay ("In/Out") determines if the composite output receives audio delayed with respect to the digital radio output. This control applies to both composite outputs—you cannot delay one output without delaying the other. This control is in the COMPOSITE sceen.

You can turn the delay on or off for the various outputs via the 8600's API (page 2-53), automation (page 2-42), and GPI remote interface (page 2-60).

Out Source allows you to determine if a given output (analog left/right, AES3 #1, AES3 #2) emits the FM-processed signal (FM), the FM-processed signal with diversity delay (FM+DELAY), the low-delay monitor signal (MONITOR; for driving talent headphones) or the HD-processed signal (HD). This versatile control allows you to selectively apply diversity delay to each output as desired.

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

When this field is highlighted, press the *Enter* button to toggle between coarse and fine adjustment. In PC Remote, drag the slider of the control with the mouse to set the delay coarsely, use the Page Up and Page Down keys for intermediate increments, and use the mouse wheel to fine-tune the delay in one-sample adjustment increments.

Section Label	Control Name	Values
Input/Output:Output1	Phone Src	HD / Monitor / FM
	Out Source (analog L/R)	HD / Monitor / FM / FM+Delay
Input/Output:Output2	Out Source (AES 1)	HD / Monitor / FM / FM+Delay
	Out Source (AES 2)	HD / Monitor / FM / FM+Delay
Input/Output:	Out Level	0… –20 dBFS; 0.1 dB steps
AES Output 1 and	Samp Rate	32, 44.1, 48, 88.2, 96 kHz
AES Output 2	HD Bandwidth	15, 16, 17, 18, 19, 20 kHz
	Word Leng	14 / 16 / 18 / 20 / 24 bits
	Dither	In / Out
	Sync	Internal / Sync In
	Format	AES3 / SPDIF
	Out Level	0… –20 dBFS; 0.1 dB steps
Input/Output: HD Digital Radio	Meter Select	FMOutLevel / HD GR
	HD HF Shelf EQ	Pre / Post
	Stereo/ Mono	Stereo / MonoL / MonoR / MonoSum
	Diversity Delay Trim	0.265 to 13.0 seconds
	(HD) Polarity	Positive / Negative
BS.1770 Safety Limiter	BS.1770 Loudness Control Threshold	–31 to –11 LKFS/LUFS
	BS.1770 Safety Limiter	On / Off

Table 3-12: HD I/O Setup Controls

Maximum available delay is approximately 13 seconds.

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

You can adjust the diversity delay time via the 8600's API. See page 2-54.

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

> The 8600 does not set the AES3 stereo / mono status bits to reflect the setting of this control. The AES3 status bits appearing at the HD output are always set "stereo" even when the two audio channels carry identical mono signals.

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

> The user should carefully test the codec in use to ascertain if lowering the bandwidth to 15 kHz improves subjective quality. This can occur because the codec then uses all of its bits to encode information in the most subjectively important part of the ear's bandwidth. If the codec employs Spectral Band Replication® technology (as does the iBiquity HD Codec), then you can set the bandwidth at 20 kHz without quality penalty, although the difference between 15 and 20 kHz is unlikely to be audible following the encode / decode cycle.

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

This control is best adjusted by observing an HD radio. Using a variable RF attenuator, vary the RF level feeding the radio to force crossfades. If you have correctly adjusted the 8600's ANALOG CHANNEL DELAY to time-align the analog and digital channels, the incorrect setting of the HD POLARITY control should be clearly audible as a momentary loudness decrease during the crossfades.

The HD POLARITY control can make it easier to match the analog and HD channel delay in an HD Radio installation. Temporarily set this control to cause a null, sum the HD and FM channels, and adjust the HD DELAY control to achieve the deepest null. This works best if the HD and FM analog loudnesses are matched; use the HD LIMIT DR control to do this.

If your facility has two FM analog exciters. one of which inverts polarity and one of which does not, you can use the 8600's FM POLARITY control to compensate when switching between transmitters. (See step 13 on page 2-37.) This function can also be controlled via the 8600's clock-based automation (see page 3- 42) and by its API (see step 13 on page 2-54).

Digital Output

You will normally use the AES2 digital output to drive the digital transmitter. However, you can use any output (ANALOG L/R, AES1, and AES2). The digital outputs have the following controls, located in the INPUT / OUTPUT screen.

Out Level ("Digital Output 1" and "Digital Output 2") sets the digital-channel output level with respect to digital full scale. It is normally set at 0 dBFS, which uses all the headroom available in the HD transmission channel. To match the loudness of the analog and digital channels at the receiver, use the LIMITER DRIVE control. This minimizes the amount of peak limiting necessary to match the loudness of the analog and digital channels at the receiver.

When the BS.1770 Safety Limiter is active, it changes the way the OUTPUT LEVEL control operates. See step 10 on page 2-32.

Samp Rate ("Sample Rate") sets the output sample rate of the digital-channel output to 32, 44.1, 48, 88.2, or 96 kHz.

> The 8600's fundamental sample rate is always 64 kHz, but the internal sample rate converter sets the rate at the 8600's digital output. This adjustment allows you to ensure compatibility with downstream equipment requiring a fixed sample rate.
>
> A 32 kHz sample rate cannot represent frequencies higher than approximately 15 kHz. Therefore, setting the sample rate to 32 kHz automatically forces the bandwidth to 15 kHz, regardless of the setting of the HD BANDWIDTH control.
>
> Using a 32 kHz sample rate to convey the FM-processed signal will increase peak overshoots because it removes the output spectrum between 14.7 kHz and 17 kHz, some of which is needed to control overshoots. Using an STL with a 32 kHz sample rate will cause overshoots for the same reason.

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

> The largest valid word length in the 8600 is 24 bits. The 8600 can also truncate its output word length to 20, 18, 16, or 14 bits. The 8600 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 8600 by turning this control on.

Sync determines if the sample rate appearing at the digital-channel output is synced to the 8600's internal clock, to an AES3 signal appearing at the 8600's digital input, or to an AES11 signal appearing at the 8600'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 rate present at the 8600'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 8600'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 8600 Input or SYNC IN, set SR SYNC to INTERNAL and select the desired output sample rate.

Format determines if the digital-channel output channel status bits follow 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.

BS.1770 Safety Limiter ON/OFF activates or defeats the BS.1770 Safety Limiter, which is applied only to the digital radio processing. This control affects the behavior of the digital output receiving the HD feed (see step 10 on page 2-32).

BS.1770 Loudness Control Threshold sets the threshold of the BS.1770 Safety Limiter and the calibration of the BS.1770 loudness meter.

When the BS.1770 Safety Limiter is OFF, the BS.1770 LOUDNESS CONTROL THRESHOLD sets the calibration of the BS.1770 Loudness Meter, such that

Function	Control Name	Values
HF Shelving Filter	HD EQ Gain	0…–6 dB; 0.5 dB steps
	HD EQ Freq	2…– 20 kHz; 1/6-octave steps
Look-Ahead Limiter	HD Limiter Drive	0…+12 dB; 1.0 dB steps
HD De-Esser	HD De-Ess	Off, –23.5…0
FM→HD Control Coupling	FM→HD Control Coupling	FM→HD; Independent
Bass Clipper	HD Bass Clip Threshold	–6.0 … 0.00
	HD Speech Bass Clip Thr	–6.0 … 0.00
	Bass Clip Shape	0 … 10

Table 3-13: Unique HD Audio Controls (found in HD Limiting page)

"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.

Unique HD Audio Controls

Included with each preset in the "HD Limiting" page in Advanced Control. The only HD controls described in this section and shown in Table 3-13 have no FM analog counterparts. There are many other HD controls that have analog FM counterparts. These are described in earlier parts of Section 3 in this manual.

FM→HD Control Coupling determines if audio controls affecting the HD equalizer and HD multiband limiter will track their counterparts in the FM analog processing chain or if the HD and FM controls can be adjusted separately. Each preset contains its own setting for this control, so some presets may be coupled while others are independent. The FM→HD mode causes almost all HD controls to track the settings of their FM counterparts. Only the controls in the HD LIMITING page are adjustable separately. The INDEPENDENT mode allows you to set the HD equalizer and HD multiband compressor/limiter audio controls independently of their FM counterparts. INDEPENDENT mode exposes HD controls that are hidden in FM→HD mode.

If a given preset has certain HD and FM controls set differently, setting the FM→HD CONTROL COUPLING to COUPLED will immediately cause the HD controls to take the same settings as their FM analog counterparts.

On 8600 PC Remote, there are buttons available on the button bar to toggle the FM→HD mode. When the FM→HD mode is INDEPENDENT, a button on the toolbar toggles between corresponding FM and HD tabs (for example, between COMPRESSORS and HD COMPRESSORS).

HD EQ Gain determines the depth of high frequency shelving equalization produced by the parametric HF shelving equalizer, which will be placed either before or after the digital-channel look-ahead limiter (depending on the setting of the HD HF SHELF EQ control in the INPUT / OUTPUT HD DIGITAL RADIO screen).

When placed before the look-ahead limiter, this equalizer is sometimes useful for reducing the audible disparity between the FM and digital-channel outputs (although the HD BAND MIX controls are probably more appropriate for this task). The digital-channel output receives no high frequency limiting or clipping and may therefore be as much as 6 dB brighter than the FM output. If you wish to reduce this difference to smooth out the audible difference between the two channels during a receiver crossfade, you can apply HF rolloff to the digital-channel channel by ear. Another reason you might want to do this is if the digital-channel channel sounds excessively bright after you have optimized the 8600's tuning for FM. Of course, you can also use the HD MB BAND MIX controls for this purpose, or use these controls in conjunction with the HF shelving equalizer.

Yet another reason to use HF rolloff in the digital-channel channel is to reduce codec artifacts at the high frequencies—the familiar "watery" sound.

When placed after the look-ahead limiter, the shelving EQ can reduce the effects of codec overshoot.

The parametric HF shelving equalizer can only produce HF rolloff. It cannot boost.

HD EQ Freq sets the corner frequency of the parametric HF shelving equalizer.

HD Limiter Dr sets the drive level to the digital-channel look-ahead limiter.

When the BS.1770 Safety Limiter is active, you must use the HD LIMITER DR control to set the amount of gain reduction that the BS.1770 Safety Limiter produces. See step 20.C) on page 2-41.

The factory default is +4. When the BS.1770 Safety Limiter is active, the HD LIMITER DR control will usually need to be set substantially below 0 dB unless the preset is quiet (like one of the CLASSICAL presets).

When setting up an HD Radio facility, set the OUT LEVEL control to 0 dBFS. Then adjust the HD LIMITER DRIVE in the on-air preset to match the loudness of the HD channel to the loudness of the FM channel at the receiver.

In an HD Radio receiver, the HD channel has 5 dB higher gain than the analog FM channel. Because the digital channel's loudness must match the FM channel during receiver crossfades, there is no need to overprocess the HD channel; you can take advantage of the 5 dB of "bonus loudness" in this channel to do 5 dB less peak limiting compared to the FM channel. *Do not match the loudness by turning down the Out Level control of the 8600 output driving the HD exciter*. This will force you to use unnecessarily large amounts of limiter gain reduction, which will waste the "bonus loudness."

HD "loudness wars" will not only reduce quality but will also cause unbalanced, obtrusive crossfades between the analog and digital channels in the radio. To brand your station's sound, you can choose the precise coloration you want on the digital channel. You can still take advantage of all of the artistic choices implicit in stereo enhancer, equalization, and multiband compression / limiting settings. Yet you do not need to use excessive peak limiting, which can only reduce quality.

HD De-Ess allows you to mix an adjustable amount of the high frequency limiter gain control signal into Band 5 of the HD processing only. This control allows you to reduce the high frequency response of the HD processing channel in a program-adaptive manner and is intended to reduce strong sibilance ("ess" sounds) that might otherwise sound obtrusive. Higher (less negative) numbers give more de-essing action but will also be more likely to reduce "sparkle" with music. If you notice excessive sibilance, we recommend that you start by setting the control at –18.0 and fine-tune it to taste from there. Once you have set the HD DE-ESS control to your liking, save the resulting preset as a User Preset.

The default setting is –18, which provides a good compromise between brightest sound (which would be accomplished by setting the control to

OFF) and control of high frequency artifacts that low bitrate codecs introduce.

Starting with V2.0 software, Band 5 in the HD processing chain has its own gain control sidechain that is independent of the FM processing chain. Therefore, you can activate the HD Band 5 compressor to either complement or replace the control that the HD DE-ESS control provides.

HD Bass Clip Threshold sets the threshold of the HD processing chain's hard bass clipper (the only bass clipper mode available in the HD chain) with reference to the threshold of the HD processing chain's peak limiter. (See *Bass Clip Thresh* on page 3-47). There are two controls, one for music mode and one for speech mode.

HD Bass Clip Shape (See *Hard Clip Shape* on page 3-49.)

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

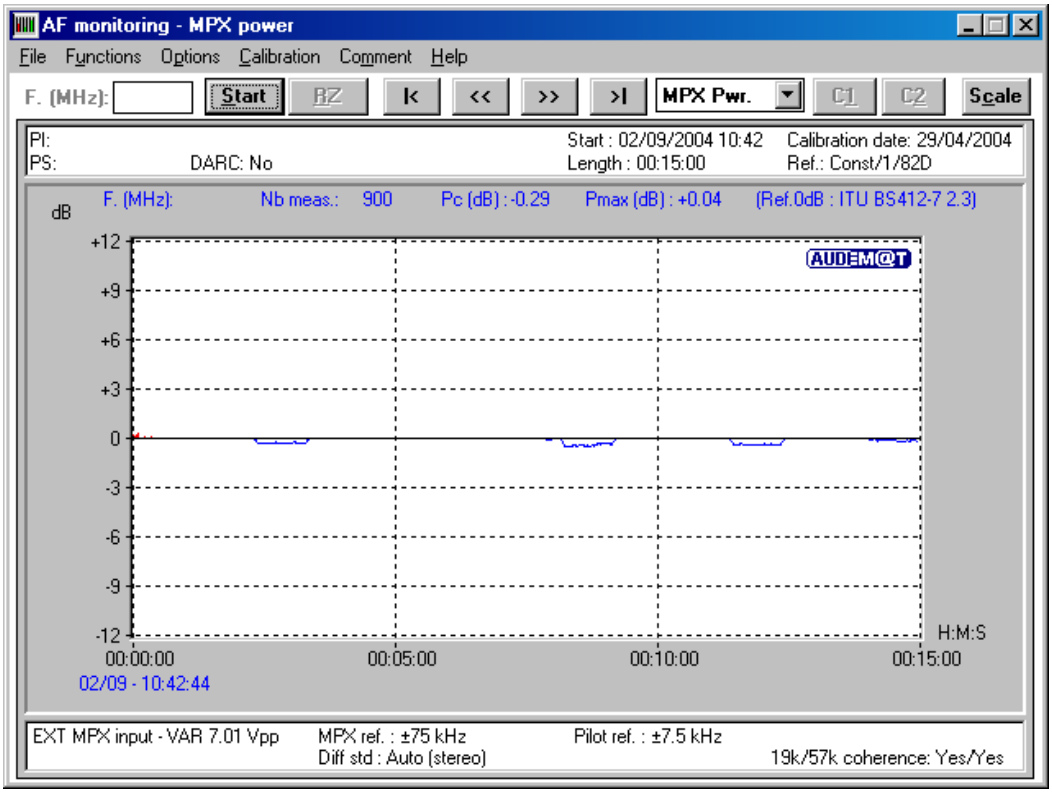


Figure 3-4: Multiplex power over 15 minute observation interval
with Multiplex power controller active, measured at the Optimod's composite output

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 3-13 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 13 on page 2-25 for instructions on setting up the BS.1770 Safety Limiter.

MPX Power Meter

Version 1.2 of 8600 software added a MPX POWER meter that 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 8600's MPX POWER meter is therefore calibrated so that it indicates 0 dB when the composite output of the 8600 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 8600'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 8600. Such overshoots necessitate turning down the 8600's output level control after it has been calibrated with tone using an FM modulation meter and the 8600'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 8600's output level control was turned down.

See *Optimal Control of Peak Modulation Levels* starting on page 1-13 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 8600 does not digitize subcarriers applied to its subcarrier inputs, the 8600'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 8600 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 8600 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[2]. 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 all 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 8600. Setting the control at "0" will correctly control the multiplex power when there is no overshoot after the 8600. This will typically be true when you are using your Optimod's built-in stereo encoder to drive the transmitter directly.

Section 1 of this manual has an extensive discussion of overshoot in transmission paths. See page 1-15 and following pages.

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 8600. For example, if RF peak deviation exceeds the peak deviation produced by the 8600'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

Starting with version1.2 software, 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 8600 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

[2] In versions below 1.2.2, the MPX Power Controller does not work correctly when the stereo encoder is in the experimental SSB mode. V1.2.3 removes this limitation.

the dynamic gain reduction produced by the multiplex power controller. See the notes on the MULTIPLEX POWER OFFSET control on page 3-45.

In version 1.2 and higher software, the MPX offset is still 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 8600'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 if 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 8600'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 8700i'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

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 3-14: Test Modes shows the facilities available, which should be self-explanatory.

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 3-14: Test Modes

Getting the Bass Sound You Want

Probably the most frequently asked question we get regarding 8600 setup is "How do I get a (such-and-such) bass sound?" It seems that individual preference varies in this area more than anywhere else.

There are no magic formulas. The 8600 has extremely versatile controls affecting bass sound and will allow you to get almost any sound you want as long as that sound respects the laws of physics—or, in this case, the laws of psychoacoustics.

The ear is far less sensitive to bass than to midrange sounds. You can see this for yourself by examining the classic Fletcher-Munson "equal-loudness" curves. This means that if you want robust bass, this will take up a great deal of room in your modulation waveform. This room could otherwise be used for midrange, where far smaller amounts of energy yield the same amount of loudness. Accordingly, there is an important tradeoff between loudness and bass—if you want more bass, you will have to accept either less loudness or noticeably more distortion, which occurs when the bass waveforms push the midrange and high frequency material into the 8600's final clipper.

There is one psychoacoustic trick you can use to create more apparent bass while using modulation headroom efficiently. For hundreds of years, pipe organ makers have tricked the ear into hearing non-existent fundamental tones (which would require huge, expensive pipes) by replacing them with several, smaller pipes tuned to the lower harmonics of the missing fundamental. In the 8600, you can use the bass clipper to make harmonic distortion for this purpose. As explained above, the bass clipper has three settings—SOFT, MEDIUM, AND HARD—that determine the amount of distortion the clipper makes when its clips bands 1 and 2. SOFT provides the purest sound, but MEDIUM and HARD create progressively more distortion on bass. Because HARD can make noticeable voice distortion, the factory programmers prefer MEDIUM for most presets. However, if you are willing to trade off voice distortion against bass punch, then you could also use HARD. HARD is particularly effective in increasing

bass punch because it flattops bass transients and this allows the waveform to accommodate fundamentals that have a larger peak level (by up to 2 dB) than the peak level of the flat-top. (The fundamental of a square wave has a peak level 2.1 dB higher than the peak level of the square wave.) In essence, by doing this, your bass fundamentals can exceed 100% modulation without having the composite stereo waveform itself exceed this level.

The attack time of the band 1 compressor affects bass punch by determining the amount of bass transient that is allowed to pass through the compressor before the attack clamps down the rest of the waveform. Any transient that passes through the band 1 compressor will hit the bass clipper, so slower attack times on band 1 will increase bass punch at the expense of distortion (particularly on voice). The BAND 1 ATTACK TIME settings in the factory presets have been adjusted with this tradeoff in mind, but you might prefer to make a different one.

The threshold of the band 1 compressor will also affect bass punch. We recommend that you carefully study the setting of this control (and the BAND 1 ATTACK TIME control) in the various 8600 factory presets before making your own adjustments, so you can get a feel for how we made the tradeoff between punch and distortion at the factory. If you set the threshold much above –6 dB, you will typically get some distortion even on steady-state waveforms (depending on where you have set the BASS CLIPPER THRESH control).

This control is the primary means of trading off bass punch against IM distortion caused the bass' pushing non-bass material into the final clippers. Set it more negative for less punch but less IM distortion.

There are two bass equalizer sections—the **low bass shelving equalizer** and the **bass parametric equalizer**. The main thing to remember about these sections is that they are static tone controls that apply coloration equally to all program material entering the main dynamics processing section of the 8600. (They do not affect the AGC section, being located after it in the signal flow.) Accordingly, the five-band compressor in the 8600 will attempt to undo any coloration added in the equalizer setting and will automatically re-equalize the sound to the standard established by the band threshold controls.

Therefore, to get bass to survive the dynamics processing in the 8600, it is usually necessary to apply substantial bass boost to the input by using the equalizer controls. (A small amount of boost will just be "automatically re-equalized" away; check the factory presets to see what we mean by "substantial.") Bear in mind that using large amounts of shelving bass boost (particularly with 12- or 18 dB/octave slopes) can cause an effective loss of mid-bass because the band 2 compressor will be forced to produce additional gain reduction.

Another important control that affects bass is the BAND 1 OUTPUT MIX control. Because this is located after the dynamics processing, the dynamics processing will not fight any adjustments you make to this control. However, the downside is that the bass compressor will not act to prevent excessive drive to the clipping system (and consequent distortion), so be very careful when boosting this control.

The crossover between band 1 and band 2 is adjustable to 100 Hz, 150 Hz, or 200 Hz by the B1 / B2 XOVER control. When the crossover is set to 100 Hz, band 1 affects extreme low bass (the kind of bass that small clock and portable radios do *not* reproduce), while band 2 affects the mid-bass and lower midrange. Setting the crossover to 200 Hz will cause more gain reduction to occur below 200 Hz because more energy is applied to the band 1 compressor. If you now increase the fixed bass boost by using the LOW BASS equalizer with an 18 dB/octave slope and 120 Hz tuning, the net result will be a dynamic reduction of bass power, typically centered around 160 Hz. If you use enough low bass boost, there will also be a slight increase in the bass power below 100 Hz or so. This 160 Hz suck-out can give an extremely solid, punchy bass sound on radios with good bass response (particularly on radios with subwoofers) but may cause smaller radios to sound thin. (This is the bass formula used in the two GREGG presets.) The rest of the presets use the 100 Hz crossover and have more mid-bass.

Bass Punch in the MX Presets

The MX presets have two controls that allow you to trade off IM distortion against bass punch: BASS PRE-LIMITING and BASS LIMITING. Setting the BASS PRE-LIMITING and BASS LIMITING controls closer to "0" preserves more bass energy below 125 Hz but simultaneously increases IM distortion. Setting these controls away from "0" tends to reduce energy below 125 Hz (although this is quite program-dependent). When it reduces bass energy below 125 Hz, the MX peak limiter tends to *increase* energy between 125 and 200 Hz, which can create the impression of more bass on smaller radios. See the discussion of these controls on page 3-51.

Summary

Bass is a matter of preference, but the canny broadcast engineer will be aware of the variability of radios out there and will not apply excessive bass boost that can sound *awful* on "boom-boxes" and other consumer radios with bass boost already built-in. It is usually wise to emulate the bass balance of hit CDs, because very experienced people who make these trade-offs every day have mastered these. The 8600 provides enormous flexibility to get the bass sound you want, but this flexibility comes at a price—you have to familiarize yourself with the relevant controls, truly understand what you are doing, and work within the laws of physics. This manual is there to help and it is worthwhile to reserve some time with if you want to become an 8600 bass expert.

Using the 8600 PC Remote Control Software

8600 and 8600FM do not use the same PC Remote software. Be sure to use 8600 PC Remote with an 8600 and 8600FM PC Remote with an 8600FM.

8600 PC Remote control software allows you to access any front-panel 8600 control. The software also gives you the ability to backup user presets, system files, and automation files on your computer's storage devices (hard drives, floppy drives, etc.) and to restore them later to your 8600.

> *Note to users familiar with Optimod-FM 8400:* 8600 PC Remote is a completely new application compared to 8400 PC Remote. 8600 PC Remote's GUI and its backup and restore functionality are different. 8600 PC Re-

mote also simplifies upgrading your Optimod's software by managing the upgrade process automatically. Therefore, even if you are familiar with 8400 PC Remote it is still worthwhile to read the following information on 8600 PC Remote.

The 8600 PC Remote software can connect to your 8600 via modem, direct serial cable connection, or Ethernet network. It communicates with your 8600 via the TCP/IP protocol, regardless of how it is connected to your 8600.

PC Remote works best on displays of 1024x768 pel or higher. Scroll bars will appear when using lower resolutions.

Before running 8600 PC Remote, you must have installed the appropriate Windows communications services on your computer. By default, the installer installs a shortcut to `8600PC.exe` on your desktop and in your Start Menu under Orban\Optimod 8600.

8600 PC Remote can control only one 8600 at a time but it can readily switch between several 8600s. 8600 PC Remote has a built-in "address book" that allows it to select and connect to:

- any 8600 on the same network as the PC,
- any 8600 that can be accessed through a modem connected to the PC via dial-up networking, and,
- any 8600 that is connected directly to the PC's serial port(s).

Before your PC can communicate with a given 8600, you must first set up a "connection," which is information that allows PC Remote to locate and communicate with the 8600.

To set up a new connection:

A) Launch 8600PC.exe.

B) Create a new 8600 connection by choosing NEW 8600 from the CONNECT file menu or by right clicking on the ALL CONNECTIONS icon in the Connections List and selecting NEW 8600.

The Connection Properties dialog box opens.

C) Enter an Alias name for your 8600 (like "KABC").

D) Leave the password field blank to prompt the user to enter a password when initiating a connection.

Refer to *Security and Passcode Programming* on page 2-43.

Otherwise, enter a password to allow PC Remote to connect to your 8600 without requiring a password when the connection is initiated.

E) If you are communicating with your 8600 through a network, select the ETHERNET CONNECTION radio button and enter the appropriate IP address, subnet mask, port, and gateway data. These must agree with the values you set

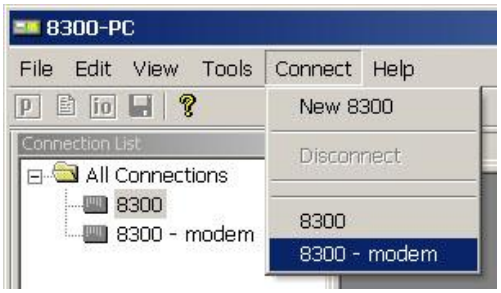
in step 1 on page 2-63. See also *Setting Up Ethernet, LAN, and VPN Connections* on page 2-68.

F) If you are communicating via a direct serial cable connection or a modem connection, follow the appropriate procedure described in *Appendix: Setting up Serial Communications,* starting on page 2-75.

G) Click OK after entering all required information.

To initiate communication:

Initiate communication by double-clicking on the desired 8600 alias in the Connections List or by selecting the desired 8600 alias from the CONNECT drop down menu. (This screenshot refers to the 8300 but applies equally to the 8600.)



- If the connection is successful, a dialog bubble will appear on the top left hand corner of the screen verifying your connection.
- If an Enter Passcode dialog box appears, enter a valid passcode and the 8600 PC Remote software will initiate a connection to the 8600 unit.

A window will appear saying, "Connecting to the 8600." A few moments later, a new message will appear: "Please Wait. Updating Local Files."

When run, the Orban PC Remote software installer makes copies of all 8600 factory preset files on your local hard drive. The PC Remote software reads these files to speed up its initialization. If any of these files have been deleted or damaged, the PC Remote software will refresh them by downloading them from the 8600. If the PC Remote software needs to do this, it can substantially increase the time required for the software to initialize, particularly through a slow modem connection.

All communications between your Optimod and PC Remote are encrypted and all transient files that PC Remote writes to your computer's hard drive are also encrypted.

When this download is finished, the main meters will appear.

- A wheel mouse is the quickest and easiest interface to use; you will rarely (if ever) have to use the keyboard.
- The help box at the bottom of the screen always presents a short help message for the function you have selected.

To modify a control setting:

A) Choose PROCESSING PARAMETERS from the EDIT menu.

B) Select menu tabs for Less-More, Stereo Enhancer, and EQ to access Basic Modify controls. All other menu tabs contain Full or Advanced Modify controls.

> You can reset any Basic Modify Control without losing LESS-MORE functionality; Full and Advanced modify control adjustments will cause LESS-MORE to be grayed-out.

- To set a control, click it (it will become highlighted) and then adjust it by dragging it with the mouse or moving the wheel on the mouse.
- You can also use the ← and → keys on the numeric keypad to adjust any control.

To recall a preset:

A) Choose RECALL PRESET from the FILE menu to bring up the RECALL PRESET FILE dialog box.

B) Click the desired preset within the dialog box to select it.

C) To put a desired preset on-air, double-click it or select it and click the RECALL PRESET button.

> Continually clicking the RECALL PRESET button will toggle between the current and previous on-air presets.

D) Click DONE to dismiss the OPEN PRESET FILE dialog box.

> The folder on your hard drive containing the preset files (both Factory and User) is automatically synchronized to the contents of its associated 8600's memory each time 8600 PC Remote connects to that 8600. The 8600's memory is the "master." This means that if you delete a user preset from the 8600's memory (whether locally via its front panel or via 8600 PC Remote), 8600 PC Remote will automatically erase this preset from this folder on your computer. *To archive a preset permanently, you must use the Backup function.* (See page 3- 91.)

To save a user preset you have created:

A) Select SAVE PRESET AS from the FILE menu to bring up the SAVE AS Dialog Box. The current preset name will appear in the File Name field.

B) Click in the field and edit it.

C) Click SAVE to save the preset to the 8600 as a User Preset.

If you have made edits to a previously existing user preset, you can select SAVE PRESET from the FILE menu to overwrite the pre-existing user preset automatically.

To back up User Presets, system files, and automation files onto your computer's hard drive:

A) Select BACKUP TO PC from the FILE Menu.

B) Click OK.

PC Remote will offer three options:

- Save backup files (User Presets, system files, and automation) in plain text.

  This allows the presets and files to be read with any text editor program and to be readily exchanged between Optimod users.

- Save backup files using the session passcode to encrypt them.

- Save backup files using the password of your choice to encrypt them.

  The encryption options prevent archived presets, system files, and automation files from being restored if the user does not have the password used for the encryption. *There is no "back door"— Orban cannot help you to decrypt a preset whose password is unknown.*

All User Preset, system, and automation files are copied from your Optimod's internal memory to a folder called "backup" on your PC. This folder is a subfolder of the folder named the same as the alias of the Optimod that you are backing up.

This folder name ("backup") and location are hard-coded into the software. If you wish to move the backup files somewhere else later, use a file manager (like Explorer) on your computer.

Starting with version 1.02 software, the folder into which PC Remote saves user preset files has been changed. Previously, PC Remote saved these files in the path C:\Program Files (x86)\Orban\Optimod 8600 PC Remote\presets\[Optimod 8600 serial number]. This is inconvenient in Windows 7 because it is difficult for Windows 7 users to modify and access files within the Program Files folder.

The new default is:

.. \Documents \Orban\Optimod 8600 PC Remote\

The actual folder location in Windows XP is:

C:\Documents and Settings\ All User\ Shared Documents\Optimod 8600 PC Remote

The actual folder location in Windows 7 and higher is:

C:\Users\Public\Documents\Orban\Optimod 8600 PC Remote

To make more than one backup archive, rename the current backup folder (for example, to "Backup1"). 8600 PC Remote will create a new backup folder the next time you do a backup, leaving your renamed backup folder untouched. Later, you will be able to restore from any folder—the Restore dialog box allows you to choose the folder containing the files to be restored.

If you try to back up a preset with the same name as a preset existing in the Backup folder, but with a different date, 8600 PC Remote will warn you and will allow you to overwrite the preset in the Backup folder or to cancel the operation. If you wish to keep the existing archived preset, you can first use a file manager to move the existing user preset in the Backup folder to another folder; then repeat the backup operation.

Note to Users Familiar with Older Version of PC Remote

To make the user presets easily accessible to all users (including those without Administrator privileges in Windows), we have moved the default location of the user preset folder from inside Windows Program Files directory, where it was in previous software versions.

For Windows XP, the new location is:

```
C:\Documents and Settings\
All Users\Documents\Orban\
Optimod 8685 PC Remote\my 8685
```

For Windows 7 and higher, the new location is:

```
C:\Users\Public\Documents\Orban\
Optimod 8685 PC Remote\my 8685
```

To restore archived presets, system files, and automation files:

In addition to restoring an archived preset to its original Optimod, you can also copy archived presets from one Optimod to another. The Optimod whose connection is active will receive the preset.

> If the preset, system file, or automation file was encrypted when it was originally saved, PC Remote will request the password under which it was encrypted.

All User Presets are compatible with all 8600 software versions and are compatible with both 8600 and 8600FM. If Orban adds new controls to a software version, the new software will assign a reasonable default value to any control missing in an old User Preset. If you archive such a User Preset after restoring it, the newly written archive file will now include the new controls (with the default values, unless you edit any of these values before you re-archive the preset).

- If you load an archived 8600 preset file into an 8600FM, the 8600FM will ignore all digital radio-specific parameters in the file.
- If you load into an 8600 a User Preset that was originally archived from an 8600FM, the 8600 will usually apply the digital radio parameters that the User Preset inherited from its source Factory Preset. The main exception is a User Preset that originally created in an 8600, was then loaded into and modified by an 8600FM, and was finally loaded into an 8600 again. In this case, any HD settings in the original 8600 User Preset are retained.

In addition, you can load archived preset files from OPTIMOD-FM 8400 and 8500. The 8600 supports all 8400 and 8500 features. 8400 presets will sound very similar on the 8600. (They will not sound 100% identical because the base sample rate of the 8400 is 32 kHz and the base sample rate of the 8500 and 8600 is 64 kHz.) 8500 presets will sound identical when running on the 8600.

A) Select RESTORE FROM PC from the FILE menu.

A standard Windows dialog box will open.

B) Select the type of files you want to restore using the FILES OF TYPE field at the bottom of the dialog box.

You can elect to restore all user presets (*.orb86user, *.orb), 8600 user presets (*.orb86user), 8400 user presets (*.orbu), system files (*.orb86setup), and automation files (*.orb86autom).

If you want to restore files from a different directory (i.e., that might have been created on a different 8600), navigate to that directory from within the dialog box.

C) To restore a single user preset:

a) Set the FILES OF TYPE field to a user preset file type (*.orb86user, *.orbu).

b) Select the desired preset in the dialog box.

c) Click the RESTORE button.

D) To restore all the user presets from a specific location:

a) Set the FILES OF TYPE field to a user preset file type (*.orb86user, *.orbu)

b) Highlight all the user presets in the dialog window

c) Click the RESTORE button.

E) To restore a system file:

a) Set the FILES OF TYPE field to the System Setup file type (*.orb86setup).

b) Select the desired system file in the dialog box.

c) Click the RESTORE button.

F) To restore an automation file:

a) Set the FILES OF TYPE field to the Automation file type (*.orb86autom).

b) Select the desired automation file in the dialog box.

c) Click the RESTORE button.

G) Click DONE to dismiss the RESTORE dialog box.

To share an archived User Preset between 8600s:

A) Navigate to the directory containing the desired User Preset from within the RESTORE FROM PC dialog box

B) Click the RESTORE button.

This User Preset will be downloaded to the 8600 to which 8600 PC Remote is currently connected.

If the User Preset is encrypted, PC Remote will request its password.

To modify INPUT/OUTPUT and SYSTEM SETUP:

Choose SETUP from the TOOLS menu.

To set a control, click it (it will become highlighted) and then use the wheel on the mouse to adjust it. You can also use the ← and → keys on the numeric keypad to adjust any control.

To modify AUTOMATION:

A) Choose AUTOMATION from the TOOLS menu.

An Automation Dialog box will open.

B) Click the NEW EVENT to create a new event

Controls to set the event type and time are available on the right hand side of the dialog box. (See *Using Clock-Based Automation* on page 2-42 for an explanation of 8600 automation.)

C) Check the ENABLE AUTOMATION check box at the top of the dialog box to enable automation.

To group multiple 8600s:

Right-click ALL CONNECTIONS in the Connections List and select NEW GROUP.

You can add multiple 8600s to a single group to help organize a network of 8600. However, only one 8600 from within a group can be connected to 8600 PC Remote at any one time.

Operation Using the Keyboard

In general, PC Remote uses standard Windows conventions for navigation.

Navigate around the screens using the TAB key. Use CTRL-TAB to move to the next tabbed screen in PC Remote.

Use the ← and → keys on the numeric keypad to adjust control settings.

To Quit the Program

Use standard Windows conventions: Press ALT-F4 on the keyboard, or click the X on the upper right corner with the mouse.

Also, please note the following behavior:

- If you close the PC Remote connection from the PC, you will be given the choice of staying connected through the ppp or disconnecting.
- If you close the connection from PC Remote but choose not to close the ppp connection, the END PC REMOTE button will remain displayed on the 8600's front panel. If you then select that button, the ppp connection will close.

This behavior ensures that a user can tell from the 8600's front panel if a remote connection is active. Users can disconnect the PC connection at the 8600 if they wish.

This minimizes the likelihood of someone's leaving a connection open while someone else tries to access that 8600.

About Aliases created by Optimod 8600 PC Remote Software

When you ADD A NEW 8600 using Optimod 8600 PC Remote, your 8600 is automatically given an 8600 Alias name to differentiate it from other 8600s. You can change the name anytime in the 8600 Properties window inside 8600 PC Remote.

When you add a new 8600 or change the name of an existing 8600 Alias, an Alias folder is created in the same location as the executable for Optimod 8600 PC Remote (usually \Program Files (x86)\Orban\Optimod 8600). The folder has the same name as the Alias name. Once you establish the initial connection to the 8600, all presets for that 8600 are automatically copied to the Alias folder; thus, the folder contains all the preset files for that 8600, both Factory and User. If you have backed up the 8600 using 8600 PC Remote, there will also be a "backup" subfolder located within the Alias folder.

PC Remote always encrypts the User Presets that it automatically copies from your Optimod to your hard drive when PC Remote connects. These are "transient" files because PC Remote automatically erases them from your hard drive when you quit PC Remote. However, PC Remote allows you to choose whether to encrypt *archived* user preset files when you archive them. (Archiving is never automatic—see *To back up User Presets, system files, and automation files onto your computer's hard drive* on page 3-91.) If you do not encrypt them, archived User Presets are text files and can be opened in a text editor (like Notepad) if you want to examine their contents.

Alias folders and their associated backup subfolders are registered in your PC's Registry. This prevents folders from being accidentally deleted or moved. If you move or delete Alias folders from the PC, the Alias folders recreate themselves in the previous location and restore their contents by copying it from their associated 8600s when 8600 PC Remote connects to such an 8600.

Multiple Installations of Optimod 8600 PC Remote

Rarely, you may want to have more than one installation of 8600 PC Remote on your computer. There are a few extra things to know if you have multiple installations.

If you install a new version of the Optimod 8600 PC Remote software on your PC, any Alias folders and backup subfolders created in an earlier software version still remain in their original location on your PC (and in its registry).

The version of 8600 PC Remote must match the version of the software in the 8600 controlled by it. Therefore, you will only need multiple installations of PC Remote (having separate version numbers) if:

- you are controlling multiple 8600s, and
- not all of your 8600s are running the same version of 8600 software, and

- you do not want to upgrade at least one controlled 8600 to the latest version of 8600 PC Remote software.

Each version of 8600 PC Remote has its own top-level folder, normally under \Program Files (x86)\Orban. (The default folder is \Program Files (x86)\Orban\Optimod 8600.) When you install a new version of 8600 PC Remote, the default behavior is to overwrite the old version, which is usually the desired behavior. To prevent the installer from overwriting the old version, you must specify a different installation folder when you install the new version (for example, \Program Files (x86)\Orban\Optimod 8600v2).

Each version of 8600 PC Remote will display *all* 8600 Aliases, even those pointing to 8600s with incompatible version numbers. If you attempt to connect to an older version of 8600 from a newer version of 8600 PC Remote, 8600 PC Remote will offer to upgrade the software in the target 8600 so that it corresponds to the version of 8600 PC Remote that is active. If you attempt to connect to newer version of 8600 from an older version of 8600 PC Remote, it will refuse to connect and will emit an error message regarding incompatible versions.

If you decide to install the new software to a different location on your PC, new Aliases created using the new software will not be located in the same place as the old Aliases.

To Move Alias Folders:

Even though each version of 8600 PC Remote can see all aliases, you may wish to move the corresponding folders so they are under the folder corresponding to the highest version of 8600 PC Remote that is currently installed on your computer (although this is not required). If your Alias folders reside in different locations, you can move all the Alias folders to the same location by using the PC Remote software. *Do not use an external file manager to do this.* The old Alias folders need to be recreated under the Optimod 8600 PC Remote software you wish to use (so that the registry entries can be correctly updated). You can do this two different ways.

- **Rename the Alias** (preferred): Start the Optimod 8600 PC Remote executable you wish to use and rename your old Aliases with a slightly different name. A new Alias folder with the new name will be created in the same location as the Optimod 8600 PC Remote executable.

- **Delete and Recreate the Alias:** Start the Optimod 8600 PC Remote executable you wish to use. Delete the old 8600 Aliases and create new ones to replace them. New Alias folders will be created in the same location as the Optimod 8600 PC Remote executable.

  Important: The deletion process will automatically erase its associated folder, including the Backup directory. If you have anything in the Backup directory that you wish to keep, you should therefore move that directory elsewhere (or transfer the desired files to another, active backup directory).

Ordinarily, the erasure process will move the Backup directory to your computer's Recycle Bin, so you can recover a Backup directory that you have accidentally deleted in this way.

White Paper: Measuring the Improvements in Optimod-FM 8600's FM Peak Limiting Technology

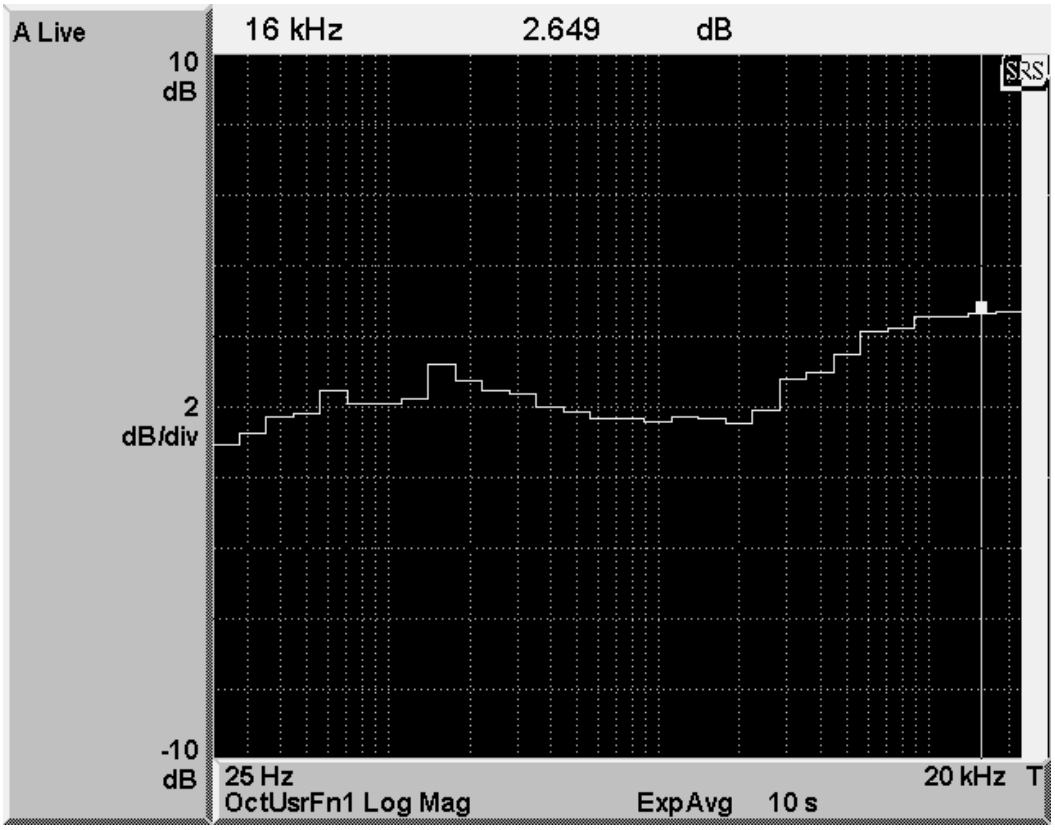


Figure 3-5
Madonna, "Get Together" remix; third-octave difference

Introduction

The 8600 offers new "MX" peak limiter technology that decreases distortion while increasing transient punch and high frequency power handling capacity. Compared to the FM-channel peak limiter in Optimod-FM 8500, the new peak limiter typically provides 2.5 to 3 dB more power at high frequencies, which minimizes audible HF loss caused by preemphasis limiting. Drums and percussion cut through the mix. Highs are airy. "Problem material" that used to cause audible distortion is handled cleanly.

While this design offers about the same loudness as 8500 processing, its main goal is to make FM analog broadcasts more competitive with the cleanliness, punch, and open high frequencies of the digital media against which FM analog transmissions now battle.

The 8600's "MX" technology peak limiter consists of two main stages—first, a pre-limiter stage to control low frequency and high frequency peaks ahead of the main peak limiter and then the main peak limiter, which uses various proprietary techniques to control distortion, minimize transient loss, and minimize high frequency loss.

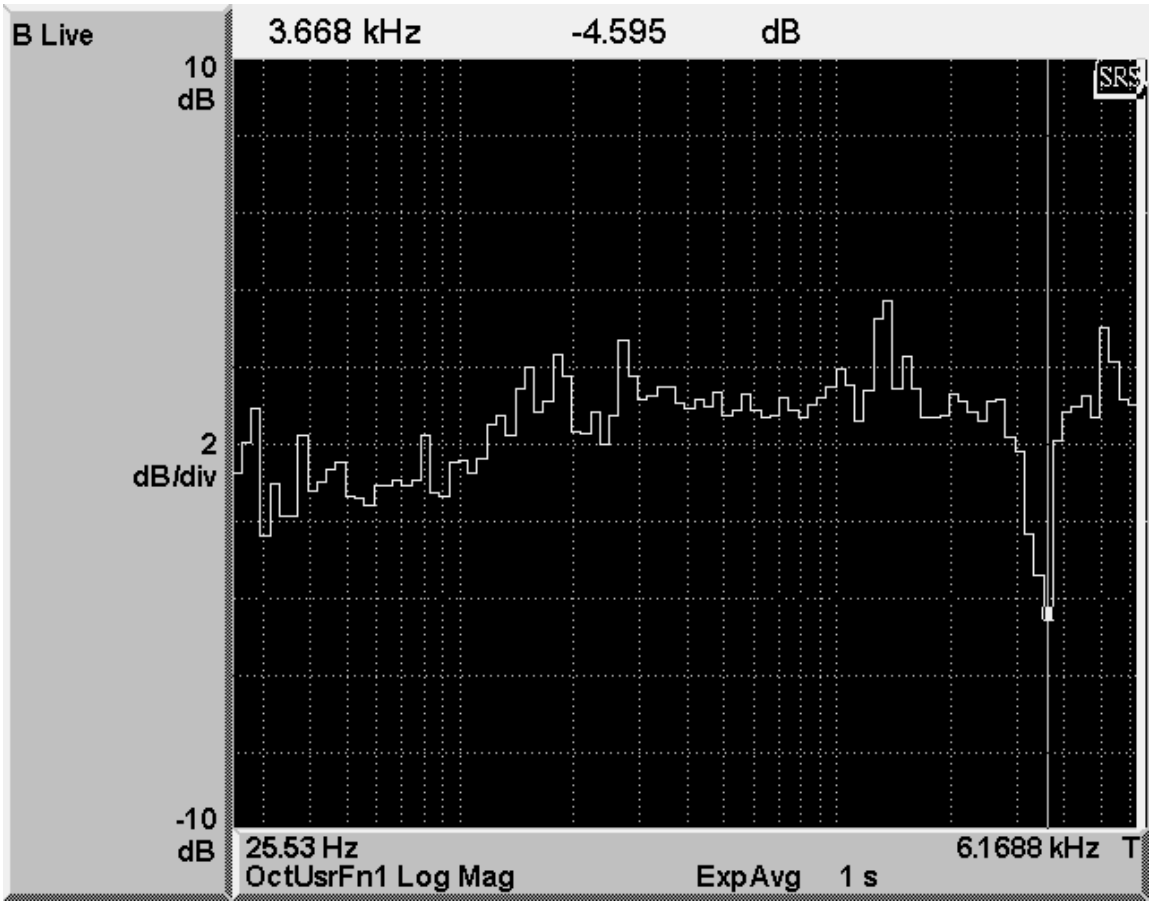


Figure 3-6
U2, "Within You Without You"

Measurements

The improved performance of the MX peak limiter can be verified by objective measurements with real-world program material. To measure the improvements in high frequency power handling capability and dynamic distortion, we use a Stanford Research Systems SR785 Two-Channel Dynamic Signal Analyzer (Figure 3-10 on page 3-104) to compare the 8500 and 8600. This analyzer has two filterbanks[3] and allows the two third-octave or twelfth-octave spectra (measured in decibel units) to be sub-

[3] ANSI standard S1.11- 1986, Order 3, Type 1-D

tracted in real time. The difference shows the difference in the output spectra of the two units under test, measured in dB.

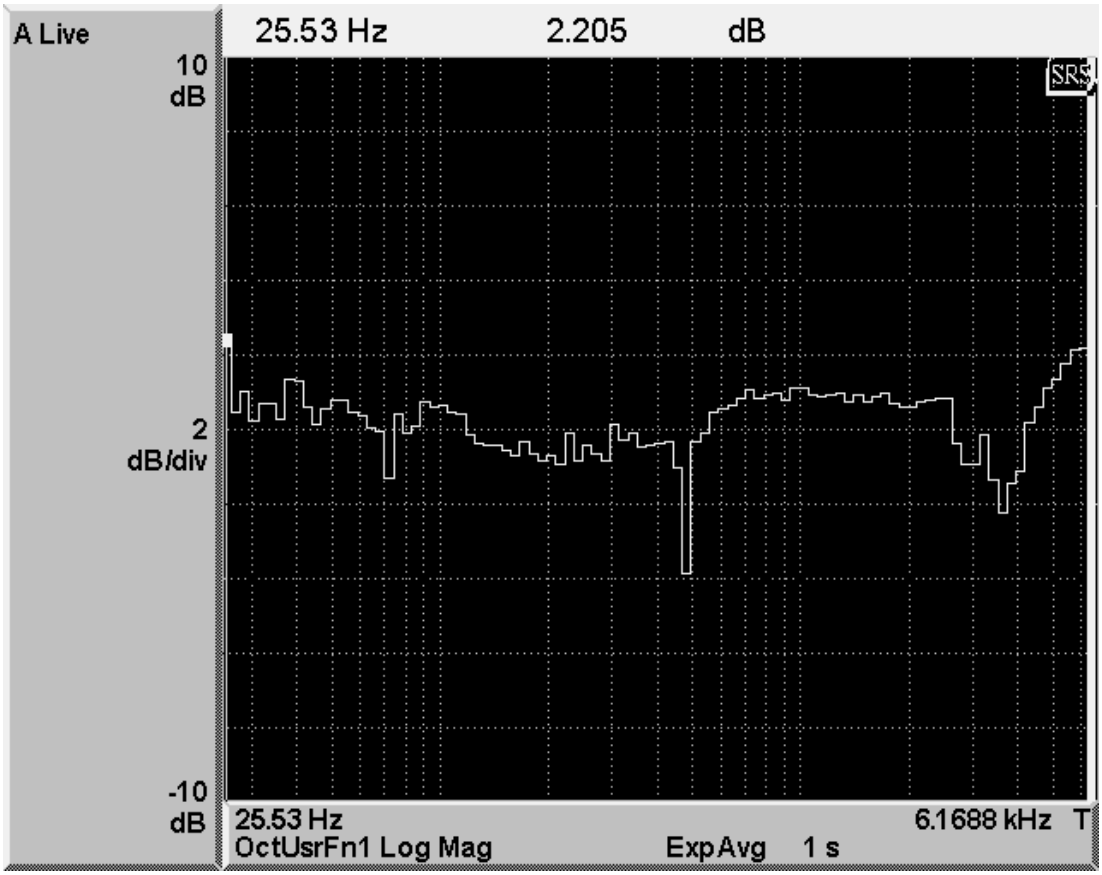


Figure 3-7
Kelly Clarkson, “Because of You” opening

Unless otherwise indicated, the measurements were made using the GREGG MX preset in the 8600 and the GREGG preset in the 8500. These presets have essentially identical loudness and are designed to have similar spectral balance in the midrange frequencies and below. A higher reading in a given frequency band indicates that that band has more power in the 8600 output than in the 8500 output.

HF Power Handling: Figure 3-5 on page 3-99 shows a typical HF improvement: about 2.6 dB more average HF energy using the new technology. (Coincidentally, there happens to be about 1 dB more energy in the 160 Hz band but this can easily be matched more closely if the user wants.)

Distortion Control: In 1977, Orban was the first company to patent[4] and commercialize a psychoacoustic model to estimate and control audible clipping distortion. The

[4] US patents 4,208,548 & 4,241,266, used in Optimod-AM 9000A

8600 also uses a psychoacoustic model, but in a far more sophisticated way that has some similarities to audio codec technology.

Figure 3-6 on page 3-100 is a twelfth-octave ratio that compares the spectrum of the 8500 and 8600 outputs in a more detailed way than the third-octave ratio seen in Figure 3-5. It shows the 8600's 4.6 dB suppression of (mainly harmonic) clipping distortion centered at 3.668 kHz caused by sustained electric guitar riding on top of bass.

Figure 3-7 on page 3-101 shows a piece of material that has been used by others to demonstrate distortion control. This material consists of Clarkson's humming over a quiet piano background. There is little program energy in the upper midrange, so any clipper-induced distortion tends to stand out.

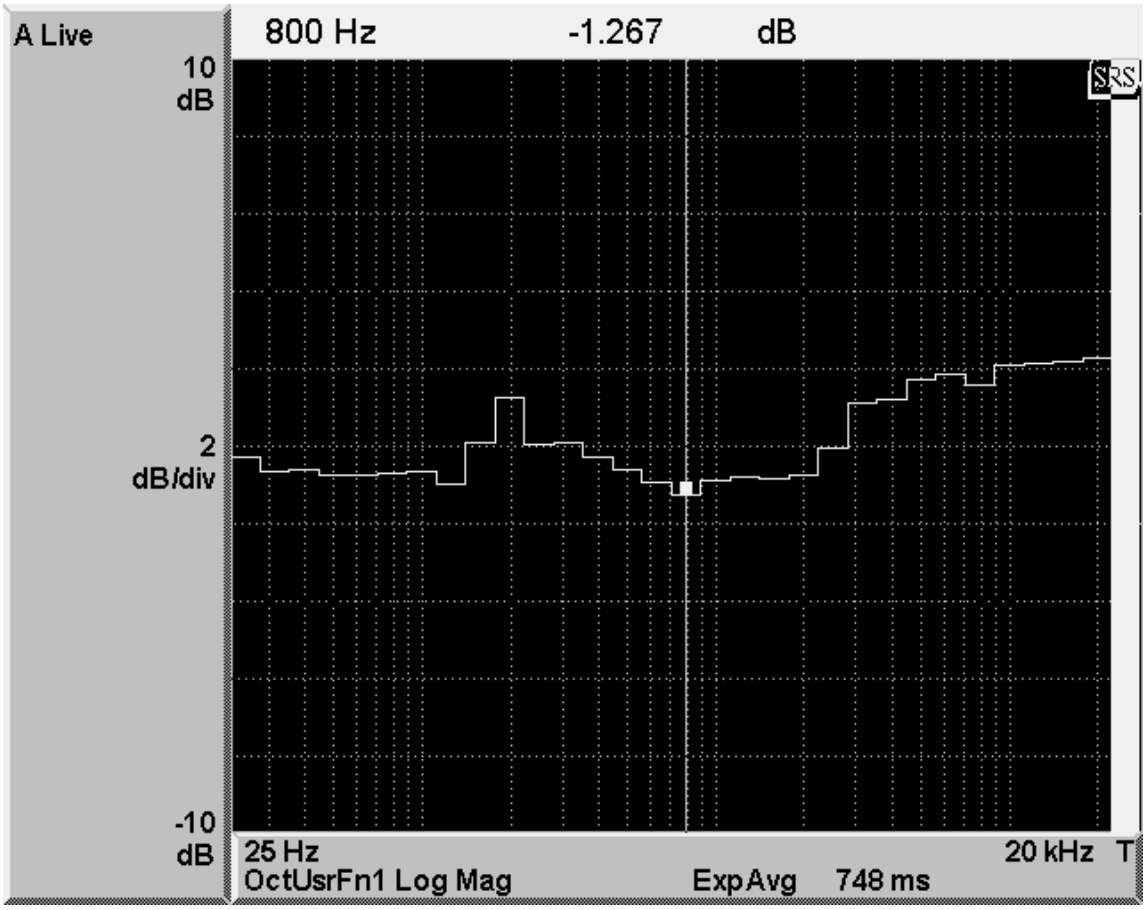


Figure 3-8
Level 42, "Starchild"

The twelfth-octave ratio shows as much as 4 dB suppression of narrowband difference-frequency IM distortion in the midrange. To cause the 8500 to produce noticeable distortion, we used the IMPACT preset, which is a louder preset than GREGG or

GREGG MX. To keep the comparison "apples and apples," we used the IMPACT MX preset in the 8600. Even with these presets, the distortion in either Orban processor was negligible compared to the effect of even modest multipath and considerably less than certain other non-Orban processors produce with this material. (Note that the increase in the output starting slightly above 6 kHz does not represent distortion but is instead another manifestation of the 8600's better HF power handling capability, which we first presented in Figure 3-5.)

Figure 3-8 on page 3-102 shows an example of program material with much more energy above 6 kHz than in the midrange, making difference-frequency intermodulation in the midrange easily audible. Although other, non-distorted material in the midrange makes the measurement less dramatic compared to the previous pictures, the combination of improved HF response (about 2 dB) and midrange IM suppression is plainly visible and very audible in the 8600.

Transient Punch: Decreasing distortion often comes at the expense of decreased transient punch because in competitively processed FM, transients must be clipped and may therefore appear incorrectly to the distortion controller as "distortion"

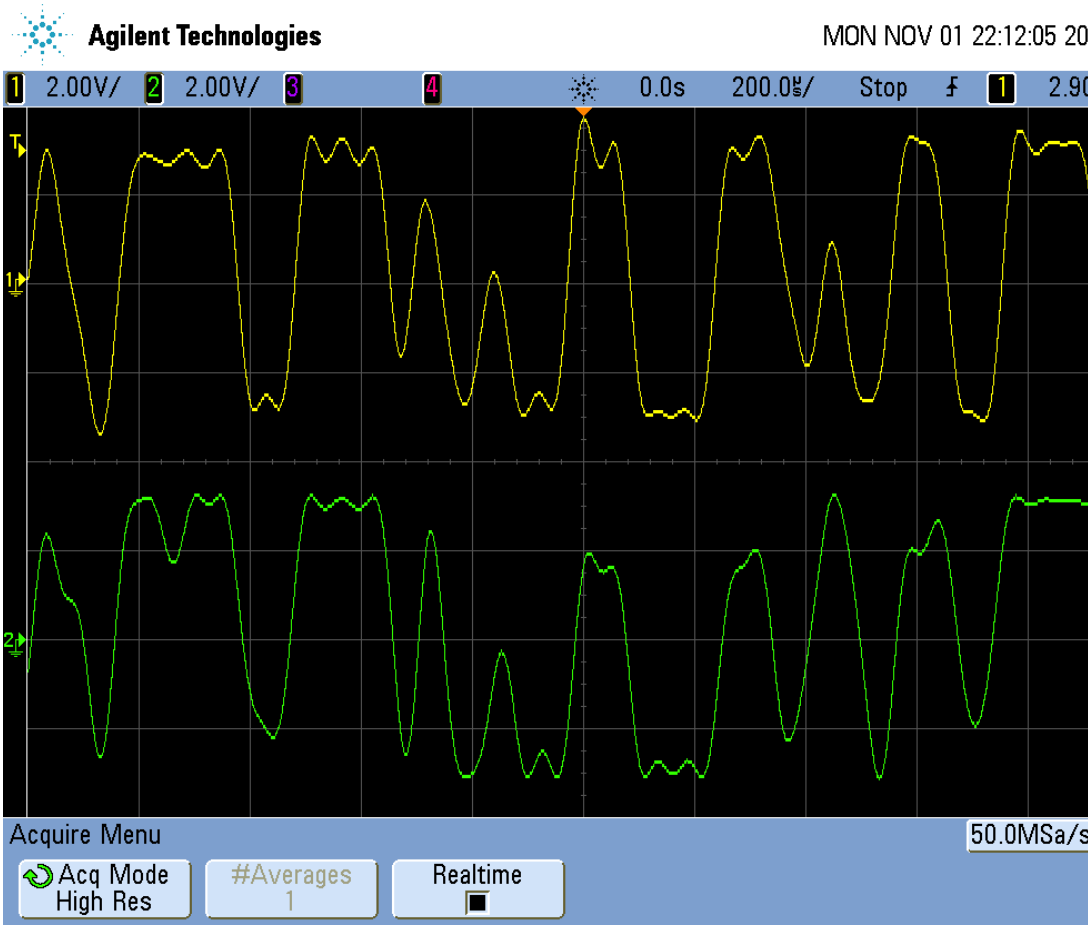


Figure 3-9
Simple Minds, "Alive and Kicking"

that requires suppression. By using sophisticated signal processing, the MX limiter not only avoids this problem but improves transient punch compared to the older technology.

The center (under the small orange triangle) of Figure 3-9 on page 3-103 shows a snare drum hit from Simple Minds, "Alive and Kicking." The original CD dates from the days before "hypercompression" in mastering ruined the sound of most CDs; the snare drum has plenty of peak level. The new limiter preserves the transient (center of the screen) about 3 dB more effectively than does the old technology (bottom).

Conclusions

The marketing of broadcast audio processors has often suffered from objectively unverifiable hyperbole. Meanwhile, traditional measurements using static test signals like single-tone harmonic distortion and multi-tone IM distortion cannot rigorously predict the subjective listening qualities of broadcast processors because these processors are purposely very nonlinear, so that the assumptions implicit in test-tone measurements (that the systems being measured are ideally linear and that their nonlinearities are weak) do not hold.

This white paper has shown that with today's sophisticated digital measurement techniques, it is possible to verify some subjective claims objectively by making measurements that use real-world program material as the excitation. These meas-

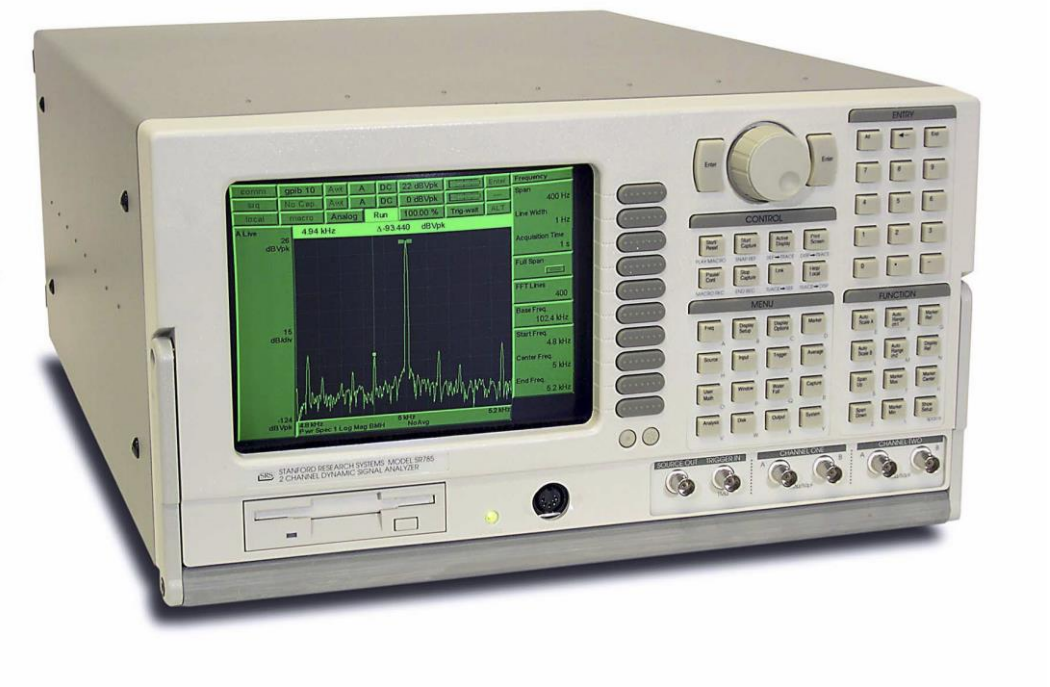


Figure 3-10
Stanford Research Systems Model SR785 Dynamic Signal Analyzer

urements have verified that the 8600's new limiter technology has significantly improved high frequency power handling capability, distortion control, and transient punch compared to Orban's previous flagship FM processor.

—Robert Orban

Section 4 Maintenance

Routine Maintenance

The 8600 OPTIMOD-FM Audio Processor uses highly stable analog and digital circuitry throughout. Recommended routine maintenance is minimal.

1. **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.

2. **Listen to the 8600's output.**

   A good ear will pick up many faults. Familiarize yourself with the "sound" of the 8600 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 8600 is at fault. The troubleshooting information in Section 5 will help you determine if the problem is with OPTIMOD-FM or is somewhere else in the station's equipment.

3. **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 8600 chassis contacts the rack.

4. **Periodically check for loss of grounding.**

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

5. **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.

Subassembly Removal and Reinstallation

See page 6-37 for the *Circuit Board Locator and Basic Interconnections* diagram.

1. **Removing the Top Cover.**

   To access the main boards, power supply board or display assembly, you must remove the top cover.

   A) Disconnect the 8600 and remove it from the rack.

      Be sure power is disconnected before removing the cover.

Hazardous voltage is exposed when the unit is open and the power is ON.

   B) Set the unit upright on a padded surface with the front panel facing you.

   C) Remove seventeen thread-forming screws and five machine screws holding the top cover in place and lift the top cover off.

      Use a #1 Phillips screwdriver.

2. **Removing the Input/Output Assembly.**

   A) Make sure that AC power is disconnected from the 8600.

   B) Remove the 14-conductor ribbon cable from the base board at JP600.

   C) Remove the two 40-conductor ribbon cables from the DSP board at J602 and J603.

   D) Remove the power cable at J601.

   E) Remove the three Phillips screws holding the rear of the Input/Output assembly to the floor of the chassis.

   F) Remove the ten Phillips screws holding the input/output assembly to the rear panel.

   G) Remove the Input/Output assembly.

3. **Removing the DSP Board.**

   A) Make sure that AC power is disconnected from the 8600.

   B) If you have not done so yet, remove the top cover (step 1 on page 4-2).

   C) Disconnect the ribbon cable from J504.

   D) Disconnect the two ribbon cables from the DSP board to J602 and J603 on the Input/Output assembly

   E) Remove the ribbon cable from J701 on the DSP board

   F) Remove the cable assembly from J200 of the DSP board.

G) Remove the six Phillips screws holding the DSP board to the bottom of the chassis.

H) Remove the DSP board.

4. Removing the Front Panel.

To service the headphone amplifier, the color LCD display, the pushbuttons, or the rotary encoder, it is first necessary to remove the front panel assembly.

A) Make sure that AC power is disconnected from the 8600.

B) Remove the ribbon cable from J200 on the display interface board.

C) Remove the cable assembly that connects through the fire wall to the headphone amplifier board.

D) Remove the six Phillips head screws that hold the front panel to the main chassis.

These are located in two groups of three on the sides of the main chassis, close to the front panel.

E) Pull the front panel toward you to remove it.

F) Do not stress the cables connecting the front panel to the main chassis.

To protect the assembly from cosmetic damage, set it down on a soft surface like foam rubber, a quilt, or a blanket.

5. Removing the Headphone Amplifier Board.

Because they are socketed, you can remove and replace the headphone amplifier driver chips without further disassembly.

If you need to remove the headphone amplifier circuit board (to access components other than the headphone amplifier driver chip):

A) Make sure that AC power is disconnected from the 8600.

B) Pull the friction-fit knob off the headphone volume control.

C) Remove three Phillips screws.

This will free the board.

6. Preparing to Remove the Rotary Encoder Board.

The circuit board containing the pushbuttons, joystick, and rotary encoder is mounted on a metal shield plate. To remove the plate, remove three screws and lift the plate off at a 45-degree angle, following the axis of the rotary encoder.

7. Removing the Rotary Encoder Board.

Remove the four screws holding the board to the standoffs and lift the board from the standoffs.

All of the knobs and buttons are friction-fit and can be removed, if necessary, by pulling them off their shafts. However, to avoid possibly damaging the rotary encoder, we advise not removing its knob unless necessary. Instead, to access the

screw partially blocked by the rotary encoder's knob, use a small screwdriver and attack the screw head from a slight angle, avoiding the edge of the knob to prevent cosmetic damage.

The pushbutton switches, joystick, and rotary encoder are all soldered to this board and can be replaced by normal solder rework techniques.

8. **Removing the Color LCD Display and carrier board.**

   A) Make sure that AC power is disconnected from the 8600.

   B) Remove the cable assembly from J14 on the base board.

   C) Remove the 33-conductor flat ribbon cable from the display interface board at J103:

      a) Carefully disconnect the cable by rotating the black "wing" at the rear of the connector 90° from horizontal to vertical.

      b) Slide the cable out of the connector.

      You may find it easier to first remove the display interface board from the CPU board stack.

   D) Remove the four screws that hold the display carrier board to the standoffs on which it is mounted. Then lift the assembly off the standoffs.

9. **Removing the RS-232 Connector Board:**

   A) If you have not done so yet, remove the top cover (step 1, above).

   B) Using a 3/16-inch hex nut driver, remove the six hex nuts holding the RS-232 connectors to the chassis.

   C) Unplug the RS-232 interface assembly from the base board.

10. **Removing the CPU board.**

    The Display Board and CPU Board are a "sandwich" assembly. The CPU board is located on top of the Display Board and is plugged into it.

    A) Make sure that AC power is disconnected from the 8600.

    B) Remove the RJ45 network cable from the CPU board

    C) Remove the four Phillips screws from the CPU board.

    D) Carefully unplug the module by pulling it evenly away from the display interface board.

11. **Removing the Display interface Board.**

    You must first remove the CPU board before removing the Display Interface Board.

    A) Unscrew the four standoffs that had supported the CPU board before it was removed.

    B) Carefully pull the Display Interface Board evenly away from the Base Board, being careful not to stress any ribbon cables still connected to it.

C) Unplug any ribbon cables from the Display Interface Board, which now can be removed completely. Refer to step (8.C)to disconnect the 33-conductor ribbon cable to J103.

12. Removing the Base Board.

You must have completed steps 9, 10, and 11 first.

A) Make sure that AC power is disconnected from the 8600.

B) Remove the three power ribbon cables from the power supply and dress them away from the Base Board.

C) Using a 3/16-inch nut driver, remove the two jackscrews and lock washers holding the DB25 connector to the rear panel.

D) Remove the four Phillips screws and four standoffs holding the Base Board to the bottom of the chassis.

E) Verify that all connectors have been removed.

F) Remove the Base Board.

13. Removing the Power Supply assembly.

To remove the power supply it is necessary to remove the 8600 from the rack and to remove the top cover. It is most convenient to remove the Power Supply Assembly if the Base Board, RS232 Board, CPU board, and Display Interface Module have been removed.

A) Be sure that the AC line cord is disconnected from the power supply.

B) Unplug the three ribbon cables from the power supply.

C) Unplug the two cable assemblies from the power transformer by squeezing the locking tab and removing the connector.

D) Remove the nut securing the green ground wire to the chassis.

E) Remove the two Phillips screws securing the mains input connector to the rear of the chassis

F) Remove the three Phillips screws at the bottom edge of the power supply

G) Remove the four Phillips screws holding the power supply assembly to the top apron of the chassis.

H) Remove the power supply assembly.

14. Reinstalling the Power Supply Assembly:

A) Hold power supply board into main chassis, so that it aligns with the four mounting holes on the top apron of the chassis.

B) Replace the four Phillips screws holding the power supply assembly to the top apron of the chassis, but do not fully tighten them yet.

C) Replace but do not fully tighten the two Phillips screws that hold the IEC connector.

D) Replace and fully tighten the three Phillips screws at the bottom edge of the power supply

E) Fully tighten the four Phillips screws holding the power supply assembly to the top apron of the chassis

F) Fully tighten the two Philips that hold the IEC connector.

G) Replace the two cable assemblies from the power transformer.

H) Replace the nut securing the green ground wire to the chassis.

I) Reattach the three ribbon cables to the power supply.

J) Reattach the two cable assemblies from the power transformer.

15. Reinstalling the I/O Board and DSP board:

Referring to steps 2 and 3, follow the instructions in reverse.

16. Reinstalling the Base Board.

Referring to step 12, follow the instructions in reverse.

Note that you cannot replace the RS-232 board, Display Interface Board, and the CPU board until you have replaced the base board.

17. Reinstalling the Display Interface Board.

Referring to step 11, follow the instructions in reverse.

- To avoid bent pins or other damage, verify that all connector pins are aligned before applying force.
- Verify that pin one of the ribbon cables (red stripe) is oriented correctly. Pin 1 is indicated by the number "1" or a square pad.
- Note that you cannot replace the CPU board until you have replaced the Base Board and the display interface board.

18. Reinstalling the CPU Board:

To avoid bent pins or other damage verify that all connector pins are aligned before applying force.

Referring to step 10, follow the instructions in reverse.

19. Reinstalling the RS-232 Board:

Referring to step 9, follow the instructions in reverse.

20. Reassembling the Color LCD Display and carrier board.

Referring to step 8, follow the instructions in reverse.

Verify that pin one of the ribbon cables (red stripe) is oriented correctly.

Pin 1 is indicated by the number "1" or a square pad.

21. Reinstalling the Rotary Encoder Board.

Referring to steps 6 and 7, follow the instructions in reverse.

Verify that pin one of the ribbon cables (red stripe) is oriented correctly.

Pin 1 is indicated by the number "1" or a square pad.

22. Reinstalling the Headphone Amplifier Board.

Referring to step 5, follow the instructions in reverse.

23. Reinstalling the Front Panel.

Referring to step 4, follow the instructions in reverse.

24. Check your work.

A) Referring to the cable wiring diagram on page 6-37, verify that all cables have been securely reattached.

B) Verify that all removed hardware has been replaced and is secure.

25 Replace the Top Cover.

Place top on the unit and reattach the seventeen thread-forming screws and five machine screws.

The 8600 can now be returned to service.

Field Audit of Performance

Required Equipment:

- Ultra-low distortion sine-wave oscillator / THD analyzer / audio voltmeter

  With verified residual distortion below 0.01%. Audio Precision APx, or similar high-performance system.

  The **NAB Broadcast and Audio System Test CD** is an excellent source of test signals when used with a high-quality CD player.

- Spectrum analyzer with tracking generator

  Stanford Research Systems SR760 or equivalent. Alternatively, a sweep generator with 50-15,000 Hz logarithmic sweep can be used with an oscilloscope in X/Y mode, or you can use a computer-controlled test set like the Audio Precision System One.

- Digital voltmeter

  Accurate to ±0.1%.

- Oscilloscope

  DC-coupled, triggered sweep, with 5 MHz or greater vertical bandwidth.

- Two 620Ω ±5% resistors.

- *Optional:* Audio Precision System 1 (without digital option) or System 2 (for digital tests).

It is assumed that the technician is thoroughly familiar with the operation of this equipment.

This procedure is useful for detecting and diagnosing problems with the 8600's performance. It includes checks of frequency response, noise and distortion performance, and output level capability.

This performance audit assesses the performance of the analog-to-digital and digital-to-analog converters and verifies that the digital signal processing section (DSP) is passing signal correctly. Ordinarily, there is a high probability that the DSP is performing the dynamic signal processing correctly. There is therefore no need to measure such things as attack and release times—these are defined by software and will automatically be correct if the DSP is otherwise operating normally.

It is often more convenient to make measurements on the bench away from high RF fields which could affect results. In a high RF field it is, for example, very difficult to accurately measure the very low THD produced by a properly operating 8600 at most frequencies. However, in an emergency situation (and is there any other kind?), it is usually possible to detect many of the more severe faults that could develop in the 8600 circuitry even in high-RF environments.

See the assembly drawings in Section 6 for component locations. Be sure to turn the power off before removing or installing circuit boards.

Follow these instructions in order without skipping steps.

Note: To obtain an unbalanced output, jumper pin 1 (ground) to pin 3 and measure between pin 1 (ground) and pin 2 (hot).

Note: All analog output measurements are taken with a 620Ω ±5% resistor tied between pin 2 and 3 of the XLR connector.

1. **Prepare the unit.**

   A) Set the GND LIFT switch to the earth ground symbol setting (left position) to connect chassis ground to circuit ground.

   B) Use the front panel controls to set the 8600's software controls to their default settings, as follows:

      a) From the main menu, choose INPUT/OUTPUT. Using the LOCATE joystick, navigate in turn to each of four INPUT/OUTPUT screens and write down the settings so you can restore them after testing.

      b) Navigate to the INPUT/OUTPUT > INPUT screen. Set controls as in the table below:

      Set Input to ..... analog
      Analog Ref. Level..... +4.0 dBu

Clip Level ..... +20.0 dBu
Right Channel Balance .....0.0 dB
DI REF VU..... –15.0 dBFS

c) Navigate to the INPUT/OUTPUT > OUPUT 1 screen, Set controls as in the table below:

Analog Out Level ..... +10.0 dBu
Analog Output Source ..... Xmitter
Analog Preemphasis .....Flat
AES 1 Out Level..... –2.8 dBFS
AES 1 Out Source ..... Xmitter
Sample Rate .....44.1 kHz
AES 1 Preemphasis.....Flat
Word Length 20
Dither.....Out

d) Navigate to the INPUT/OUTPUT UTILITIES screen. Set the MULTIPLEX POWER THRESHOLD control to OFF.

e) Navigate to SYSTEM SETUP. Select TEST MODES. Then Activate Bypass mode: Navigate to the BYPASS button and press ENTER. Set controls as in the table below:

Mode ..... Bypass
Frequency.....400 Hz
Mod Level..... 100%
Mod Type .....L+R
Bypass Gain .....0 dB

NOTE: Bypass defeats all compression, limiting, and program equalization but retains the selected preemphasis (either 50μs or 75μs).

2. Test the power supply

A) If the power supply is entirely dead and the fuse is not blown, verify that the primary winding of the power transformer is intact by measuring the resistance of the power supply at the IEC AC line connector.

For 115-volt operation, the resistance should be approximately 7.6Ω.

For 230-volt operation, the resistance should be approximately 27Ω.

B) The green LED power indicator on the upper right of the front panel display monitors the DC power supply outputs. If one or more power supply voltages are out of tolerance, red flashes will report them according to the table below. If there are multiple values out of tolerance, they are reported one after another in a continuous loop, with one green flash indicating the beginning of each count.

Number of Red Flashes	Problem With
1	+ unregulated supply
2	+15V or –15V
3	+5V or –5V

Number of Red Flashes	Problem With
4	+5V Digital
5	Analog ↔ Digital ground connection broken
6	DSP A +3.3V supply
7	DSP B +3.3V supply
8	CPU +3.3V supply
9	CPU +2.5V supply

Table 4-1: Decoder Chart for Power Supervisor

You can monitor power supply voltages at connector J7 on the power supply board (see page 6-51 for the parts locator drawing and page 6-52 for the schematic). When one faces the connector, the voltages can be found on the pins in the following pattern:

(1) + unreg.	(3) digital gnd	(5) +15V	(7) +5 V digital	(9) –5V analog
(2) - unreg	(4) chassis gnd	(6) -15V	(8) +5V analog	(10) NC

Table 4-2: Layout Diagram of J7, with expected voltages on each pin

C) Measure the regulated voltages at J7 with the DVM and observe the ripple with an oscilloscope, AC-coupled. The following results are typical:

Power Supply Rail	DC Voltage (volts)	AC Ripple (mV p-p)
+15VDC	+15 ± 0.5	<20
–15VDC	–15 ± 0.5	<20
+5VDC	+5 ± 0.25	<20
–5VDC	–5 ± 0.25	<20
Digital +5VDC	+5 ± 0.25	[Obscured by noise]

Table 4-3: Typical Power Supply Voltages and AC Ripple

3. Check Analog Output Trim Levels.

A) Verify 8600 software controls are set to their default settings. (Refer to page 4-8.)

B) Feed the 8600 output with the built-in 400 Hz Test tone.

To turn on the TEST tone:

a) Navigate to SYSTEM SETUP.

b) Choose TEST MODES.

c) Navigate to the TONE button and press ENTER.

C) Connect the audio voltmeter to the Left Analog Output.

D) Adjust output trim VR200 to make the meter read +10.6 dBu. (0 dBu = 0.775V rms.) Verify a frequency reading of 400 Hz.

E) Verify THD+N reading of <0.05% (0.02% typical) using a 22 kHz low pass filter in the distortion analyzer.

F) Recall the bypass preset: Navigate to the BYPASS button and press ENTER.

Bypass defeats all compression, limiting, and program equalization but retains preemphasis.

G) Verify a reading (noise) of <–80 dBu at the output of the unit.

H) Repeat steps (C) through (G) for the Right Analog Output, adjusting output trim VR201 at step (D).

4. Check frequency response of Analog I/O.

A) Verify 8600 software controls are set to their default settings. (Refer to page 4-8.)

B) Be sure you are still in BYPASS mode [see step (3.F)].

C) Connect the oscillator to the Left Analog Input XLR connector.

D) Inject the Analog Input XLR connector with a level of 0 dBu (20 dB below the ANALOG CLIP LEVEL setting) with the oscillator set to 100 Hz.

This is 20 dB below the clip level, which allows headroom for preemphasis. 75μs preemphasis will cause 17 dB of boost at 15 kHz.

E) Connect the audio analyzer to the 8600's Left Analog Output XLR connector.

F) Verify a level of 0 dBu ±1 dB. Use this level as the reference level.

G) Verify that frequency response at 50 Hz, 100 Hz, 400 Hz, 5 kHz, and 15 kHz is within ±0.1 dB of the reference level.

This procedure tests the analog input circuitry, the A/D converter, the DSP, the DAC, and the analog output circuitry.

H) Repeat steps (C) through (G) for the right channel.

5. Check distortion performance of Analog I/O.

A) Verify 8600 software controls are set to their default settings. (Refer to page 4-8.)

B) Be sure you are still in BYPASS mode [see step (3.F)].

C) Press ESCAPE until you see the Main Meter screen

D) Connect a THD analyzer to the Left Analog Output XLR connector. Set the THD analyzer's bandwidth to 22 kHz.

E) Connect the oscillator to the Left Analog Input XLR connector.

F) For each frequency used to measure THD, adjust the output level of the oscillator to make the COMP meter on the 8600 read 100.

You will have to reduce the output level of the oscillator at higher frequencies to compensate for the preemphasis boost in the 8600.

G) Measure the THD+N at the frequencies listed below.

Frequency	THD+N Typical	THD+N Maximum
50 Hz	0.015%	0.03%
100 Hz	0.015%	0.03%

400 Hz	0.015%	0.03%
1 kHz	0.015%	0.03%
2.5 kHz	0.015%	0.03%
5 kHz	0.015%	0.03%
7.5 kHz	0.015%	0.03%
10 kHz	0.015%	0.03%
15 kHz	0.015%	0.03%

H) Repeat the above measurements for the right channel. Connect the oscillator to the right analog input and the distortion analyzer to the right analog output.

I) Disconnect the oscillator and THD analyzer from the 8600.

6. Test Digital Sample Rate Converter (Receiver).

A) Verify 8600 software controls are set to their default settings. (Refer to page 4-8.)

B) Be sure you are still in BYPASS mode [see step (3.F)].

C) Navigate to INPUT/OUTPUT. On screen INPUT/OUTPUT > INPUT, SET INPUT TO: DIGITAL.

D) Connect the digital source generator to the AES3 Digital Input XLR connector of the 8600.

E) Set the frequency of the digital source generator to 400 Hz and its output level to 6 dB below full scale.

F) Inject the Digital Input with a sample rate of 32 kHz, 44.1 kHz, 48 kHz, 88.2 kHz, and 96 kHz. Use 24-bit words.

G) Listen to the analog outputs of the 8600 and verify that the output sounds clean and glitch-free regardless of the input sample rate.

H) Leave the digital source generator connected to the 8600.

7. Test Digital Sample Rate Converter (Transmitter).

A) Set the sample rate of the digital source generator to 48 kHz.

B) Navigate to Screen INPUT/OUTPUT > OUPUT1.

C) Connect an AES3 analyzer (like the Audio Precision System 2) to the 8600's AES3 digital output #1.

D) Change the SAMP RATE to 32 kHz, 44.1 kHz, 48 kHz, 88.2 kHz, and 96 kHz, and verify that the frequencies measured at the 8600's AES3 output follow the chart below within given tolerances:

Sample Rate	Tolerance (PPM)	Tolerance (Hz)
32 kHz	50 PPM	±1.60 Hz
44.1 kHz	50 PPM	±2.2.0 Hz
48 kHz	50 PPM	±2.40 Hz
88.2 kHz	50 PPM	±4.41 Hz

96 kHz	50 PPM	±4.80 Hz

E) Navigate to Screen INPUT/OUTPUT > OUPUT2 and repeat steps (C) and (D) for AES Output #2.

F) Disconnect the digital source generator from the 8600.

8. Test the 8600's stereo encoder.

A) Connect an accurate stereo monitor like the Belar FMMS-1 ("Wizard") stereo demodulator to the 8600's COMPOSITE OUTPUT 1. This is labeled OUTPUT and appears on a BNC connector on the 8600's rear panel.

> **NOTE:** The recommended Belar monitor is the only instrument we have encountered that can accurately measure the performance of the 8600's stereo encoder. With most older-technology monitors, you will be measuring the performance of the monitor, not the 8600's encoder. Of course, we have not evaluated every monitor on the market.

B) Navigate to the 8600's SYSTEM SETUP and choose TEST MODES.

C) Choose TONE. Set the test tone parameters as follows:

Frequency	400 Hz
Mod Level	91%
Mod Type	L+R
Test: Pilot	On

D) Navigate to INPUT/OUTPUT and then to screen INPUT/OUTPUT > COMPOSITE. Set the OUTPUT 1 LEVEL to make the stereo monitor read 100% total modulation.

E) Navigate to the 8600's SYSTEM SETUP and choose TEST MODES.

F) Measure the L–R level on the stereo monitor at several frequencies, in units of dB below 100% modulation. This is the main channel to subchannel crosstalk. It should not exceed –70 dB, 50-15,000 Hz.

G) Set the MOD. TYPE to L–R. Measure the L+R level on the stereo monitor at several frequencies, in units of dB below 100% modulation. This is the subchannel to main channel crosstalk. It should not exceed –70 dB, 50-15,000 Hz.

H) Set the MOD. TYPE to LEFT. Measure the Right level on the stereo monitor at several frequencies, in units of dB below 100% modulation. This is left into right stereo separation. It should not exceed –55 dB, 50-15,000 Hz and will typically be –60 dB or better.

I) Set the MOD. TYPE to RIGHT. Measure the Left level on the stereo monitor at several frequencies, in units of dB below 100% modulation. It should not exceed –55 dB, 50-15,000 Hz and will typically be –60 dB or better.

J) Set the MOD. TYPE to L–R and the FREQUENCY to 5000.0 HZ. Measure the 38 kHz subcarrier suppression on the stereo monitor. It should not exceed –65 dB.

K) Measure the Pilot Modulation on the stereo monitor. It should read 0%.

L) Set the MOD. LEVEL to 0.0%. Measure the de-emphasized noise at the left and right outputs of the stereo monitor. It should not exceed –80 dB below 100% modulation.

M) Repeat steps (C) through (L) for the 8600's COMPOSITE OUTPUT 2.

N) Measure pilot tone injection: Using the stereo monitor, verify that pilot tone injection is between 8% and 10% modulation. If it is outside these parameters, it can be adjusted by navigating to INPUT/OUTPUT and then to screen INPUT/OUTPUT > COMPOSITE. Adjust PILOT LEVEL as necessary.

O) If the measured pilot level varies by more than a few tenths of percent from the pilot level indicated, this indicates there may be a problem elsewhere—either in your measuring setup, or with the 8600.

P) Measure pilot tone frequency: With the MOD. LEVEL still set to 0.0%, connect a frequency counter to either of the 8600's composite outputs. Verify that the pilot tone frequency is 19,000 Hz ±1 Hz.

9. Optional tests.

A) You can test each GPI input for functionality in the obvious way, by programming a function for it and then verifying that the function executes when you activate the input. To program a GPI input, navigate to SYSTEM SETUP > NETWORK / REMOTE 1.

B) *You can test the RS232 Port 1 for functionality by verifying that you can connect to a PC through a null modem cable. See Installing 8600 PC Remote Control Software on page 2-66.*

10. Return OPTIMOD-FM to service.

A) Remove the 620Ω resistors connected across the outputs.

B) Navigate to the INPUT/OUTPUT screen and restore your normal operating parameters in all four screens, using the notes you made in step (1.B)a) on page 4-8.

C) Navigate to SYSTEM SETUP. Select TEST MODES. Then activate Operate mode: Navigate to the OPERATE button and press ENTER.

D) Recall your normal operating preset.

Section 5 Troubleshooting

Problems and Potential Solutions

Always verify that the problem is not the source material being fed to the 8600, or in other parts of the system.

RFI, Hum, Clicks, or Buzzes

A grounding problem is likely. Review the information on grounding on page 2-12. The 8600 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.

Particularly if you are using a long run of coaxial cable between the 8500 and the exciter, a ground loop may inject noise into the exciter's composite input—especially if the exciter's input is unbalanced. A Jensen JT-123-BMCF transformer can almost always cure this problem—see page 1-15.

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. See step 12 on page 2-24.

If you are using the ITU412 multiplex power controller and have edited a factory preset (including by use of LESS-MORE), you may have to readjust the MULTIPLEX POWER OFFSET control in your edited preset so that the average indication on the MULTIPLEX POWER meter is 2 to 3 dB of gain reduction. (See the notes on the MULTIPLEX POWER OFFSET control on page 3-45.)

Poor Peak Modulation Control / Low On-Air Loudness

The 8600 ordinarily controls peak modulation to an accuracy of ±2%. This accuracy will be destroyed if the signal path following the 8600 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. Section 1 of this manual contains a complete discussion of the various things that can go wrong.

Because of its 64 kHz base sample rate, the output bandwidth of the 8600's FM channel processing is slightly higher (17 kHz) than that of Orban processors using 32 kHz base sample rate. The signal path to the transmitter must be essentially flat to 17 kHz to take advantage of the 8600's excellent peak control at its digital and analog left/right outputs. Using a link with 32 kHz sample rate to pass the 8600's output will increase overshoot by as much as 1.5 dB compared to links with 44.1 kHz or higher sample rate that have essentially flat audio response to 20 kHz. If you have a 32 kHz STL, we recommend placing the 8600 at the transmitter and using the STL it to pass the unprocessed audio to the 8600's input(s). This will allow you to use the 8600's composite output to drive the transmitter. Thanks to the 8600's "Half-Cosine Interpolation" composite limiter, the composite output has exceptionally low overshoot.

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.

If you have inadvertently activated the ITU412 multiplex power controller (see step 12 on page 2-24) then the average modulation can be very low and peak modulation can be held below 100%, depending on program material.

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 8600'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 LIMIT 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 headroom of the unit's analog-to-digital (A/D) converter must be correctly matched to the peak audio levels expected in your sys-

tem (using SYSTEM SETUP). If your peak program level exceeds the peak level you have specified on setup, the 8600's A/D converter will clip and distort. (See page 2-27). The 8600's INPUT LEVEL meters will clearly indicate any such clipping.

If you are using the 8600'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 8600's AGC. In this case, reduce the setting of the AGC's MAXDELTAGR controls.

If you are using an external processor ahead of the 8600, be sure it is not clipping or otherwise causing problems.

Audible Noise on Air

(See also "RFI, Hums, Clicks, or Buzzes" on page 5-1.)

Excessive compression will always exaggerate noise in the source material.

The 8600 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.

There are two independent compressor gate circuits in the 8600. The first affects the AGC and the second affects the Multiband Compressor. Each has its own threshold control. (See MB GATE THRESH on page 3-63.)

In the Multiband structure, *dynamic single-ended noise reduction* (see MB DOWN EXPANDER on page 3-64) 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 8600'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 8600 to digitize the input. (This ratio is better than 108 dB.) It is important to drive the 8600 with professional levels (more than 0 dBu reference level) to achieve adequately low noise. (Clipping occurs at the level set by the CLIP LEVEL control located in INPUT/OUTPUT > INPUT.)

The 8600'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 8600'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 8600, the STL's noise level can severely limit the overall noise performance of the system because compression in the 8600 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 an Orban studio level control device like the 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. (See *Studio-Transmitter Link,* starting on page 1-15.)

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 DACs or other analog circuitry. If it appears only when you use the composite output, then it is likely a problem in the composite DACs 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 8600 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 8600 and the rest of the transmission system are operating correctly, subcarriers should experience no interference.

Interference from the stereo into a subcarrier is best diagnosed with a spectrum analyzer. First examine the spectrum of the 8600'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 8600's composite output and the transmitter has sufficient headroom.

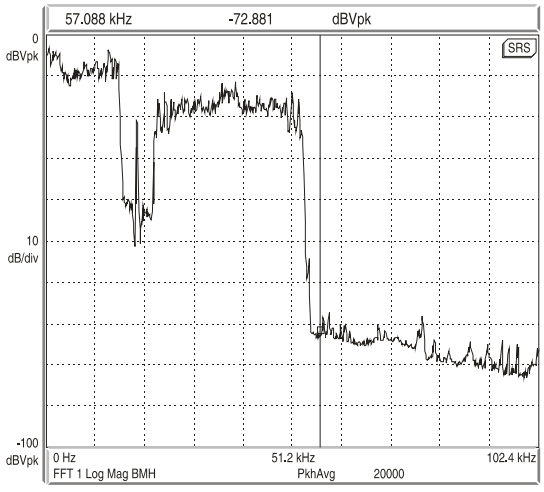


Figure 5-1: Typical 8700 baseband spectrum with heavy processing, 0-100 kHz.

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 any of the Five-Band structures, 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 preemphasis, you must set the 8600's output to Flat in the System Setup / Output screen to prevent double preemphasis, which will cause very shrill sound (and very poor peak modulation control).

You will *always* achieve better peak control by defeating the preemphasis and input filters of an external stereo encoder, permitting the 8600 to perform these functions without overshoot. Section 1 of this manual contains a detailed explanation of these, and other, system design considerations.

Dull Sound

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

If the 8600's output is set to FLAT in INPUT/OUTPUT > OUTPUT1 or OUTPUT2, there will be no preemphasis 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 8600 can generate test tones itself. The 8600 can also be put into Bypass mode (locally or by remote control) to enable it to pass externally generated tones at any desired level. (See *Test Modes* on page 3-85.)

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 8600, or the 8600 should be temporarily switched to BYPASS to pass the tones.

System Receiving 8600's Digital Output Will Not Lock

Be sure that the 8600's output sample rate is set match the sample rate that the driven system expects. Be sure that the 8600's output mode (AES3 or SPDIF) is set to match the standard expected by the driven system.

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; there is no frequency trim available. The crystal is a surface-mount part and is not easily field-replaceable.

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 8600'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.

Talent Complains About Delay in Their Headphones

Dedicate an output to low-delay Monitor mode and use this to drive headphones. See *Monitoring on Loudspeakers and Headphones* on page 1-26, and step 15 on page 2-38. Alternatively, use an Ultra-Low-Latency (UL) preset. See *Ultra-Low-Latency Five-Band* on page 3-20.

HD Output Sounds Too Bright

Enable adjustment of the HD output's spectral balance by putting the on-air preset into INDEPENDENT mode. Do this by setting the FM→HD CONTROL COUPLING control in the HD LIMITING page of ADVANCED CONTROL to INDEPENDENT. You can also toggle this control via two buttons on the button bar in 8600 PC Remote. See page 3-72.

In this mode, the HD processing channel's equalizer, five-band compressor, and band-mix controls are independent of the corresponding controls in the FM channel and can be adjusted separately. You can therefore fine-tune high frequencies by adjusting the equalizer, parameters in the band 4 and band 5 compressors, the band 4 and band 5 BAND MIX controls, and the HD DE-ESS control. One of the most effective ways to tame harsh high frequencies dynamically is to activate the HD channel's band 5 compressor. See *HD Audio Controls* on page 3-79.

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

Adjust the HD DE-ESS control and/or activate the HD channel's band 5 compressor. See page 3-80.

HD and FM Levels Do Not Match When the Receiver Crossfades

Adjust the HD LIMIT DR control in the on-air preset to match levels. See page 3-80.

Do not match levels by adjusting the output level of the output driving the HD exciter. *Only use this control to match the peak levels of the Optimod output to the exciter*. In other words, if the exciter is set to clip when it receives at –3 dBFS, adjust the Optimod output level to –3 dBFS. This uses all of the headroom available in the

transmission channel, minimizing the amount of look-ahead limiting that the Optimod needs to do.

If you have accidentally turned on the BS.1770 Safety Limiter, this will cause the loudness to be lower than expected in HD Radio applications. Turn it off. See step 10 on page 2-32.

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. See step 10 on page 2-32.

Loudness Drops Momentarily During HD Radio Analog/Digital Crossfades

The analog and digital channels in your transmission path have reversed polarity with respect to each other and a phase cancellation is occurring in the radio during crossfades. You can correct this with the HD POLARITY or FM POLARITY controls. See page 3-76.

HD Frequency Response is Limited to 15 kHz

The HD BANDWIDTH control might be set to 15 kHz. See page 3-76.

Even if the HD BANDWIDTH control is set to 20 kHz, bandwidth will be limited to 15 kHz if the HD output's sample rate is set to 32 kHz and/or if the sample rate of the audio applied to the 8600's digital input is 32 kHz. 20 kHz response requires sample rates to be set to 44.1 kHz or higher.

You Cannot Set Any Output to Emit an HD Signal

From the main menu, Locate to SYSTEM SETUP > DIAGNOSTICS to see if your unit is an 8600FM. The 8600FM is the same as the 8600 except that the 8600FM does not provide digital radio processing. The 8600FM can be upgraded to an 8600 in the field by installing the plug-in CPU board contained in the 8600UPG/HD upgrade kit, which can be purchased from your Orban dealer.

The Left and Right Audio Channels are Reversed at the Digital 1 or Digital 2 Outputs only

Your unit's DSP board has a firmware bug that can be fixed in software. Refer to *AES Output L/R Channel Reversal Bug* on page 6-13.

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. See step 12 on page 2-24 and step 13 on page 2-25 .

Digitized SCA inputs do not appear at the analog composite outputs and vice-versa (MPX hardware only)

This is normal: Non-digitized SCA inputs are summed into the analog composite outputs in the analog domain after D/A conversion and digitized SCA inputs are only summed into the digital composite outputs. 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.

General Dissatisfaction with Subjective Sound Quality

The 8600 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.

Compared to competitive processors, the 8600 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 8600, 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.

Section 1 of this manual provides a thorough discussion of system engineering considerations, particularly with regard to minimizing overshoot and noise.

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. See step 20.C) on page 2-41.

Security Passcode Lost (When Unit is Locked Out)

Please see *If You Have Forgotten Your All-Screens Passcode* on page 2-47.

Connection Issues between the 8600 and a PC, Modem, or Network

- **User Interface Slowdown**: The more user presets you make, the more slowly the 8600 will respond to front-panel commands. Delete any user presets you do not need.
- **Software Updates**: Close any running Windows programs before attempting to update.
- **Interrupted Software Updates**: If you canceled an update before it completed, wait at least one minute before attempting your next update.
- **Software Updates via Modem**: If you are updating via the modem, do not change the "connection type" parameter on the 8600 while the modem is connected or attempting to connect.
- **Security Passcode**: An ALL SCREENS (administrator) security passcode is required for upgrading, regardless of whether you are using a Direct, Modem, or Ethernet connection.

- **Passcode Format**: The passcode is case-sensitive. When entering it into Windows' Dial-up Connection dialog box, it must be typed exactly as it was originally entered into the Security screen.

Troubleshooting Connections

- If you get an error message such as "the specified port is not connected" or "There is no answer"…

  You may have the wrong interface type set on your 8600. Navigate to SETUP > NETWORK & REMOTE > NETWORK / SERIAL INTERFACE TYPE and check the interface setting.

  If you are connecting via Direct Serial Connection or modem, review the Properties you have set on that connection. Double-check to ensure that you have set Windows parameters as described in *Appendix: Setting Up Serial Communications* on page 2- 75.

- If your Direct Connect does not work:

  A) Check to make sure that the cables are connected properly.

  B) Check that you are using a null modem cable.

  C) Ensure that the null modem cable is connected to the 8600's serial connector.

- If your Modem Connect does not work:

  A) Ensure that the modem cables and phone lines are connected properly.

  B) Check that you have entered the correct phone number for connection.

  C) Check that you have entered the passcode correctly on the 8600 and the passcode has also been entered correctly on your PC.

  D) Ensure that you enabled the correct PC modem port settings.

  E) Ensure that the external modem attached to your 8600 is set to AUTO ANSWER.

  F) *Make sure that the only "Allowed Network Protocol" is TCP/IP. "NetBUI" and "IPX / SPX Compatible" must not be checked.*

- If you cannot connect to your computer through a crossover Ethernet cable:

  You must set your Windows networking to provide a static IP address for your computer because your Optimod does not contain a DHCP server.

You Cannot Access the Internet After Making a Direct or Modem Connection to the 8600:

If you are connected to the 8600 via modem or direct connect, *you cannot access any other TCP/IP connection*. The PPP connection becomes the default protocol and the default gateway defaults to the 8600 unit's IP address. This means that all existing network connections point to the 8600 unit. To correct this:

A) In Start / Settings / Network and Dialup Connections, open the direct or modem connection you are using to connect to 8600.

B) Select "Properties."

C) Click the tab that reads "Networking."

D) Highlight "Internet protocol (TCP/IP)."

E) Select "Properties."

F) Select "Advanced."

G) Uncheck the "Use default gateway on remote network" box.

H) Select "OK."

If this "Use default gateway on remote network" box is not selected, the gateway will not point to the 8600 unit when you establish a direct or modem connection.

OS-Specific Troubleshooting Advice

Troubleshooting Windows XP Direct Connect:

If you are having trouble establishing a connection, check your New Connection's properties to make sure they are set up correctly:

A) Click "Start / Programs / Accessories / Communications / Network Connections" to bring up the Network Connections screen.

B) In the "Network Connections" window, right-click "Optimod 8600 - Direct" and choose "Properties."

C) The "Properties" window opens for "Optimod 8600 - Direct."

D) Click the "Networking" tab.

E) Set "Type of dial-up server I am calling" to "PPP: Windows 95 / 98 / NT4 / 2000, Internet"

F) Select the "Settings" button and make sure all PPP settings are unchecked, then click "OK."

G) In "This connection uses the following items," uncheck all except for "Internet Protocol (TCP/IP)." You can also leave "QoS Packet Scheduler" checked if you like.

H) In "This connection uses the following items," select "Internet Protocol (TCP/IP)" and then click the "Properties" button. The "Internet Protocol (TCP/IP) Properties" window opens.

I) Choose "Obtain an IP address automatically" and "Obtain DNS server address automatically"

J) Click the "Advanced…" button on the "Internet Protocol (TCP/IP)" Window.

K) In the "Advanced TCP/IP Settings" select the "General" Tab; make sure that no check boxes are checked.

L) Click "OK" to dismiss the "Advanced TCP/IP Settings" window.

M) On the "Properties" window for "Optimod 8600 – Modem" click the "Advanced" tab.

N) Click "OK" to dismiss the window whose name is your new connection.

O) Click "Cancel" to dismiss the "Connect [nnnn]" dialog box

P) Restart your computer.

This resets the serial port and reduces the likelihood that you will encounter problems connecting to the 8600.

Troubleshooting Windows XP Modem Connect:

If you are having trouble establishing a connection, check your New Connection's properties to make sure they are set up correctly.

A) Click "Start / Programs / Accessories / Communications / Network Connections" to bring up the Network Connections screen.

B) In the "Network Connections" window, right-click "Optimod 8600 - Modem" and choose "Properties."

The "Properties" window opens for "Optimod 8600 - Modem."

C) Click the "Networking" tab.

D) Set "Type of dial-up server I am calling" to "PPP: Windows 95 / 98 / NT4 / 2000, Internet"

E) Select the "Settings" button. Make sure all PPP settings are unchecked and then click "OK."

F) In "This connection uses the following items," uncheck all except for "Internet Protocol (TCP/IP)." You can also leave "QoS Packet Scheduler" checked if you like.

G) In "This connection uses the following items," select "Internet Protocol (TCP/IP)" and then click the "Properties" button.

H) The "Internet Protocol (TCP/IP) Properties" window opens.

I) Choose "Obtain an IP address automatically" and "Obtain DNS server address automatically."

J) Click the "Advanced…" button on the "Internet Protocol (TCP/IP)" Window.

K) In the "Advanced TCP/IP Settings," select the "General" Tab; make sure that no check boxes are checked.

L) Click "OK" to dismiss the "Advanced TCP/IP Settings" window.

M) Click "OK" to dismiss the window whose name is your new connection.

N) Restart your computer. (This resets the serial port and reduces the likelihood that you will encounter problems connecting to the 8600.)

Troubleshooting IC Opamps

IC opamps are operated such that the characteristics of their associated circuits are essentially independent of IC characteristics and dependent only on external feedback components. The feedback forces the voltage at the (–) input terminal to be extremely close to the voltage at the (+) input terminal. Therefore, if you measure more than a few millivolts difference between these two terminals, the IC is probably bad.

Exceptions are opamps used without feedback (as comparators) and opamps with outputs that have been saturated due to excessive input voltage because of a defect in an earlier stage. However, if an opamp's (+) input is more positive than its (–) input, yet the output of the IC is sitting at –14 volts, the IC is almost certainly bad.

The same holds true if the above polarities are reversed. Because the characteristics of the 8600's circuitry are essentially independent of IC opamp characteristics, an opamp can usually be replaced without recalibration.

A defective opamp may appear to work, yet have extreme temperature sensitivity. If parameters appear to drift excessively, freeze-spray may aid in diagnosing the problem. Freeze-spray is also invaluable in tracking down intermittent problems. But *use it sparingly,* because it can cause resistive short circuits due to moisture condensation on cold surfaces.

Technical Support

If you require technical support, contact Orban customer service using the information found at http://www.orban.com/contact/. Be prepared to describe the problem accurately. Know the serial number of your 8600 — 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 8600 that we may post from time to time. Manuals (in .pdf form) and 8600 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, 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 fixed quickly 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. In addition, when you return a product to the factory for service, please include a letter describing the problem.

Please refer to the terms of your Limited 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 9″ (H) x 15.5″ (D) x 22″ (W) — 23 cm (H) x 40 cm (D) x 56 cm (W), with a minimum bursting test rating of 200 pounds (91 kg). Place the chassis in a plastic bag (or wrap it in plastic) to protect the finish, then pack it in the carton with at least 1.5 inches (4 cm) of cushioning on all sides of the unit. "Bubble" packing sheets, thick fiber blankets, and the like are acceptable cushioning materials; foam "popcorn" and crumpled newspaper are not. Wrap cushioning materials tightly around the unit and tape them in place to prevent the unit from shifting out of its packing.

Close the carton without sealing it and shake it vigorously. If you can hear or feel the unit move, use more packing. Seal the carton with 3-inch (8 cm) reinforced fiberglass or polyester sealing tape, top and bottom in an "H" pattern. Narrower or parcel post type tapes will not withstand the stresses applied to commercial shipments. Mark the package with the name of the shipper and with these words in red: **DELICATE INSTRUMENT, FRAGILE!**

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

Section 6
Technical Data

Specifications

It is impossible to characterize the listening quality of even the simplest limiter or compressor based on specifications, because such specifications cannot adequately describe the crucial dynamic processes that occur under program conditions. 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

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; Analog Processing Chain): Follows standard 50μs or 75μs preemphasis 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.

Sample Rate: 64 kHz to 512 kHz, depending on processing being performed.

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. It is primarily governed by the dynamic range of the A/D converter, which has a specified overload-to–noise ratio of 110 dB. 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 Separation: > 55 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.

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 8600's HD lowpass filter cutoff frequency to 15 kHz.)

Installation

Delay

Defeatable Analog FM Processing delay: 0.27 to 8.192 seconds for the 8-second board and 12.000 seconds for the 16-second board, adjustable in one-sample increments at 64 kHz sample rate to allow the delays of the analog and digital paths in the HD Radio system to be matched at the receiver output. To avoid introducing artifacts in the analog FM path, its delay is held constant and HD delay is varied if the adjustment does not exceed ±1.3 seconds. Larger adjustments change the analog FM delay and re-center the HD delay within the ±1.3 second window.

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. To achieve these delays, the diversity delay must be defeated on all outputs: analog, digital, IP, and composite; otherwise, the delay will be 1.3 seconds longer.

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 –4.0 to +13.0 dBu (VU).

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. The analog output can emit the analog FM processed signal, the digital radio processed signal, or the low-delay monitor signal. The FM processed signal can be 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).

Crosstalk: <= –70 dB, 20 Hz - 15 kHz.

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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.

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 (+) and 3 electronically balanced, floating and symmetrical.

D/A Conversion: 24 bit 128x oversampled.

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 kHz, 44.1 kHz, 48 kHz, 88.2 kHz, and 96 kHz automatically selected.

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

Input Reference Level: Variable within the range of –30 dBFS to –10 dBFS.

J.17 De-emphasis: Software-selectable.

Filtering: RFI filtered.

Digital Audio Outputs

Configuration: Two outputs, each stereo per the AES3 standard. The outputs can be independently set to emit the analog FM processed signal, the digital radio processed signal, or the low-delay monitor signal. The FM processed signal can be configured in software as flat or pre-emphasized to the chosen processing preemphasis (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. Each output can apply J.17 preemphasis if desired.

Sample Rate: Internal free running at 32 kHz, 44.1 kHz, 48 kHz, 88.2 kHz, or 96 kHz, selected in software. (Use 44.1 kHz or higher for best peak control.) Can also be synced to the AES3 SYNC input or the AES3 digital input at 32 kHz, 44.1 kHz, 48 kHz, 88.2 kHz, and 96 kHz, as configured in software.

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.

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

Output Level (100% peak modulation): –20.0 to 0.0 dBFS software controlled.

Filtering: RFI filtered.

Frequency Response (Digital Audio Output (from Digital Radio Processing Chain)): For output sample rates of 44.1 kHz and above, the frequency response from input to DR-configured output is ±0.10 dB, 2.0 Hz - 20 kHz; flat or with J.17 preemphasis applied. The user may specify lowpass filtering to constrain the bandwidth to 15, 16, 17, 18, or 19 kHz.

Relative Time Delay between FM and HD Outputs: Depends on setting of analog processing channel diversity delay. Once set, this delay is constant regardless of processing preset in use.

Digital Sync Input

Configuration: Used for synchronization of either or both AES3 signals to an external reference provided at this input. Can also be used to receive the output of a ratings encoder Rating Encoder Loop-through Mode. See *Connecting a Ratings Encoder* on page 2-8.

Sampling Rate: 32 kHz, 44.1 kHz, 48 kHz, 88.2 kHz, and 96 kHz, automatically selected.

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

Filtering: RFI filtered.

Wordclock/10 MHz Sync Reference Input (MPX hardware only)

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 singe-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).

Composite Baseband Outputs

Number of Outputs: 2 x analog; 2 x AES3 (MPX hardware only)

Subcarrier Modulation Mode: Standard double sideband mode or compatible single sideband/vestigial sideband mode with SSB/VSB crossover at 150 Hz. SSB/VSB mode adds approximately 12 ms of delay to the composite output signal.

Configuration: Two analog outputs, each with an independent software-controlled output level control, output amplifier and connector.

Source Impedance: 0Ω voltage source or 75Ω, jumper-selectable. Single-ended, floating over chassis ground.

Load Impedance: 37Ω or greater. Termination not required or recommended.

Maximum Output Level: +12.0 dBu (8.72 Vp-p).

Minimum Output Level: –12 dBu (0.55 Vp-p) for 0.5 dB adjustment resolution.

Pilot Level: Adjustable from 6.0% to 12.0%, software controlled.

Pilot Stability: 19 kHz, ±0.5 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 bandwidth, 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, 30 Hz - 15 kHz.

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: 60 dB relative to 9% pilot injection, ±250 Hz (up to 2 dB composite processing drive).

Subcarrier Protection (60-100 kHz): > = 70 dB (referenced to 100% modulation; with up to 2 dB composite limiting drive; measured with 800 line FFT analyzer using "maximum peak hold " display).

57 kHz (RDS/RBDS) Protection: 50 dB relative to 4% subcarrier injection, ±2.0 kHz (up to 2 dB composite processing drive)

ANALOG COMPOSITE OUTPUTS.

Connectors: Two BNC, floating over chassis ground, EMI suppressed.

Maximum Load Capacitance: 0.047µF (0Ω source impedance). Maximum cable length of 100 feet / 30 meters RG–58A/U.

Filtering: RFI filtered.

AES3 COMPOSITE OUTPUTS.

Connectors: One XLR-type, appearing on breakout cable.

Sample Frequency: 192 kHz

Extension to Interim Industry Standard: The 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.

Subcarrier (SCA) Inputs

NON-DIGITIZED INPUTS

Non-Digitized Inputs: 2 x non-digitized analog are summed into the analog composite outputs but do not appear at the digital composite outputs.

SCA Sensitivity (Both Inputs): Variable from 220 mV p-p to > 10 V p-p to produce 10% injection. Sensitivity is screwdriver-adjustable by trim pots that are accessible through holes in the rear panel.

Connectors: Two BNC, unbalanced and floating over chassis ground, EMI suppressed.

19 kHz Pilot Reference: SCA2 input jack can be jumpered to provide a 19 kHz pilot reference output or to serve as a second SCA input. The phase of the reference signal can be set to 0°, 90°, 180°, or 270° with respect to the pilot tone appearing at the composite output. The default is 0°.

DIGITIZED INPUTS (MPX hardware only)

Number of Inputs: 2 x digitized analog (MPX hardware only). These are summed into the digital composite but do not appear on the analog composite outputs.

Resolution: 16-bit conversion

Impedance: > 600Ω

Remote Computer Interface

Supported Computer and Operating System: IBM-compatible PC running Microsoft Windows® XP and higher.

Configuration: TCP/IP protocol via direct cable connect, modem, or Ethernet interface. Suitable null modem cable for direct connect is supplied. Modem and other external equipment is not supplied.

Serial Connectors: RS–232 port (3) DB–9 male, EMI-suppressed. Serial Connector 1 uses PPP to provide for direct or modem connection to the 8600 PC Remote application. Serial Connector 2 supports communication to a computer or remote control system via simple ASCII commands for administration and recalling presets. Serial Connector 3 is reserved for future developments.

Ethernet Connector: Female RJ45 connector for 10-100 Mbps networks using CAT5 cabling. Native rate is 100 Mbps. Provides for connection to the 8600 PC Remote application through either a network, or, using a crossover Ethernet cable, directly to a computer.

Ethernet Networking Standard: TCP/IP.

Remote Control (GPI) Interface

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

Voltage: 6 - 15V AC or DC, momentary or continuous. 9VDC 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 (x2)

Configuration: NPN open-collector

Maximum Voltage: 30 VDC positive. Protected against reverse polarity by a diode; do not exceed 50 mA through this protection diode.

Maximum Current: 30 mA; current must be limited externally.

Power

Voltage: 100–132 VAC or 200–264 VAC, switch-selected on the rear panel, 50–60 Hz, 50 VA. 8600J version for 90-117 VAC.

Connector: IEC, EMI-suppressed. Detachable 3-wire power cord supplied.

Grounding: Circuit ground is independent of chassis ground' can be isolated or connected with a rear panel switch.

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

Physical and Environmental

Dimensions (W x H x D): 19" x 5.25" x 15.5" / 48.3 cm x 8.9 cm x 39.4 cm. Depth shown indicates rack penetration; overall front-to-back depth is 17.75" / 45.1 cm. Three rack units high.

Display: 5.7" / 14.4 cm (diagonal) quarter-VGA color display.

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

Humidity: 0–95% RH, non-condensing.

RFI / EMI: Tested according to Cenelec procedures.

Shipping Weight: 40 lbs. / 18.1 kg

Warranty

Five Years, Parts and Service: Subject to the limitations set forth in Orban's Standard Warranty Agreement (page 1-30).

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

Circuit Description

This section provides a detailed description of user-serviceable circuits used in the 8600. We do not provide detailed descriptions of the digital circuitry because most of this is built with surface-mount components that cannot be removed or replaced with typical tools available in the field. Field repair ordinarily consists of swapping entire PC boards.

The section starts with an overview of the 8600 system, identifying circuit sections and describing their purpose. Then each user-repairable section is treated in detail by first giving an overview of the circuits followed by a component-by-component description.

The drawing on page 6-37 shows circuit board locations.

Overview

The Control Circuits control the DSP, display, and input/output sections of the 8600 system.

The Input Circuits include the connectors and RF filtering for the analog and digital audio inputs, as well as the circuitry to interface these inputs to the digital processing.

The Output Circuits include the connectors and RF filtering for the analog and digital audio outputs and the circuitry to interface the digital processing to these outputs.

The DSP Circuits implement the bypass, test tone, and audio processing using digital signal processing.

The Power Supply provides power for all 8600 circuit sections.

A block diagram of the DSP signal processing appears on page 6-83.

Control Circuits

The control circuit is based on an AMD Elan SC520 microprocessor, which is a 586-class processor running an Orban executable program over a third-party real-time operating system. A flash memory emulates a hard drive. The memory is non-volatile and does not rely on a battery to retain information when mains power is off.

The flash memory holds the operating system, the Orban executable program, and all preset files, both factory and user. It also contains a write-protected "boot segment" that functions as a boot ROM.

The control circuits process and execute user-initiated requests to the system. The source of these requests is the front panel buttons, joymouse, and rotary encoder, the rear panel RS-232 ports, Ethernet port, and the optically isolated General Purpose Interface. These changes affect hardware function and/or DSP processing. The control circuits also send information to the LCD display.

The control circuit communicates with the DSP and display circuitry through the SC520's general-purpose bus.

The SC520 periodically refreshes a watchdog timer. If the timer times out without being refreshed, it assumes that the control program has crashed and automatically reboots the SC520. The DSP chips will continue to process audio until the time comes

in the boot process to reload DSP program code into them. At this point, the audio will mute for about 2 seconds until the DSP code download has finished. If you hear a 2-second audio mute on air, you can assume that the 8600 has rebooted for some reason. Be prepared to convey this fact to Orban customer service if you call for technical assistance.

The control board is divided into two assemblies: a "base board," which has interface circuitry, and a CPU board, which plugs into the base board and which contains the CPU, the Ethernet interface chip, the flash memory, the DRAM, and the real-time clock, which keeps time for the 8600's automation functions. The real-time clock is backed up by a DL2032 battery so that it keeps accurate time even when the 8600 is powered down. The battery is socketed and can be readily accessed by removing the 8600's top cover; the battery is located on the foil (top) side of the CPU board.

User Control Interface and LCD Display Circuits

The user control interface enables the user to control the 8600's functionality. A rear panel GPI connector allows optically isolated remote control of certain functions, such as recalling presets, via contact closure. An Ethernet port and three RS-232 serial connectors allow you to connect a modem or computer to the 8600. Front panel pushbutton switches select between various operational modes and functions. A rotary encoder allows the user to adjust parameters and enter data.

1. Remote Interface and RS-232 Interfaces

Located on base board

A remote interface connector and circuitry implements remote control of certain operating modes; OPTIMOD-FM 8600 has eight remote contact closure inputs.

A valid remote signal is a momentary pulse of current flowing through remote signal pins. Current must flow consistently for 50msec for the signal to be interpreted as valid. Generally, the 8600 will respond to the most recent control operation, regardless of whether it came from the front panel, remote interface, or RS-232.

Component-Level Description:

After being current limited by resistors, the GPI control signals are applied to two quad optoisolators, U10, 12, and then to the control circuitry.

Octal driver U1 buffers the RS-232 port, which is located on a small daughter board.

U10, 12 and U1 are socketed for easy field replacement in the event of overload, lightning damage, etc. All other circuitry is surface-mount and is not field-repairable.

2. Color LCD Display

The 5.7" color LCD is an active-matrix quarter-VGA panel. The CPU addresses its controller chip through the CPU's ISA bus.

The backlight on the display has a finite lifetime (normally a few years of continuous operation). Therefore, the 8600 always implements a screen-saver timeout. This is not user-adjustable.

Input Circuits

This circuitry interfaces the analog and digital inputs to the DSP. The analog input stages scale and buffer the input audio level to match it to the analog-to-digital (A/D) converter. The A/D converts the analog input audio to digital audio. The digital input receiver accepts AES3-format digital audio signals from the digital input connector and sample rate-converts them as necessary. The digital audio from the A/D and SRC is transmitted to the DSP.

1. Analog Input Stages

Located on Input/output board

The RF-filtered left and right analog input signals are each applied to a floating, balanced amplifier. The amplifier's output feeds a circuit that scales, balances, and DC-biases the signal. This circuit feeds an RC low-pass filter that applies the balanced signal to the analog-to-digital (A / D) converter.

Note that the small RFI "tee" filter assemblies connected to the input and output connectors are socketed and user-replaceable.

Component-Level Description:

The left channel balanced audio input signal is applied to the filter / load network made up of L100-103 and associated resistors and capacitors. (There are solder pads available in the PC board to accept an optional 600Ω termination load [R106] on the input signal if the user wishes to install one.) A conventional three-opamp instrumentation amplifier (IC100 and associated circuitry) receives the input signal. R110-114 and quad analog switch IC101 make up the circuit that sets the gain of IC100. The switches in IC101 set the gain of the instrumentation amplifier by switching resistors in parallel with R104. (Smaller total resistances produce larger gains.)

IC100 feeds IC104 and associated components. This stage balances, DC-biases, and scales the signal to the proper level for the analog-to-digital (A / D) converter IC107. IC105A and associated components comprise a servo amp to correctly DC-bias the signal feeding the A/D converter. R137-139, C109, C110 make an attenuator / RC filter necessary to filter high frequency energy that would otherwise cause aliasing distortion in the A/D converter.

The corresponding right channel circuitry is functionally identical to that just described.

IC100, 101, 102, 103 are socketed for easy field replacement. All other circuitry

is surface-mounted and is not field-replaceable.

2. Stereo Analog-to-Digital (A / D) Converter

Located on Input/output board

The A/D converter, IC107, is a stereo 24-bit sigma-delta converter. (This is a surface-mount part and is not field-replaceable,)

The A/D oversamples the audio, applies noise shaping, and filters and decimates to 64 kHz sample rate, which is the fundamental sample rate in the 8600.

3. Digital Input Receiver and Sample Rate Converter (SRC)

Located on Input/output board

The integrated receiver and input sample rate converter, IC500, accepts digital audio signals using the AES3 interface format (AES3-1992). The built-in sample rate converter (SRC) accepts and sample-rate converts any of the "standard" 32 kHz, 44.1 kHz, 48 kHz, 88.2 kHz, and 96 kHz rates in addition to any digital audio sample rate within the range of 32 kHz and 96 kHz. The SRC converts the input sample rate to 64 kHz, which is the 8600's fundamental sample rate.

This chip is surface-mounted and not field-replaceable.

Output Circuits

This circuitry interfaces the DSP to the analog and digital audio outputs. The digital audio from the DSP is transmitted to the digital-to-analog converter (D / A) and output sample rate converter (SRC). The digital-to-analog (D / A) converter converts the digital audio words generated by the DSP to analog audio. High-speed D/A converters do the same for the composite outputs, each of whose outputs is smoothed by a passive LC reconstruction filter. The analog output stages scale and buffer the D/A output signal to drive the analog output XLR connectors with a low impedance balanced output. The digital output transmitter accepts the digital audio words from the output sample rate converter (SRC) and transmits them in AES3-format digital audio signals on the digital output connector.

1. Stereo Digital-to-Analog (D / A) Converter

Located on Input/output board

The D / A, IC211, is a stereo, 24-bit delta-sigma converter. It receives the serial left and right audio data samples from the DSP at 64 kHz sample rate and converts them into audio signals requiring further, relatively undemanding analog filtering. IC211 is surface-mounted and is not field-replaceable.

2. Analog Output Stages

Located on Input/output board

The left and right analog signals emerging from IC211 are each filtered, amplified, and applied to a floating-balanced integrated line driver, which has a 50Ω output impedance. The line driver outputs are applied to the RF-filtered left and

right analog output connectors. These analog signals can represent either the transmitter or monitor output of audio processing.

Component-Level Description:

IC201 and associated components filter the left channel signal emerging from IC211. The purpose of these stages is to reduce the out-of-band noise energy resulting from the delta-sigma D / A's noise-shaping filter and to translate the differential output of the D/A converter into single-ended form. These components apply a 3rd order low-pass filter to the differential signal from the D / A. This filter does not induce significant overshoot of the processed audio, which would otherwise waste modulation.

IC203 is used to set the analog output level. It is a digitally controlled gain block that sets its gain according to signals on its three digital input lines.

IC204B and associated components form a low-frequency servo amplifier to remove residual DC from the signal. The 0.15Hz −3 dB frequency prevents tilt-induced overshoot in the processed audio.

IC204A buffers the output of IC203 and implements de-emphasis if desired. FET switches Q200 and Q201 implement 75μs and 50μs de-emphasis respectively. This analog de-emphasis rolls off any digital noise produced by earlier circuitry and also helps implement independent de-emphasis settings between the analog and digital outputs.

The buffered and optionally de-emphasized output of IC204 is applied to IC207, a balanced output line driver. This driver emulates a floating transformer; its differential output level is independent of whether one side of its output is floating or grounded. IC207 and its right channel counterpart IC208 are socketed for easy field replacement. All other circuitry is surface-mounted.

The corresponding right channel circuitry is functionally identical to that just described.

3. Digital Sample Rate Converter (SRC) and Output Transmitter

Located on Input/output board

An integrated output sample rate converter (SRC) and AES3 line driver chip, IC502, converts the 64 kHz 8600 system sample rate to any of the standard 32 kHz, 44.1 kHz, 48 kHz, 88.2 kHz, and 96 kHz rates, and also contains a digital audio interface transmitter to encode digital audio signals using the AES3 interface format (AES3-1992). This chip is surface-mounted and is not field-replaceable.

4. Composite Output Circuit

Located on the Input/output board

A composite D/A converter and reconstruction filter drive two digitally controlled attenuators that permit the levels of the two composite outputs to be set independently. The SCA inputs are summed with the composite output before the digitally controlled attenuators, so the attenuator adjusts the level of the entire composite signal, attenuating the SCA and stereo signals to the same extent.

The second SCA input can be jumpered to serve as a pilot reference source for RDS generators, which is how it is shipped from the factory.

Component-Level Description:

We will describe composite output #1. IC300 is a high-speed D/A converter chip that receives the digital composite signal at a 512 kHz sample rate. It drives buffer amplifier IC308A. IC308A drives a fifth-order passive LC reconstruction filter C336-C339, L300-L301, R301-303. (This filter is equalized and phase-corrected in DSP to obtain excellent flatness and phase-linearity. This achieves high stereo separation.)

IC302A buffers the output of the anti-imaging filter. IC302B is a servo amplifier to remove DC offset at the output of IC302A. IC401B accepts the SCA inputs, summing them with the composite stereo output of IC302A. Any contribution from the SCA inputs is therefore are not indicated on the COMPOSITE LEVEL meter displayed by the 8600, because this meter indicates only the composite signal generated by the DSP.

Digitally controlled attenuator IC402B receives the output of IC401B and sets the composite output level. IC3B and IC1, a composite high-current buffer amplifier, receives the output of IC402B and drives the composite output connector J4B through an RFI attenuator network and optional 75Ω build-out resistor R411.

The pilot reference D/A converter IC400 receives serial data from the DSP circuitry. After being buffered and low-pass filtered by IC401A, the resulting 19 kHz sine wave signal can be connected to J5A through jumper J400.

The composite line driver amplifiers are socketed for easy field replacement; all other components are surface-mounted and are not field-replaceable.

DSP Circuit

The DSP circuit consists of nine Freescale (formerly Motorola) 250 MHz DSPB56724 dual-core 24-bit fixed-point DSP chips that execute DSP software code to implement digital signal processing algorithms. Sufficient external memory is installed to implement a diversity delay at 64 kHz sample rate. The MX peak limiter also uses the external memory for various purposes.

The algorithms filter, compress, and limit the audio signal. The nine DSP chips, each operating at approximately 500 million instructions per second (MIPS), for a total of 4500 MIPS, provide the necessary signal processing. A sampling rate of 64 kHz and power-of-two multiples thereof, up to 512 kHz, is used.

System initialization normally occurs when power is first applied to the 8600 and can occur abnormally if the 8600's watchdog timer forces the SC520 to reboot. Upon initialization, the SC520 CPU downloads the DSP executable code stored in the flash memory. The time between application of power and completion of DSP code download is approximately 7 seconds. Once a DSP chip begins executing its program, execution is continuous. The SC520 provides the DSP program with parameter data

(representing information like the settings of various processing controls), and extracts the front panel metering data from the DSP chips.

During system initialization, the SC520 queries the DSP hardware about its operational status and will display an error message on-screen if the DSP fails to initialize normally. Please note any such messages and be ready to report them to Orban Customer Service.

The DSP chips are located on the DSP board—see the drawings starting on page 6-60. IC801 is a local switching-type voltage regulator on the DSP board that derives the +3.3 V supply for the DSP chips from the +RAW unregulated system voltage, while IC802 creates a +1.2 V regulated supply from +RAW.

AES Output L/R Channel Reversal Bug

In the year 2011, some 8600s were shipped with a firmware bug on the DSP board that caused a left/right audio channel reversal and a one-sample delay difference between the left and right audio channels. This bug affects only the two digital outputs, not the analog L/R or composite outputs.

If your unit has the bug, you can fix it permanently by performing the following procedure:

A) Upgrade to version 1.2.1 or higher software, following the instructions under the heading "Upgrading Your 8600 Software." This appears in the 8600_x.x.x_Readme.htm file that is copied to your computer when you run the Setup8600_X.X.X.X_PC_Remote.exe file. 8600_x.x.x_Readme.htm contains the latest version of the instructions below.

B) Use the 8600's built-in test tone generator in SYSTEM SETUP>TEST to determine if your unit has the bug. (Refer to Test Modes on page 3-85 for more information on the tone generator.)

   Set the test mode to TONE and the tone generator's TONE MOD TYPE to LEFT. Observe whether this tone appears on the left or right audio channel of whichever AES output is most conveniently accessible. (If one output has the channels swapped, then both outputs do.) If and only if the tone appears on the right channel, complete the procedure described in steps (C) through (E) below.

C) Disconnect AC power from your Optimod and then reapply power. Within two seconds after the "Optimizing…" screen appears, press the ESCAPE button. The Choose If TDF Modifications Are Installed screen will appear.

D) Using the joystick, LOCATE to the *Cancel* button and then press the *Enter* button. The AES 1 and 2: L/R Output Swap screen appears.

E) Using the joystick, LOCATE to the *Change State* button and then press the *Enter* button. This will correct the problem. Your Optimod will then automatically reboot into its normal operating mode.

   The correction that you just implemented is permanently recorded in the EEPROM on your Optimod's control board. Only the procedure described in steps (C) through (E) above can change it. The correction will survive

future software upgrades and will also remain active even if you reset your Optimod to its factory defaults by following the procedure described in *If you have forgotten your "All Screens" passcode* starting on page 2-47.

The guiding principle is that the correction resides in permanent memory on the control board but it corrects a bug in the DSP board. Therefore, if you break this pairing in the future (i.e., if either board is replaced for service), you may have to repeat steps (C) through (E) above. It is wise to repeat step (B) above if you have replaced either board. (We expect that all service replacement DSP boards provided by us in the future will not have the bug.)

Power Supply

Warning! Hazardous voltages are present in the power supply when it is connected to the AC line.

The power supply converts an AC line voltage input to various power sources used by the 8600. To ensure lowest possible noise, four linear regulators provide ±15VDC and ±5VDC for the analog circuits. A switching regulator provides high current +5VDC for the digital circuits. An unregulated voltage feeds local regulators.

The power supply circuits are straightforward and no explanation is required beyond the schematic itself. Be aware that C1, C4, C5, and C12 in the switching regulator are premium-quality low-ESR capacitors and must be replaced with equivalent types to ensure proper operation of the switching supply.

The output of the power supply is monitored by the power-indicator LED circuit, which causes the power LED to flash according to a preset code to diagnose problems with the various power supplies in the 8600. See step (2.B) on page 4-9.

Abbreviations

Some of the abbreviations used in this manual may not be familiar to all readers:

A/D (or A to D)	analog-to-digital converter
AES	Audio Engineering Society
AGC	automatic gain control
A-I	analog input
A-O	analog output
BAL	balanced (refers to an audio connection with two active conductors and one shield surrounding them).
BBC	British Broadcasting Corporation
BNC	a type of RF connector
CALIB	calibrate
CIT	composite isolation transformer
CMOS	complementary metal-oxide semiconductor
COFDM	Coded Orthogonal Frequency Division Multiplex—a robust type of digital modulation using many narrow-bandwidth, low data rate, mutually non-interfering carriers to achieve an ag-

	gregate high data rate with excellent multipath rejection.
COM	serial data communications port
D/A (or D to A)	digital-to-analog converter
dBm	decibel power measurement. 0 dBm = 1mW applied to a specified load. In audio, the load is usually 600Ω. In this case only, 0 dBm = 0.775V rms.
dBu	decibel voltage measurement. 0 dBu = 0.775V RMS. For this application, the dBm-into-600Ω scale on voltmeters can be read as if it were calibrated in dBu.
DI	digital input
DJ	disk jockey, an announcer who plays records in a club or on the air
DO	digital output
DOS	Microsoft disk operating system for IBM-compatible PC
DSP	digital signal processor (or processing). May also refer to a special type of microprocessor optimized for efficiently executing arithmetic.
EBU	European Broadcasting Union
EBS	Emergency Broadcasting System (U.S.A.)
EMI	electromagnetic interference
ESC	escape
FCC	Federal Communications Commission (USA regulatory agency)
FDNR	frequency-dependent negative resistor—an element used in RC-active filters
FET	field effect transistor
FFT	fast Fourier transform
FIFO	first-in, first-out
G/R	gain reduction
HD Radio	*See IBOC*
HF	high-frequency
HP	high-pass
IBOC	"In-Band On-Channel"—a form of digital radio commercialized by iBiquity Corporation where the digital carriers use a form of COFDM modulation and share the frequency allocation of the analog carriers. Also known by its trademarked name of "HD Radio."
IC	integrated circuit
IM	intermodulation (or "intermodulation distortion")
I/O	input/output
ITU	International Telecommunications Union (formerly CCIR). ITU-R is the arm of the ITU dedicated to radio.
JFET	junction field effect transistor
LC	inductor / capacitor
LCD	liquid crystal display
LED	light-emitting diode
LF	low-frequency
LP	low-pass
LVL	level
MHF	midrange / high-frequency
MLF	midrange / low-frequency
MOD	modulation
N&D	noise and distortion
N / C	no connection
OSHOOT	overshoot
PC	IBM-compatible personal computer
PCM	pulse code modulation
PPM	peak program meter
RAM	random-access memory
RC	resistor / capacitor
RDS/RBDS	Radio (Broadcasting) Data Service—a narrowband digital subcarrier centered at 57 kHz in the FM baseband that usually provides program or network-related data to the consumer in

	the form of text that is displayed on the radio. Occupied bandwidth is ±2500 Hz.
REF	reference
RF	radio frequency
RFI	radio-frequency interference
RMS	root-mean-square
ROM	read-only memory
SC	subcarrier
SCA	subsidiary communications authorization — a non program-related subcarrier in the FM baseband above 23 kHz (monophonic) or 57 kHz (stereophonic)
S / PDIF	Sony / Philips digital interface (standardized as IEC958)
TRS	tip-ring-sleeve (2-circuit phone jack)
THD	total harmonic distortion
TX	transmitter
μs	Microseconds. For FM preemphasis, the +3 dB frequency is 1 / (2 π τ), where τ is the preemphasis time constant, measured in seconds.
VCA	voltage-controlled amplifier
VU	volume unit (meter)
XLR	a common style of 3-conductor audio connector
XTAL	crystal

Parts List

Many parts used in the 8600 are surface-mount devices (“SMT”) and are not intended for field replacement because specialized equipment and skills are necessary to remove and replace them. The list below includes substantially all of the parts used in the 8600 (including surface-mount devices) and inclusion of a part in this list does not imply that the part is field-replaceable.

See the following assembly drawings for locations of components.

Obtaining Spare Parts

Special or subtle characteristics of certain components are exploited to produce an elegant design at a reasonable cost. It is therefore unwise to make substitutions for listed parts. Consult the factory if the listing of a part includes the note “selected” or “realignment required.”

Orban normally maintains an inventory of tested, exact replacement parts that can be supplied quickly at nominal cost. Standardized spare parts kits are also available. When ordering parts from the factory, please have available the following information about the parts you want:

Orban part number
Reference designator (e.g., C3, R78, IC14)
Brief description of part
Model, serial, and “M” (if any) number of unit — see rear-panel label

To facilitate future maintenance, parts for this unit have been chosen from the catalogs of well-known manufacturers whenever possible. Most of these manufacturers have extensive worldwide distribution and can be contacted through their web sites.

Base Board

BASE BOARD		
PART #	***DESCRIPTION***	***COMPONENT IDENTIFIER***
42008.020Q	SUBASSEMBLY FLAT CABLE-40P- 2""	J7
16013.000.01	HEATSINK, CLIP ON TO 220	H1
20040.604.01	RESISTOR, METALFIM, 1/8W, 1%, 604 Ω	R28, 30, 33, 35, 37, 39, 44, 46, 48, 49, 50, 51, 52, 53, 54, 55
20080.301.01	RESISTOR, METALFIM, 1/2W, 1%, 301 Ω	R47
20121.100.01	RESISTOR, METALFIM, 1/8W, 1%, 10Ω, 1206	R43, 45
20121.750.01	RESISTOR, THIN FILM, 1/8W, 1%, 75 Ω	R82, 83, 84
20128.002.01	RESISTOR 2.0 Ω 1% 0805	R22, R23, R24, R25
20129.301.01	RESISTOR, 301Ω, 0805	R59, R77
20130.100.01	RESISTOR, 1.00K 1% 0805	R79
20130.162.01	RESISTOR, 1/8W, 1%, 1.62K, 0805	R41, 42
20130.200.01	RESISTOR, 2.00K, 0805	R4, R56, R62
20130.249.01	RESISTOR, 1/8W, 1%, 2.49K, 0805	R76
20130.562.01	RESISTOR, 1/8W, 1%, 5.62K, 0805	R57
20131.100.01	RESISTOR, 10K, 0805	R5, 6, 15, 16, 17, 26, 60, 61, 63, 65, 67, 68, 69, 70, 71, 73, 74, 75, 80, 81, 102, 103, 104
20131.140.01	RESISTOR, 14.0K, 0805	R58, 64
20131.301.01	RESISTOR, 30.1K, 0805	R72
20132.100.01	RESISTOR, 100K, 0805	R1, 2, 3, 7, 8, 9, 10, 11, 12, 13, 14, 20, 27, 29, 31, 32, 34, 36, 38, 40, 66, 85, 86, 87, 88, 89, 90, 91, 92, 93
20132.332.01	RESISTOR, 332K, 0805	R78
21139.000.01	CAPACITOR, X7R, 0.1UF, 10%, 0805	C3, 6, 7, 8, 9, 10, 11, 12, 13, 18, 21, 24, 30, 32, 33, 34, 35, 38, 39, 43
21147.022.01	CAPACITOR, 22pf, 0805, 1%	C40, 41
21319.610.01	CAPACITOR, 10uf, TANT, SMT	C1, 4, 14, 15, 17, 19, 22, 36, 37
21322.547.01	CAPACITOR, 4.7uf, TANT, 6032B	C2, 5, 20, 23
22016.000.01	DIODE, MMSZ5231B, SOD-123	D12
22083.015.01	DIODE, VOLTAGE SUPPRESSOR, 15 VOLT	D11
22101.001.01	DIODE, 1N4148WT/R	D1, 3, 4, 5, 6, 9, 10
22209.000.01	DIODE, SCHOTTKY 1A, 60V, SMD	REF, , NO, STUFF, D7, D8
23214.000.01	TRANSISTOR NPN MMBT3904	Q1, Q3, Q4, Q5
23606.201.01	TRANSISTOR, PWR, NPN	Q2

BASE BOARD		
PART #	***DESCRIPTION***	***COMPONENT IDENTIFIER***
24635.000.01	IC 74HCT374	U4
24900.000.01	IC, HEX INVERTER, SMT	U11, U13
24967.000.01	IC, 74ACT245DW	U3, U5
24978.000.01	IC, 74ACT244SC	U14, 15
24979.000.01	IC, BAT54C-7	D13, 14, 15, 16, 17
24982.000.01	IC, 74HC4051M	U19
24983.000.01	IC, 7064STC100-10	U1
24984.000.01	IC, LP2987IM-5.0	U20
25008.000.01	IC, PS2506-4	U10, 12
25112.001.01	LED, RED/GREEN, BI-CLR/POLR	
27017.025.01	CONNECTOR, RIGHT ANGLE, PC BOARD MOUNT, 25P	J10
27147.018.01	IC, SOCKET, DIP, 18 PIN, DUAL	SU18
27223.002.01	CABLE, FLAT, 2 LONG, 14 CONDUCTOR	J8
27371.040.01	CONNECTOR HEADER PC104 STACK 40P	HEADER2
27371.064.01	CONNECTOR HEADER PC104 STACK 64P	HEADER1, HEADER3
27406.014.01	CONNECTOR, SOCKET, STRIP, 14 PIN	J2
27421.004.01	CONNECTOR, HEADER, DOUBLE ROW, 4P, 2 X 2	J3B, J6
27421.006.01	CONNECTOR, HEADER, DOUBLE ROW, 6P, 2 X 3	J5
27421.010.01	CONNECTOR, HEADER, DOUBLE ROW, 23", 2 X 5	J12
27421.050.01	CONNECTOR HEADER STR .23 2x25	J9
27426.003.01	CONNECTOR, HEADER, 3 PIN, SINGLE RW	J11
27451.005.01	CONNECTOR, STR, DBL ROW, 26 PIN	J4
27451.024.01	HEADER, STR, DBLROW, PCMOUNT	J1
27500.000.01	CONNECTOR MOL53047-0510 5PIN	J14
28086.000.01	CRYSTAL, 4.0 MHz, HC49US	X1
29521.000.01	INDUCTOR, 3.9UH, JM391K	L1, L2, L3
32166.000.06	CIRCUIT BOARD, BASE BOARD	
44093.100.01	SOFTWARE PIC 8300 U18	U18
47010.016.01	SUBASSEMBLY RECPTL-W/SHRINK	J3A
47010.017.01	SUBASSEMBLY RECPTL-W/SHRINK	J3A

CPU board

CPU BOARD		
PART #	***DESCRIPTION***	***COMPONENT, IDENTIFIER***
20128.010.01	RESISTOR, 10Ω,0805	R31, R34
20128.022.01	RESISTOR, 22Ω 1% 0805	R5, R6
20128.332.01	RESISTOR, 33.2Ω,0805	R10, R11, R14
20128.499.01	RESISTOR, 49.9Ω 1% 0805	R19, R20, R21, R22, R23
20129.160.01	RESISTOR, 160Ω 1% 0805	R24, R25
20129.330.01	RESISTOR, 330Ω 1% 0805	R12, R16
20129.470.01	RESISTOR, 470Ω 1% 0805	R13, R15
20130.100.01	RESISTOR, 1.00K 1% 0805	R17, R35
20130.475.01	RESISTOR, 4.75KΩ,0805	R3, R4, R7, R8, R26, R27, R28, R29, R30, R32
20130.931.01	RESISTOR, 9.31KΩ, 1%, 0805	R33
20131.100.01	RESISTOR, 10KΩ,0805	R1, R2, R9
20131.147.01	RESISTOR, 1 / 8W,1%,14.7KΩ,0805	R18
20233.102.01	RESISTOR NETWORK 1K CTS745C 8R BUSSED	RN1
20233.472.01	RESISTOR NETWORK 4.7K CTS745C 8R BUSS	RN2, RN3, RN4
20237.472.01	RESISTOR NETWORK 8R, ISO, 5%	RN5
21139.000.01	CAPACITOR, X7R,0.1uF,10%,0805	C8, C9, C20, C21, C177, C179, C182
21141.000.01	CAPACITOR, NPO,1000pF,1%,0805	C10
21142.000.01	CAPACITOR, NPO,100pF,1%,0805	C2
21146.310.01	CAPACITOR, .01uF,0805,10%	C11, 126, 127, 133, 134, 150, 152, 154, 156, 158,160, 162, 180
21167.047.01	CAPACITOR, 4.7pF 50V X7R 0805	C1
21170.018.01	CAPACITOR, 18pF 1% 50V COG 0805	C3, C4, C5, C6, C7
21171.105.01	CAPACITOR, 1uF X7R 0805	C14, 17, 125, 132, 151, 153, 155, 157, 159, 161, 175, 176, 178, 181, 183
21322.547.01	CAPACITOR, 4.7uF,TANT,6032B	C12
21325.610.01	CAPACITOR, 10uF 10% TANT 6032-B	C13, C15, C16, C18
22101.001.01	DIODE,1N4148WT / R	D1, D2, D3
24331.025.01	IC VOLTAGE REGULATOR LT1963-2.5 SOT223	U14
24331.033.01	IC VOLTAGE REGULATOR LT1963-3.3 SOT223	U15
24541.000.01	IC SDRAM MT48LC16 TSOP54P	U2, U3
24542.000.01	IC FLASH MEMORY E28F128 TSOP56	U4

CPU BOARD		
PART #	***DESCRIPTION***	***COMPONENT, IDENTIFIER***
24543.000.01	IC CY2305 0DLYBuF 8P	U11
24544.000.01	IC NM93C46 SEEPROM TSSOP	U12
24653.000.01	IC PWRST MIC8114 SOT143	U5
24670.000.01	IC 10 / 100BT NIC NATIONAL	U10
24965.000.01	IC,74ALVC164245DGG	U7, U8, U9
24972.520.01	IC MICROPROCESSOR ELANSC520 BGA388	U1
27306.000.01	CONN RJ45 PCMT W / MAGS	J1
27370.040.01	CONN SCKT PC104 40PIN	P2
27370.064.01	CONN SCKT PC104 64PIN	P1, P3
28031.000.01	HOLDER,BATTERY,LITH CELL	BT1HLDR
28041.000.01	CELL,COIN,BATTERY,LITH,3V	BT1
28089.000.01	OSC 33MHZ SG636 4P SMD	X1
28090.000.01	IC TCXO DS32KHZ 36P BGA	U13
28091.000.01	CRYSTAL 25MHZ RXD MP35L SMD	Y1
32200.000.02	CPU BOARD ASSEMBLY DRAWING	
32201.000.02	PCB CPU BOARD 8600	
44094.100.01	FIRMWARE 8600 U6 20LV8D	
62200.000.02	SCHEMATIC, CPU BOARD 8600	

RS-232 Board

RS232 BOARD		
PART #	***DESCRIPTION***	***COMPONENT IDENTIFIER***
20128.000.01	RESISTOR,0Ω,0805	R8, R9, R10, R11, R12, R13, R14, R15, R16, R17, R18, R19, (NO STUFF)
20129.110.01	RESISTOR 110Ω 0805 1%	R5
20131.100.01	RESISTOR,10K,0805	R6, R7
20132.100.01	RESISTOR,100K,0805	R1, R2, R3, R4
21139.000.01	CAPACITOR,X7R,0.1UF,10%, 0805	C1, C2, C3, C4, C5, C6, C7, C8, C9, C10, C11, C12, C13, C14, C15, C17, C18, C19, C20, C22
21147.033.01	CAPACITOR 33pf 0805	C16
21319.610.01	CAPACITOR,10uf,TANT,SMT	C21
24966.000.01	IC,ST16C554DCJ68	U4
24968.000.01	IC,MAX208ECNG	U1, U2, U3
27017.009.01	CONNECTOR,RT AGL,PC MNT,9P	J1, J2, J3
27183.024.01	CONNECTOR SOCKET DIP 24-PIN SMT	SU1, SU2, SU3
27489.050.01	CONNECTOR STACKER 50 PIN	J4
29521.000.01	INDUCTOR,3.9UH,JM391K	L1

Power Supply

POWER SUPPLY		
PART #	***DESCRIPTION***	***COMPONENT IDENTIFIER***
10012.404.01	SCREW MS SEM P / P 4-40 X 1 / 4	
15025.000.01	TRANSISTOR, MOUNTING KIT, TO 220	HW1, HW2, HW3, HW4, HW5
15061.005.01	LED MOUNT, 1 POSITION, 0.240" HIGH	H1, H2, H3, H4
20020.025.01	RESISTOR, 1 / 4W, 0Ω, (JUMPER)	R1
21129.410.01	CAPACITOR, AXIAL LEADS, 0.1uF, 50V, 20%	C6, C10, C11, C12, C15, C19, C20, C21
21227.710.01	CAPACITOR, RADIAL LEADS 100uF 16V HFS	C1
21227.747.01	CAPACITOR, RADIAL LEADS 470uF 16V HFS	C4, C5
21230.710.01	CAPACITOR, RADIAL LEADS 100uF 50V HFS	C22
21255.000.01	CAPACITOR, SNAP-IN, 6800uF, 16V, 20%	C13, C14
21256.000.01	CAPACITOR, RADIAL LEADS, 1000uF, 35V, 20%	C17, C18
21263.710.01	CAPACITOR, RADIAL LEADS, 100uF, 25V, 10%	C2, C3, C8, C9
21307.522.01	CAPACITOR, RADIAL LEADS, 2.2uF, 35V, 10%	C7, C16
22004.056.01	ZENER-DIODE-1W-5%-5.6V-1N	CR19, CR20
22015.000.01	DIODE-SHOTTKY RECTIFIER-SBL	CR21, CR22, CR23
22083.022.01	DIODE, VOLTAGE SUPPRESSOR, 22 VOLT	CR2, CR13, CR14
22083.033.01	DIODE, VOLTAGE SUPPRESSOR, 33 VOLT	CR9, CR10
22083.068.01	DIODE, VOLTAGE SUPPRESSOR, 6.8 VOLT	CR4, CR17, CR18
22201.400.01	DIODE, RECTIFIER IN4004 PRV400V	CR5, CR6, CR7, CR8, CR11, CR12, CR15, CR16
22208.040.01	DIODE, SHOTTKY-31DQ04-3.3	CR3
22500.271.01	ZENER, TRANSORB, VARISTOR	V1, V2
24303.901.01	IC, LINEAR, DC REGULATOR, 15V NEG	U2
24304.901.01	IC, REGULATOR	U1
24307.901.01	IC, LINEAR, DC REGULATOR, 5V POS	U3
24308.901.01	IC, LINEAR, DC REGULATOR, 5V NEG	U4
24323.000.01	IC, SIMPLE SWITCH, 0 TO 220	U5
26143.000.01	SWITCH, SLIDE, VOLT, 115 / 230	SW1
26146.000.01	SWITCH, SLIDE, SPDT, VERTICAL MOUNT	SW2

POWER SUPPLY		
PART #	***DESCRIPTION***	***COMPONENT IDENTIFIER***
27060.000.01	CONNECTOR, VERTICAL HEADER	J1
27421.010.01	CONNECTOR, HEADER, DOUBLE ROW , 23", 2 X 5	J7
27426.003.01	CONNECTOR, HEADER, 3-PIN, SINGLE ROW	J6 (OPTIONAL FAN CONNECTOR)
27451.003.01	HEADER, STR, DOUBLE ROW, PCMOUNT	J3
27451.004.01	HEADER, STR, DOUBLE ROW, PCMOUNT	J4
27451.024.01	HEADER, STR, DOUBLE ROW, PCMOUNT	J5
27493.000.01	CONNECTOR, VERTICAL, HEADER, 6 POS.	J2
27711.206.01	TERM, CRIMP, RING, INSULATED, 6R	LUG
28004.150.01	FUSE, 3AG, SLOBLO,1/2 AMP	F1
28112.003.01	KNOB-FUSE-DOM-GRY-FOR 281	H7
28112.005.01	BODY-FUSEHOLDER-PC MNT	H6
29262.000.01	LINE FILTER, PC MOUNT, 1A	A1
29519.000.01	INDUCTOR-TORODIAL- 7.7UH	L2
29526.000.01	INDUCTOR, PE92108K	L1
50286.000.02	HEATBAR POWER SPLY 8600	HS1

Input/Output (I/O) Board

INPUT/OUTPUT BOARD		
PART #	***DESCRIPTION***	***COMPONENT, IDENTIFIER***
20039.750.01	RESISTOR, METALFILM, 1/8W, 1%, 75.0Ω	R411, 420
20040.604.01	RESISTOR, METALFILM, 1/8W, 1%, 604Ω	R106, 119, 412, 421
20041.100.01	RESISTOR, METALFILM, 1/8W, 1%, 1.00 kΩ	R100, 107, 115, 120, 400, 401
20058.187.01	RESISTOR, METALFILM, 1/8W, 0.1%, 1.87K	R301
20058.205.01	RESISTOR, 1/8W.1%, 2.05K	R302
20121.100.01	RESISTOR, METALFILM, 1/8W, 1%, 10Ω, 1206	R154, 200, 232, 531, 540, 703
20121.750.01	RESISTOR, THIN FILM, 1/8W, 1%, 75Ω	R158, 416, 530, 604, 605, 606
20122.110.01	RESISTOR, THIN FILM, 1/8W, 1%, 110Ω	R238, 330, 500, 514, 517, 532, 538, 541, 543, 544
20123.100.01	RESISTOR, THIN FILM, 1/8W, 1%, 1k	R304, 600, 601, 602, 603, 701
20123.150.01	RES-TF-1.8W-1%-SMT 1	R131, 134, 140, 141, 144, 146
20123.499.01	RESISTOR, THIN FILM, 1/8W, 1%, 4.99K	R101, 103, 105, 108, 116, 118, 121, 124

INPUT/OUTPUT BOARD		
PART #	***DESCRIPTION***	***COMPONENT, IDENTIFIER***
20124.100.01	RESISTOR THIN FILM 1/8W 1% 1206 10K	R110, 125, 237, 243, 244, 406, 407, 409, 413, 414, 700, 704, 705
20124.200.01	RESISTOR, THIN FILM, 1/8W, 1%, 20.0K	R402, 404, 417, 418
20126.100.01	RESISTOR, METALFILM, 1/8W, 1%, 1.00M	R142, 152, 225, 231, 306
20128.000.01	RESISTOR, 0Ω, 0805	R608, 609, 610
20129.150.01	RESISTOR, 1/8W, 1%, 150Ω, 0805	R138, 151, 235, 236
20129.249.01	RESISTOR, 1/8W, 1%, 249Ω, 0805	R137, 139, 149, 150, 155
20129.768.01	RESISTOR, 1/8W, 1%, 768Ω, 0805	R111, 126, 403, 404
20130.162.01	RESISTOR, 1/8W, 1%, 1.62KΩ, 0805	R132, 153, 156, 157, 502, 515, 534, 539
20130.210.01	RESISTOR, 1/8W, 1%, 2.10KΩ, 0805	R112, 127
20130.249.01	RESISTOR, 1/8W, 1%, 2.49KΩ, 0805	R300
20130.348.01	RESISTOR, 1/8W, 1%, 3.48KΩ, 0805	R204, 210, 217, 220
20130.562.01	RESISTOR, 1/8W, 1%, 5.62KΩ, 0805	R113, R128
20130.845.01	RESISTOR, 1/8W, 1%, 8.45KΩ, 0805	R201, 202, 205, 207, 208, 211, 212, 214, 215, 218, 305
20131.113.01	RESISTOR, 1/8W, 1%, 11.3KΩ, 0805	R206, 219, 233, 234
20131.143.01	RESISTOR, 1/8W, 1%, 14.3KΩ, 0805	R221, 224, 227, 230
20131.147.01	RESISTOR, 1/8W, 1%, 14.7KΩ, 0805	R114, 129
20131.200.01	RESISTOR, 20.0K 1% 0805	R402, 410, 417, 418, 419
20131.249.01	RESISTOR 1% 24.9K 0805	R203, 209, 213, 216, 405
20131.499.01	RESISTOR, 1/8W, 1%, 49.9KΩ, 0805	R222, 223, 228, 229, 239, 240, 241, 242, 501, 504, 513, 524, 526, 533, 535, 536, 537, 542, 702
20131.825.01	RESISTOR, 1/8W, 1%, 82.5KΩ, 0805	R104, 123, 303, 408, 415
20132.154.01	RESISTOR, 1/8W, 1%, 154KΩ, 0805	R328
20151.365.01	RESISTOR, 0.1% 3.65KΩ, 0805	R130, 133, 135, 136, 143, 145, 147, 148
20151.536.01	RESISTOR, 0.1%, 5.36KΩ, 0805	R102, 109, 117, 122
20511.310.01	TRIMPOTS, 10KΩ, 20%, TOP ADJ	VR200, 201, 400, 401
21112.210.01	CAPACITOR, CER, .001UF, 1KV, 10%	C100, 102, 104, 106
21123.510.01	CAPACITOR, RADIAL LEADS, 1.0UF, 50V, 20%	C224, 230
21137.447.01	CAPACITOR .47UF 25V 10% 1206	C113, 117, 340

		INPUT/OUTPUT BOARD
PART #	***DESCRIPTION***	***COMPONENT, IDENTIFIER***
21138.247.01	CAPACITOR, SMD1206, 4700PF, 50V, 5%	C109, 110, 115, 116, 411, 412, 512, 518, 526, 531
21139.000.01	CAPACITOR, X7R, 0.1UF, 10%, 0805	C111, 118, 119, 120, 121, 123, 124, 125, 126, 127, 128, 202, 203, 211, 212, 214, 215, 233, 302, 303, 306, 309, 404, 407, 410, 500, 501, 502, 510, 513, 522, 523, 527, 528, 532, 600, 601, 602, 603, 604, 605, 606, 607, 608, 609, 610, 612, 613, 616, 617, 618, 619, 620, 621, 622, 623, 624, 625, 626, 628, 629, 632, 633, 634, 635, 636, 637, 638, 639, 640, 641, 642, 643, 644, 648, 649, 650, 651, 656, 657, 658
21140.000.01	CAPACITOR, NPO, 470PF, 1%, 0805	C217, 218, 219, 220, 336, 339
21141.000.01	CAPACITOR, NPO, 1000PF, 1%, 0805	C226, 228, 337, 517, 521, 524, 529, 652, 653, 654, 655
21142.000.01	CAPACITOR, NPO, 100PF, 1%, 0805	C338
21143.000.01	CAPACITOR, NPO, 1500PF, 1%, 0805	C221, 222, 225, 227, 408
21144.000.01	CAPACITOR, 5%, 100V, 47PF, 1206	C101, 103, 105, 107, 108, 114, 333
21145.000.01	CAPACITOR, NPO, 5%, 100V, 33PF-1206	C1, 2, 231, 335, 351, 406, 533, 534
21154.433.01	CAPACITOR, .33uf, 0805, 10%	C503, C511, C525, C530
21156.020.01	CAPACITOR, 12pf, 1206	C223, 229, 334, 402
21263.710.01	CAPACITOR, RADIAL LEADS, 100uF, 25V, 10%	C304, 305, 308
21318.510.01	CAPACITOR, TANTALUM, 1.0uF, 35V, B-CASE	C200, 201, 232, 700, 701
21319.610.01	CAPACITOR, 10uf, TANTALUM, SMT	C112, 122, 129, 130, 131, 210, 213, 216, 300, 301, 307, 310, 403, 405, 409, 646, 647
22101.001.01	DIODE, 1N4148WT/R	CR101, 102, 106, 107
22102.001.01	DIODE, SIGNAL, 1N5711TR	CR700
22106.000.01	DIODE, SMCJ26C, TRANZORB	CR100, 103, 104, 105, 202, 203, 204, 205
23415.000.01	TRANSISTOR, JFET SST113 SMT	Q200, 201, 202, 203
24024.000.01	IC, OPA2134PA	IC3, 100, 102
24025.000.01	IC, BUF634P, DIP8	IC1, IC2
24538.000.01	IC PCM1744 D/A SOIC14	IC400
24634.000.01	IC, OCTAL 3 STATE NONINVR	IC505, 506, 705
24728.302.01	IC, QUAD, SPST SW, DIP/16	IC101, 103
24748.000.01	IC, LM339M S014	IC210
24752.000.01	IC DIGIPOT DS1267 SO1C16	IC402
24759.000.01	IC SMPL RT CONV CS8420 D1	IC500, 502, 507, 508
24857.000.01	IC 74HC374 DLATCH SOL20	IC108, 209, 706
24858.000.01	IC, SO/14, SMT	IC604

INPUT/OUTPUT BOARD		
PART #	***DESCRIPTION***	***COMPONENT, IDENTIFIER***
24900.000.01	IC, HEX INVERTER, SMT	IC603, 707
24924.000.01	IC CSS3310KS	IC203
24951.000.01	IC HC151 8CH MUX SOIC16	IC708, 709
24957.000.01	IC, PCM1704U	IC300
24958.000.01	IC, DRV134PA-DIP	IC207, IC208
24960.000.01	IC, OPA2134UA	IC104, 105, 106, 201, 202, 204, 206, 302, 401
24961.000.01	IC, OPA627AP	IC308
24963.000.01	IC, 5383 VS	IC107
24970.000.01	IC, PIC16C 67-20L	IC700
24980.000.01	IC, 74ACT32D	IC702, 703, 704
24992.000.01	IC, 74AHCT244 SOIC	IC605, 606
24997.000.01	IC, DAC AK4393 SSOP28	IC211
27053.003.01	CONNECTOR, MALE, INSERT, RIGHT-ANGLE	J201, 202, 502, 504
27054.003.01	CONNECTOR, FEM, INSERT, RIGHT-ANGLE	J100, 103, 500, 503
27055.000.01	CONNECTOR DUAL BNC	J4, J5
27147.008.01	IC, SOCKET, DIP, 8 PINS, DUAL	IC1, IC2, IC3, 100, 102, 207, 208
27147.016.01	IC, SOCKET, DIP, 16 PIN, DUAL	IC101, 103
27174.044.01	IC, SOCKET, 44 PIN, LOW PROFILE	IC700
27401.000.01	CONNECTOR, JUMPER, RECPT, BLACK	J2, J3, J400
27406.014.01	CONNECTOR, SOCKET, STRIP, 14 PIN	JP600
27408.003.01	CONNECTOR, 3P SOCKET STRIP	L1, L2, L3, 100, 102, 104, 106, 200, 201, 202, 203, 400, 401, 402
27421.004.01	CONNECTOR, HEADER, DOUBLE ROW, 4P, 2 X 2	J2, J3, 400
27426.005.01	HEADER, UNSHRD	J700, NO, STUFF
27451.004.01	HEADER, STR, DBLROW, PCMOUNT	J601
27451.007.01	CONNECTOR, DOUBLE ROW, PC-MOUNT, 40 PIN	J602, 603
27630.001.01	JUMPER, PC-MOUNT, TEST POINT	TP600, 607
29015.000.01	AES3 TRANFORMER, SURFACE-MOUNT	T500, 502, 503, 504
29506.001.01	BEAD- FERRITE- ON WIRE	L500, 501, 504, 505, 506, 507, 508, 509
29508.210.01	FILTER-EMI SUPPRESSION-50V-	L1, L2, L3, L100, 102, 104, 106, 200, 201, 202, 203, 400, 401, 402
29521.000.01	IND, 3.9UH, JM391K	L5, L6, L7, 204, 205, 206, 207, 403, 404, 405
29522.000.01	IND, 1200UH, 5%, 1-M-10-22	L101, 103, 105, 107
29532.156.01	IND 560UH 10% 370MA	L600, 601

INPUT/OUTPUT BOARD		
PART #	***DESCRIPTION***	***COMPONENT, IDENTIFIER***
29707.002.01	INDUCTOR, SELECTED, 3.501mh	L300
29707.003.01	INDUCTOR, SELECTED, 3.39mh	L301
21174.000.01	CAPACITOR .047 1206 X7R	C400, 401

DSP Board

DSP BOARD		
PART #	***DESCRIPTION***	***COMPONENT IDENTIFIER***
44129.100.01	FIRMWARE FPGA J703 86xx DSP	IN, CIRCUIT, PROG, -, IC703
27401.000.01	CONNECTOR JUMPER RECPT BLACK	J705, (PIN, 1-2)
27468.006.01	CONNECTOR 6P MOLEX PCMNT	J901
24943.000.01	IC 74VHC08	IC602
27630.001.01	JUMPER PC-MOUNT TEST PT	TP801, TP805, TP901, TP902, (NO, STUFF, TP802, TP803, TP804)
20135.002.01	RESISTOR 5% 2Ω 0805	R803, R804
20129.110.01	RESISTOR 0805 110Ω 1% 1/8W	R817, R818
20237.472.01	RESISTOR NETWORK 8R ISO 5%	RN401, RN402, RN907, RN908, RN909
20131.825.01	RESISTOR 1/8W 1% 82.5K 0805	R909, R910, R911, R912
20130.845.01	RESISTOR 1/8W 1% 8.45K 0805	R901, R902, R903, R904, R905, R906, R907, R908
20130.348.01	RESISTOR 1/8W 1% 3.48K 0805	R815, R819, R820
20130.162.01	RESISTOR 1/8W 1% 1.62K 0805	R812
20130.100.01	RESISTOR 1.00K 1% 0805	R711, R816
20129.150.01	RESISTOR 1/8W 1% 150Ω 0805	R712
29552.000.01	INDUCTOR POWER 1.5uH 9A SMD	L801, L802
20130.499.01	RESISTOR R0805 4.99K 1% 1/8W	R210, R705, R813, R814, R821
29537.102.01	INDUCTOR 0805 FERRITE 1000Ω	L701, L703, L704
20132.100.01	RESISTOR 1/8W 1% 100K R0805	R706, R707, R708, R709, R710
20131.100.01	RESISTOR 10K 0805	R201, R202, R203, R204, R205, R206, R207, R208, R209, R603, R604, R806, R807, R808, R809, R810, R811
20128.249.01	RESISTOR 24.9K 1% 0805	R805
20128.075.01	RESISTOR 75Ω 1% 0805	R601, R602
20128.022.01	RESISTOR 22Ω 1% 0805	R701, R702, R703, R704
27451.002.01	HEADER STR DBL ROW PCMT	J704
27426.003.01	CONNECTOR HEADER 3PIN SINGLE ROW	J705
27451.007.01	CONNECTOR DOUBLE ROW PC-MOUNT 40 PIN	J601
27451.003.01	HEADER STR ` PC-MOUNT	J801
27421.010.01	CONNECTOR HEADER DOUBLE ROW 23" 2 X 5	J700, J703, J802
42008.020.03	FLAT CABLE ASSEMBLY 40P 2"	J701, J702
29512.000.01	CHK SHIELD 1670-1 250uH	L100
24333.000.01	IC ST23-LM4041 ADJ	IC803
24997.000.01	IC DAC AK4393VSP VSOP	IC903
24960.000.01	IC OPA2134UA	IC902

		DSP BOARD
PART #	***DESCRIPTION***	***COMPONENT IDENTIFIER***
24843.000.01	IC CD74HC4046AM 16-SOIC	IC704
24348.000.01	IC REG SYNC BUCK CLK 4A	IC801, IC802
24795.000.01	IC XC95144XL-10TQG144C	IC703
24948.000.01	IC 74LVC2244 DRVR20 SOIC	IC706, IC707, IC708
28093.000.01	OSCILLATOR VCXO 3.3V 27MHz FVXO-	IC701
24766.000.01	IC PLL1707DBQ	IC702
24421.000.01	IC SN74ALVCH16373DGGR	IC411, IC412, IC917, IC918, IC919
24622.000.01	IC 74AHCT04 INVERT SOIC	IC601, IC705
24945.000.01	IC 74AHC541 OCTLBUF SOL20	IC604
24794.000.01	IC DSP DSPB56724AG	IC201, IC202, IC203, IC204, IC205, IC206, IC207, IC208, IC209
24793.000.01	IC MT48LC2M32B2P-6	IC401, IC402, IC907, IC908, IC909
24649.000.01	IC SN74AC11D AND-GATE	IC301, IC302, IC603
22104.000.01	DIODE SCHOT RECT 1N5818	CR802
22083.068.01	DIODE VLTG SUPRSR 6.8 VL	CR801
22083.033.01	DIODE VLTG SUPRSR 33 VLT	CR803
21319.610.01	CAPACITOR 10uf TANTALUM SMT	C831, C832
21230.710.01	CAPACITOR RAD LDS 100UF 50V HFS	C833
21227.747.01	CAPACITOR RAD LDS 470UF 16V HF	C836
21140.000.01	CAPACITOR NPO 470PF 1% 0805	C911, C912
21139.101.01	CAPACITOR CER 100PF 10% 50V0805	C821, C822, C823, C824, C825, C826
21325.622.01	CAPACITOR 22UF 10% TANTALUM 3528	C801, C802, C803, C804
21185.220.01	CAPACITOR 22uF 6.3V X5R 20% 080	C815, C816
21325.610.01	CAPACITOR 10UF 10% TANTALUM 3528	C700, C805, C806, C807, C808
21138.247.01	CAPACITOR SMD1206 4700PF 50V 5%	C829, C830
21139.000.01	CAPACITOR X7R 0.1UF 10% 0805	C501, C502, C503, C504, C505, C506, C507, C508, C509, C511, C512, C513, C514, C515, C516, C517, C518, C519, C521, C522, C523, C524, C525, C526, C527, C528, C529, C531, C532, C533, C534, C535, C536, C537, C538, C539, C541, C542, C543, C544, C545, C546, C547, C548, C549, C551, C552, C553, C554, C555, C556, C557, C558, C559, C561, C562, C563, C564, C565, C566, C567, C568, C569, C571, C572, C573, C574, C575, C576, C577, C578, C579, C604, C703, C704, C705, C706, C707, C708, C709, C710, C711, C712, C713, C714, C715, C817, C818, C819, C820, C834, C835, C904, C905, C906, C913, C914
21171.105.01	CAPACITOR 1uf X7R 0805	C441, C442, C451, C452, C461, C462, C471, C472, C581, C582, C583, C584, C585, C586, C587, C588, C589, C591, C592, C593, C594, C595, C596, C597, C598, C599, C701, C702, C809, C810, C811, C812, C813, C814, C901, C902, C903, C947, C948, C949,

DSP BOARD		
PART #	***DESCRIPTION***	***COMPONENT IDENTIFIER***
		C957, C958, C959, C967, C968, C696, C977, C978, C979
21146.310.01	CAPACITOR 0.01uf 0805 10% 0805	C101, C102, C103, C104, C105, C106, C107, C108, C109, C111, C112, C113, C114, C115, C116, C117, C118, C119, C201, C202, C203, C204, C205, C206, C207, C208, C209, C211, C212, C213, C214, C215, C216, C217, C218, C219, C221, C222, C223, C224, C225, C226, C227, C228, C229, C231, C232, C233, C234, C235, C236, C237, C238, C239, C241, C242, C243, C244, C245, C246, C247, C248, C249, C251, C252, C253, C254, C255, C256, C257, C258, C259, C261, C262, C263, C264, C265, C266, C267, C268, C269, C271, C272, C273, C274, C275, C276, C277, C278, C279, C281, C282, C283, C284, C285, C286, C287, C288, C289, C291, C292, C293, C294, C295, C296, C297, C298, C299, C301, C302, C401, C402, C411, C412, C421, C422, C431, C432, C601, C602, C603, C716, C717, C718, C719, C720, C721, C722, C723, C724, C725, C726, C827, C828, C907, C908, C909, C917, C918, C919, C927, C928, C929, C937, C938, C939

Interface Board

INTERFACE BOARD		
PART #	***DESCRIPTION***	***COMPONENT IDENTIFIER***
20128.000.01	RESISTOR, 0Ω, 0805	R102, R103, R104, R105, R116, R117, R128, R129, R130, R131, R138, R139, R140, R209
20128.010.01	RESISTOR, 10Ω, 0805	R121
20130.100.01	RESISTOR, 1.00K 1% 0805	R107, R126, R127, R165, R166, R167, R202
20131.100.01	RESISTOR, 10KΩ, 0805	R133, R134, R135, R136, R137, R203, R204, R205, R206, R207
20132.100.01	RESISTOR, 100KΩ, 0805	R118, R119, R120, R141, R142, R143, R144, R145, R146, R147, R148, R149, R150, R151, R152, R153, R154, R155, R156, R157, R158, R159, R160, R161, R162, R163, R164, R201, R208
21139.000.01	CAPACITOR, X7R, 0.1UF, 10%, 0805	C100, C101, C102, C203, C204, C205, C206, C207, C208, C212, C213, C214, C215, C216, C217, C218, C219, C223, C224
21151.020.01	CAPACITOR, 20pf, 0805	C221, C222

INTERFACE BOARD		
PART #	***DESCRIPTION***	***COMPONENT IDENTIFIER***
21171.105.01	CAPACITOR 1uf X7R 0805	C202
21319.610.01	CAPACITOR, 10uf, TANTALUM, SMT	C200, C201
23214.000.01	TRANSISTOR NPN MMBT3904	Q100
24747.000.01	IC, MAX6501, TEMP 65c	IC201
24756.000.01	IC, GRAPHICS, S1D13706F00A	IC104
24965.000.01	IC, 74ALVC164245DGG	IC100, IC101, IC102
24994.000.01	IC, 74ACT04, SOIC 14P	IC103
27183.018.01	CONNECTOR SOCKET DIP 18-PIN SMT	IC202
27373.040.01	CONNECTOR SOCKET PC104 STACKING	J100
27373.064.01	CONNECTOR SOCKET PC104 STACKING	J101, J102
27414.016.01	HEADER PIN 2X8 RIGHT-ANGLE	J200
27756.000.01	CONNECTOR, HOUSING, RT AGL SMT	J105
27757.033.01	CONNECTOR, FPC, 33PIN ROT-LOCK	J103
28089.000.01	OSCILLATOR 33MHZ SG636PCE 4P SMD	Y100
28092.000.01	CRYSTAL, 4.000MHZ, SMD	Y200
32256.000.02	PCB INTERFACE 8600 DEP/DE	
44105.100.01	FIRMWARE PIC DISP INT 8600	IC202

Headphone Board

HEADPHONE BOARD		
PART #	***DESCRIPTION***	***COMPONENT IDENTIFIER***
20039.100.01	RESISTOR, MF, 1/8W, 1%, 10 Ω	R2, R3, R8
20131.100.01	RESISTOR, 10K, 0805	R1, R6
20131.499.01	RESISTOR, 1/8W, 1%, 49.9K, 0805	R10, R13
20543.103.01	TRIMPOTS, AUDIO, 10K, TAPER A	R12
21139.000.01	CAPACITOR, X7R, 0.1UF, 10%, 0805	C5, C6, C7, C8, C9, C10, C11, C12
21142.000.01	CAPACITOR, NPO, 100PF, 1%, 0805	C2, C4
21445.510.01	CAPACITOR, METALIZED POLYESTER, 1.0UF, 50V, 5%	C1, C3
24025.000.01	IC, BUF634P, DIP8	U2, U3, U4
24960.000.01	IC, OPA2134UA	U1
27090.000.01	JACK, SOCKET, 1/4" PHONE, RIGHT ANGLE	J1
27147.008.01	IC, SOCKET, DIP, 8 PINS,	SU2, SU3, SU4

HEADPHONE BOARD		
PART #	***DESCRIPTION***	***COMPONENT IDENTIFIER***
	DUAL	
27408.003.01	CONNECTOR, 3P SOCKET STRIP	SL1, SL2, SL3
27758.006.01	CONNECTOR 6P MOLEX RIGHT ANGLE	J2
29508.210.01	FLTR-EMI SUPPRESSION-50V-	L1, L2, L3
32091.000.05	CIRCUIT BOARD HEADPHONE BOARD	
57148.000.01	BRACKET, PHONE 8600 TYPE	

Encoder Board

ENCODER BOARD		
PART #	***DESCRIPTION***	***COMPONENT IDENTIFIER***
15061.003.01	LED MOUNT, T-1, 0.220	LEDMNT
21129.410.01	CAPACITOR, AXL LDS, 0.1uF, 50V, 20%	C1
25122.000.01	LED, T 1 3/4 RED/GRN	LED, 1
26088.000.01	ENCODER, ROTARY, NOBLE 8600	EN1
26128.000.01	SWITCH, JOYMOUSE, PUSH, 8600	A1
26304.001.01	SWITCH, PUSH-MOM, SPST	SW1, SW2
27421.016.01	CONNECTOR, HEADER, STR, .23", 2 X 8	JP1
32101.000.03	CIRCUIT BOARD ENCODER DPL/DEL	

LCD Carrier Board

LCD CARRIER BOARD		
PART #	***DESCRIPTION***	***COMPONENT IDENTIFIER***
11568.000.01	SCREW, THREAD-FORM 3x6MM	
21112.282.01	CAPACITOR, CERAMIC, 0.0082UF, 1KV, 10%	C1
22106.000.01	DIODE, SMCJ26C, TRANZORB	D3
22209.000.01	DIODE, SHOT 1A, 60V, SMD	D1, D2
24758.000.01	LCD BACKLIGHT DRVR/CCFT	A1
25409.000.01	LCD DISPLAY 320 X 240	LCD1
27420.002.01	CONNECTOR 2 PIN RT ANGLE	J1
27757.033.01	CONNECTOR, FPC, 33PIN ROT-LOCK	J2, J3
32271.000.02	CIRCUIT BOARD, LCD	

		LCD CARRIER BOARD
PART #	***DESCRIPTION***	***COMPONENT IDENTIFIER***
	CARRIER 8600	

Digital MPX (Retrofit for pre-MPX 8600s)

		DIGITAL MPX (Retrofit for pre-MPX units)
PART #	***DESCRIPTION***	***COMPONENT IDENTIFIER***
27401.000.01	CONNECTOR JUMPER RECPT BLACK	J105p2-3, J106p2-3, J202p2-3, J203p2-3
27408.003.01	CONNECTOR 3PIN SOCKET STRIP	S-L200, S-L201
24517.000.01	IC ICS570B	IC105, IC106
24795.288.01	IC XC95288XL-10TQG144C	IC102
27353.015.01	CONNECTOR DB15 HD RA BRD LCK	J107
24524.000.01	IC ADC 16BIT ADS1601IPFBT	IC202, IC203
24523.000.01	IC dsPIC33EP512GP806-I-PT	IC201
20130.400.01	RESISTOR 402 OHM 1%1/8W 0805	R200
20132.619.01	RESISTOR 1/8W 1% 619 ohm 0805	R202, R203
27451.002.01	HEADER STR DBL ROW PC-MOUNT	J104
27473.000.01	CONNECTOR HEADER DUAL ROW 2X7 2mm	J102
24711.000.01	IC LMV7219M5 COMPARATOR	IC107
27426.005.01	HEADER UNSHRD	J101, J201
27426.003.01	CONNECTOR HEADER 3PIN SINGLE ROW	J105, J106, J202, J203
21325.610.01	CAPACITOR 10UF 10% TANTALUM 3528	C222, C223, C224, C225, C226, C227, C228, C229
24772.000.01	IC PIC18LF4420 MICRO CTRL	IC101
27630.001.01	JUMPER PC MOUNT TEST PT	TP200
29539.000.01	CHOKE COMMON MODE 1K OHM SM	L101, L102, L204, L205
21138.247.01	CAPACITOR SMD1206 4700PF 50V 5%	C202, C203
21139.000.01	CAPACITOR X7R 0.1UF 10% 0805	C102, C103, C104, C113, C114, C115, C116, C117, C118, C119, C120, C121, C122, C230, C231, C232, C233, C234, C235, C236, C237, C238, C239, C240, C241, C242, C243, C244, C245, C246, C247, C248, C249, C250, C251, C252, C253, C254, C255, C256, C257, C258, C259, C260, C261, C262, C263, C264, C265, C266, C267
29015.000.01	AES3 TRANSFORMER	T101
20129.110.01	RESISTOR 0805 110 OHM 1% 1/8W	R112
20129.475.01	RESISTOR 475 OHM 1% 1/8W	R201

		DIGITAL MPX (Retrofit for pre-MPX units)
PART #	***DESCRIPTION***	***COMPONENT IDENTIFIER***
	0805	
20128.000.01	RESISTOR 0 OHM 0805	R111
29508.210.01	FILTER EMI SUPPRESS 100V 10	L200, L201
29506.001.01	BEAD FERRITE ON WIRE	L103, L104, L105, L106, L206, L207, L208, L209
24960.000.01	IC OPA2134UA	IC204, IC205
24672.000.01	IC DIGITAL INTERFACE TRANSMIT	IC103
21171.105.01	CAPACITOR 1uf X7R 0805	C100, C101, C112, C201, C204, C205, C206, C207, C208, C209, C210, C211, C212, C213, C214, C215, C216, C217, C218, C219, C220, C221
21146.310.01	CAPACITOR 0.01uf 0805 10% 0805	C105, C106, C107, C108, C109, C110, C111
21142.000.01	CAPACITOR NPO 100PF 1% 0805	C268, C269, C270, C271, C272, C273
20511.310.01	TRIMPOT 10K 10% 3/8TOP ADJ	VR202, VR203
20133.100.01	RESISTOR 1/8W 1% 1.0M 0805	R119
20132.100.01	RESISTOR 1/8W 1% 100K R0805	R101, R115, R116, R117, R118, R204, R205
20131.100.01	RESISTOR 10K 0805	R103, R104, R105, R106, R107, R108, R109, R110, R210, R211, R212, R213, R214, R215, R216, R217, R218, R219, R220, R221
20130.100.01	RESISTOR 1.00K 1% 0805	R102, R222
20129.150.01	RESISTOR 1/8W 1% 150 OHM 0805	R113, R114
20128.499.01	RESISTOR 49.9 OHM 1% 0805	R206, R207, R208, R209
24753.000.01	IC AD1895 SRC 192KHZ	IC104
29535.000.01	INDUCTOR 3.9uH CHIP 1008	L202, L203
22210.000.01	DIODE MBR530 SOD123	CR101, CR102, CR201, CR202

Digital MPX (for 8600MPX Units)

		DIGITAL MPX (for 8600MPX units)
PART #	***DESCRIPTION***	***COMPONENT IDENTIFIER***
27058.000.01	CONNECTOR BNC RIGHT ANGLE 75 OHM	J108, J204, J205
27053.003.01	CONNECTOR MALE INSERT RIGHT ANGLE	J109
24517.000.01	IC ICS570B	IC105, IC106
24524.000.01	IC ADC 16BIT ADS1601IPFBT	IC202, IC203
24523.000.01	IC dsPIC33EP512GP806-I-PT	IC201
24795.288.01	IC XC95288XL-10TQG144C	IC102

		DIGITAL MPX (for 8600MPX units)
PART #	***DESCRIPTION***	***COMPONENT IDENTIFIER***
20130.400.01	RESISTOR 402 OHM 1%1/8W 0805	R200
20132.619.01	RESISTOR 1/8W 1% 619 ohm 0805	R202, R203
27451.002.01	HEADER STR DBL ROW PC-MOUNT	J104
27473.000.01	CONNECTOR HEADER DUAL ROW 2X7 2mm	J102
24711.000.01	IC LMV7219M5 COMPARATOR	IC107
27426.005.01	HEADER UNSHRD	J101, J201
27426.003.01	CONNECTOR HEADER 3PIN SINGLE ROW	J105, J106, J202, J203
21325.610.01	CAPACITOR 10UF 10% TANTALUM 3528	C222, C223, C224, C225, C226, C227, C228, C229
24772.000.01	IC PIC18LF4420 MICRO CTRL	IC101
27401.000.01	CONNECTOR JUMPER RECPT BLACK	J105p2-3, J106p2-3, J202p2-3, J203p2-3
27630.001.01	JUMPER PC MOUNT TEST PT	TP200
29539.000.01	CHOKE COMMON MODE 1K OHM SM	L101, L102, L204, L205
27408.003.01	CONNECTOR 3PIN SOCKET STRIP	S-L200, S-L201
21138.247.01	CAPACITOR SMD1206 4700PF 50V 5%	C202, C203
21139.000.01	CAPACITOR X7R 0.1UF 10% 0805	C102, C103, C104, C113, C114, C115, C116, C117, C118, C119, C120, C121, C122, C230, C231, C232, C233, C234, C235, C236, C237, C238, C239, C240, C241, C242, C243, C244, C245, C246, C247, C248, C249, C250, C251, C252, C253, C254, C255, C256, C257, C258, C259, C260, C261, C262, C263, C264, C265, C266, C267
29015.000.01	AES3 TRANSFORMER	T101
20129.110.01	RESISTOR 0805 110 OHM 1% 1/8W	R112
20129.475.01	RESISTOR 475 OHM 1% 1/8W 0805	R201
20128.000.01	RESISTOR 0 OHM 0805	R111
29508.210.01	FILTER EMI SUPPRESS 100V 10	L200, L201
29506.001.01	BEAD FERRITE ON WIRE	L103, L104, L105, L106, L206, L207, L208, L209
24960.000.01	IC OPA2134UA	IC204, IC205
24672.000.01	IC DIGITAL INTERFACE TRANSMIT	IC103
21171.105.01	CAPACITOR 1uf X7R 0805	C100, C101, C112, C201, C204, C205, C206, C207, C208, C209, C210, C211, C212, C213, C214, C215, C216, C217, C218, C219, C220, C221

		DIGITAL MPX (for 8600MPX units)
PART #	***DESCRIPTION***	***COMPONENT IDENTIFIER***
21146.310.01	CAPACITOR 0.01uf 0805 10% 0805	C105, C106, C107, C108, C109, C110, C111
21142.000.01	CAPACITOR NPO 100PF 1% 0805	C268, C269, C270, C271, C272, C273
20511.310.01	TRIMPOT 10K 10% 3/8TOP ADJ	VR202, VR203
20133.100.01	RESISTOR 1/8W 1% 1.0M 0805	R119
20132.100.01	RESISTOR 1/8W 1% 100K R0805	R101, R115, R116, R117, R118, R204, R205
20131.100.01	RESISTOR 10K 0805	R103, R104, R105, R106, R107, R108, R109, R110, R210, R211, R212, R213, R214, R215, R216, R217, R218, R219, R220, R221
20130.100.01	RESISTOR 1.00K 1% 0805	R102, R222
20129.150.01	RESISTOR 1/8W 1% 150 OHM 0805	R113, R114
20128.499.01	RESISTOR 49.9 OHM 1% 0805	R206, R207, R208, R209
24753.000.01	IC AD1895 SRC 192KHZ	IC104
29535.000.01	INDUCTOR 3.9uH CHIP 1008	L202, L203
22210.000.01	DIODE MBR530 SOD123	CR101, CR102, CR201, CR202

Schematics and Parts Locator Drawings

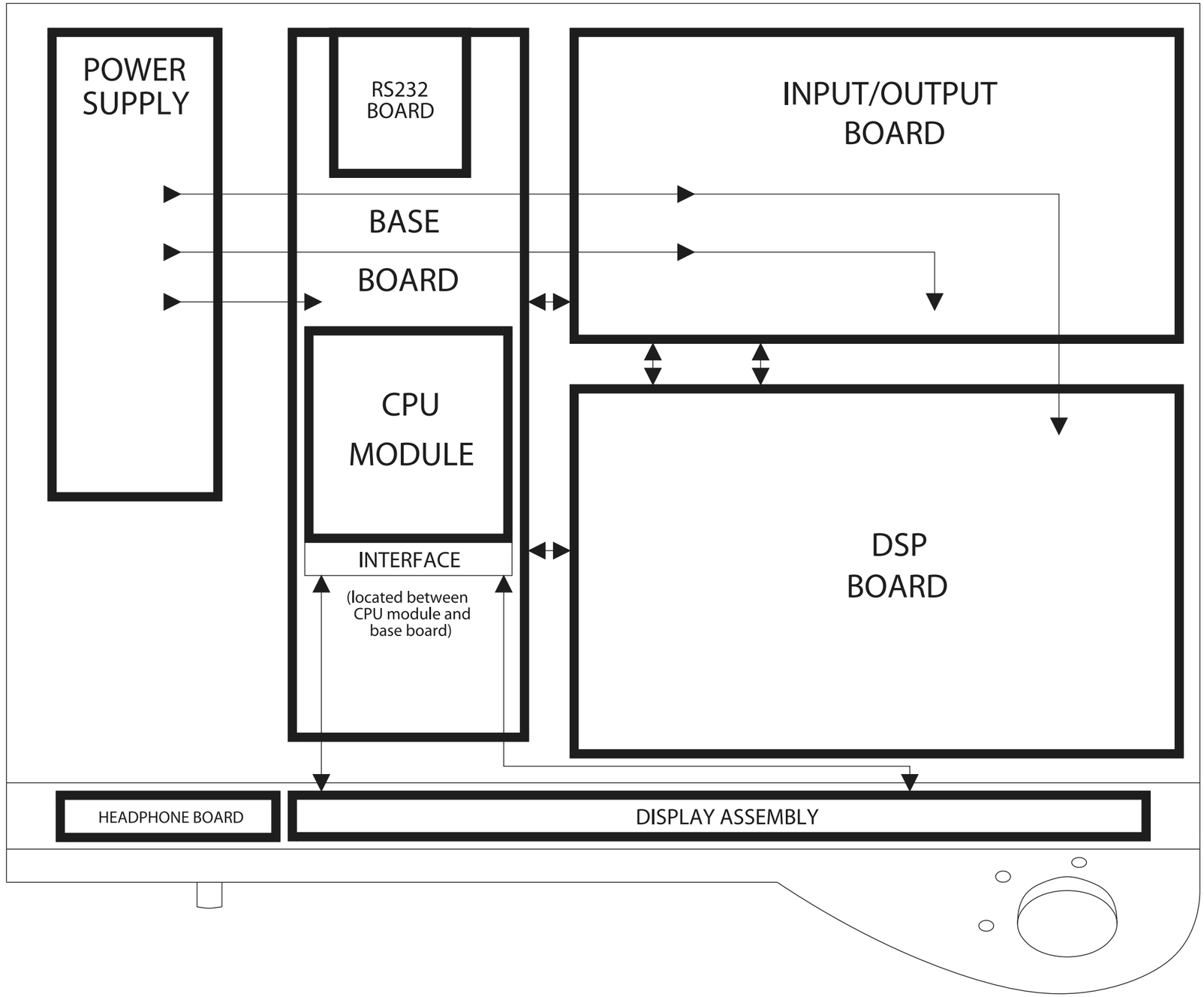
These drawings reflect the actual construction of your unit as accurately as possible. Any differences between the drawings and your unit are probably due to product improvements or production changes since the publication of this manual.

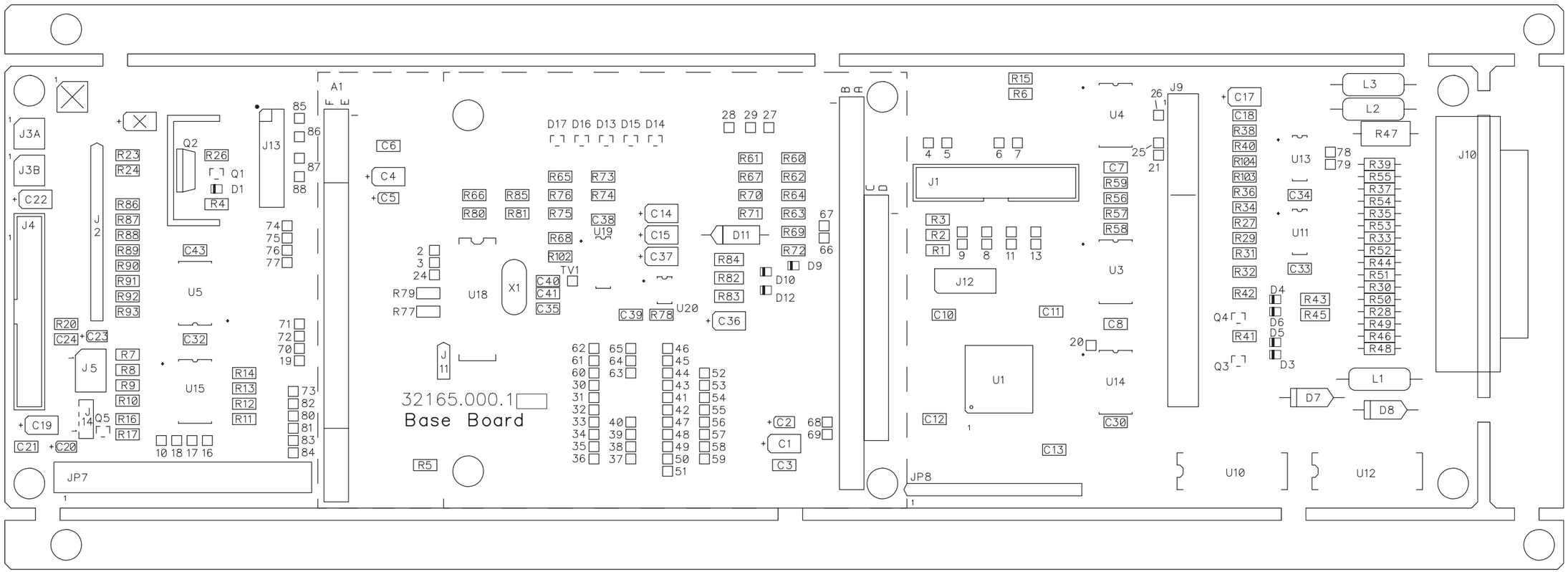
If you intend to replace parts, please read page 6-16. Please note that because surface-mount parts are used extensively in the 8600, few parts are field-replaceable. Servicing ordinarily occurs by swapping circuit board assemblies. However, many vulnerable parts connected to the outside world are socketed and can be readily replaced in the field.

Function	Description	Drawing	Page
Chassis	Circuit Board Locator and basic interconnections	Top view (not to scale)	6-37
Base Board	Glue logic; supports CPU board and RS-232 daughterboard. *Contains:*	Parts Locator Drawing	6-38
	System Connections	Schematic 1 of 4	6-39
	CPU board interface	Schematic 2 of 4	6-40
	Power Supply Monitor	Schematic 3 of 4	6-41

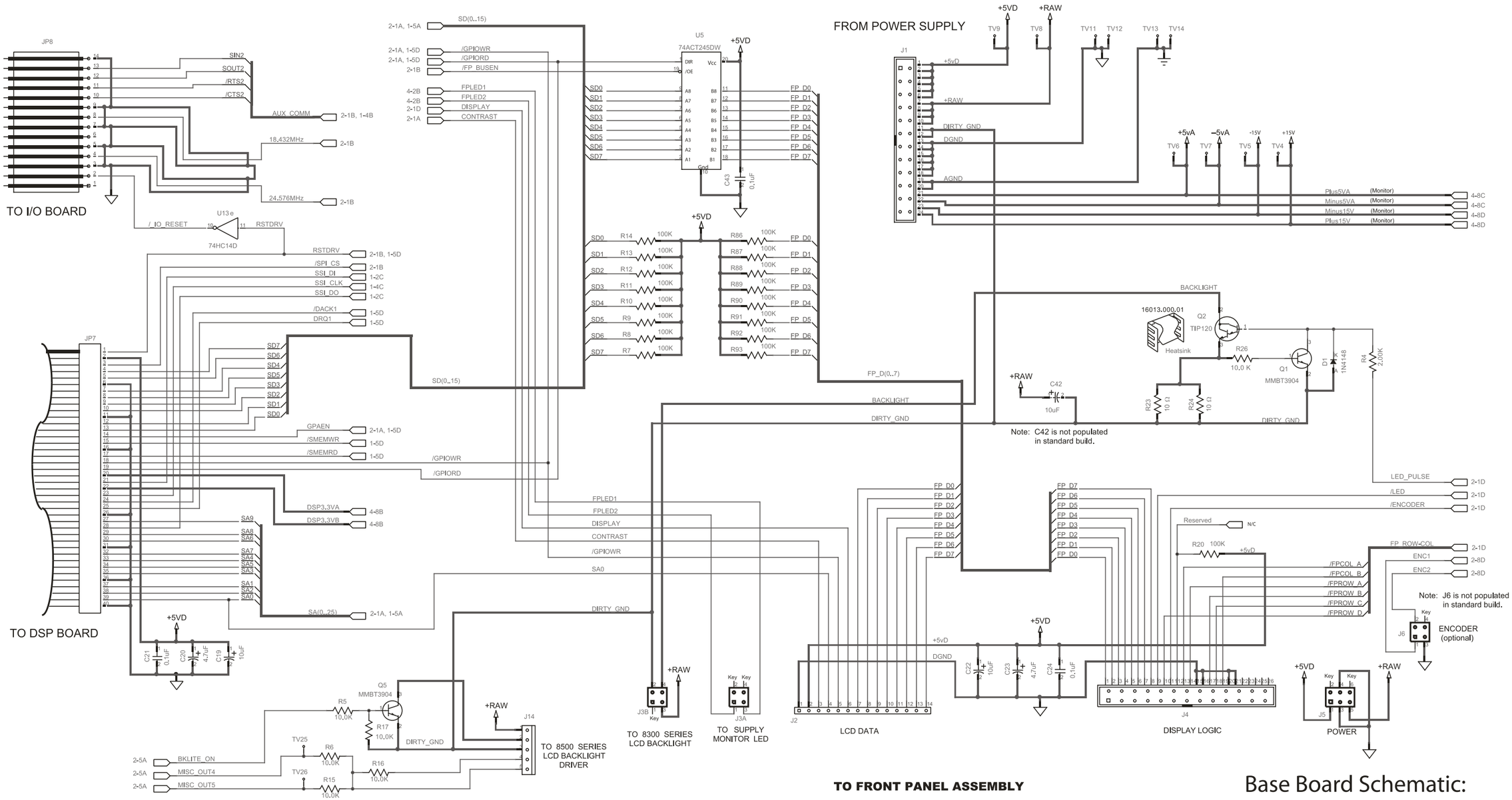
Function	Description	Drawing	Page
	CPLD, General Purpose Interface, and Remotes	Schematic 4 of 4	6-42
CPU board	Control microprocessor. Services front panel, serial port, Ethernet, DSP board, and control board. Resides on base board. ***Contains:***	Parts Locator Drawing	6-43
	Ethernet	Schematic 1 of 5	6-44
	General Purpose Bus	Schematic 2 of 5	6-45
	Memory	Schematic 3 of 5	6-46
	Miscellaneous Functions	Schematic 4 of 5	6-47
	Power and Ground Distribution	Schematic 5 of 5	6-48
RS-232 Board	Supports Serial Port	Parts Locator Drawing	6-49
		Schematic 1 of 1	6-50
Power Supply	±15V analog supply; ±5V analog supply; +5V digital supply	Parts Locator Drawing	6-51
		Schematic 1 of 1	6-52
I/O Board	Analog Input/output AES3 Input/output Composite Output SCA Input. ***Contains:***	Parts Locator Drawing	6-53
	L and R Analog Inputs	Schematic 1 of 6	6-54
	L and R Analog Outputs	Schematic 2 of 6	6-55
	Composite / SCA	Schematic 3 of 6	6-56
	Digital I/O	Schematic 4 of 6	6-57
	Control and Miscellaneous	Schematic 5 of 6	6-58
	Interface and Power Distribution	Schematic 6 of 6	6-59
DSP Board	DSP Chips; Local +3.3V regulator. ***Contains:***	Parts Locator Drawing	6-60
	Interconnects	Schematic 1 of 9	6-61
	Enhanced Serial Audio Interface (ESAI)	Schematic 2 of 9	6-62
	Control Interface	Schematic 3 of 9	6-63
	External Memory Controller Interface 1	Schematic 4 of 9	6-64
	Power and Ground	Schematic 5 of 9	6-65
	86xx 8-Bit Control Interface	Schematic 6 of 9	6-66
	Clock Generation and CPLD	Schematic 7 of 9	6-67
	Power Distribution	Schematic 8 of 9	6-68
	External Memory Controller Interface 2	Schematic 9 of 9	6-69
Front-Panel Boards	LCD Carrier	Parts Locator Drawing	6-70
	LCD Carrier	Schematic 1 of 3	6-71
	Headphone and Encoder Board	Parts Locator	6-72

Function	Description	Drawing	Page
		Drawings	
	Headphone Board	Schematic 2 of 3	6-73
	Encoder Board	Schematic 3 of 3	6-74
Front-Panel Interface Board		Parts Locator Drawing	6-75
		Schematic 1 of 2	6-76
		Schematic 2 of 2	6-77
Digital MPX Board	Retrofit board for pre-MPX 8600s	Parts Locator Drawing	6-78
	For 8600MPX ***Contains:***	Parts Locator Drawing	6-79
	Interconnects	Schematic 1 of 3	6-80
	Digital Composite and Sync	Schematic 2 of 3	6-81
	Digital SCA Inputs and dsPIC	Schematic 3 of 3	6-82
DSP Block Diagram	8500-Style Presets	Drawing 1 of 2	6-83
	8600-Style Presets	Drawing 2 0f 2	6-84

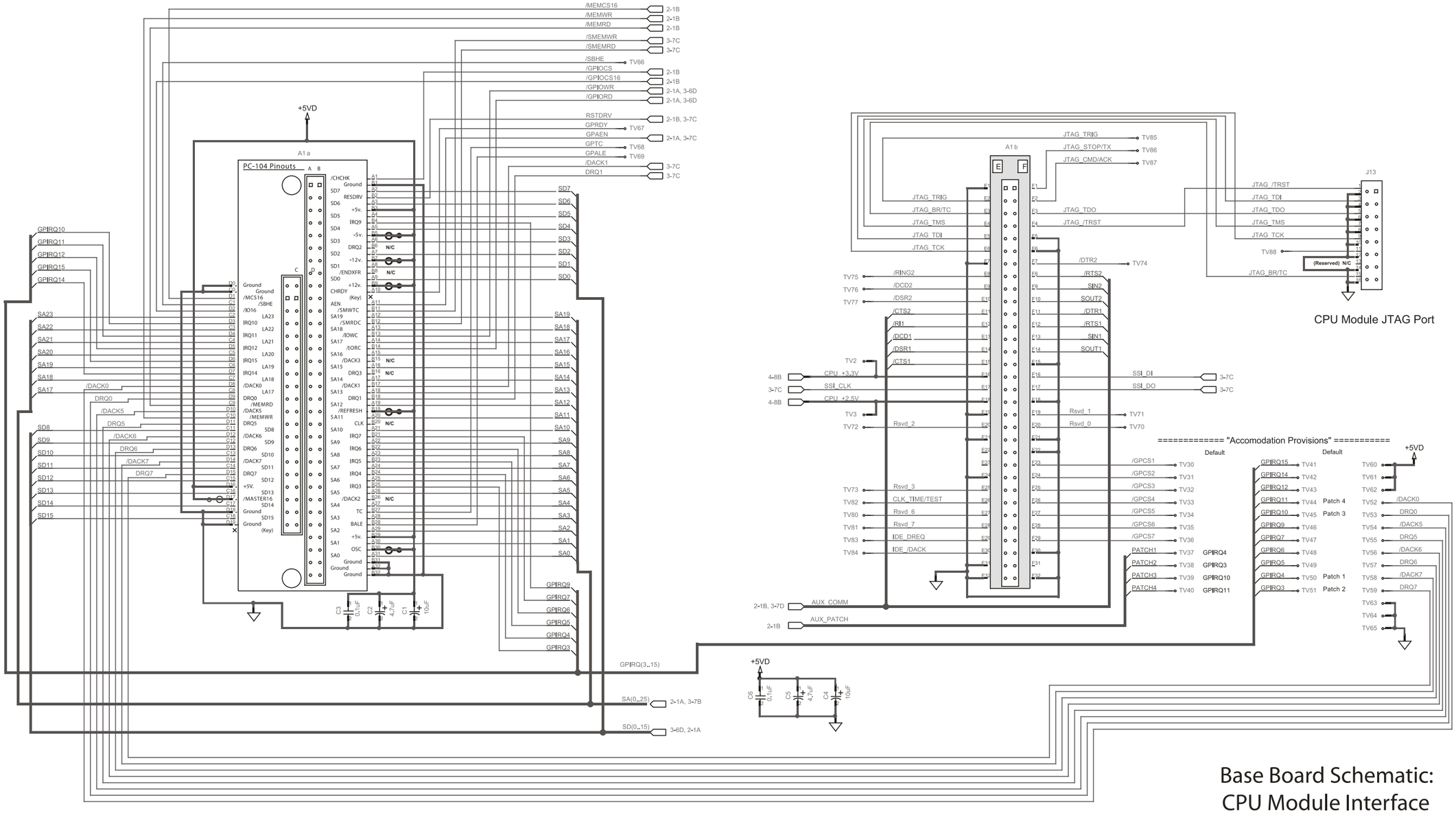

POWER SUPPLY
RS232 BOARD
BASE BOARD
CPU MODULE
INTERFACE
(located between CPU module and base board)
INPUT/OUTPUT BOARD
DSP BOARD
HEADPHONE BOARD
DISPLAY ASSEMBLY



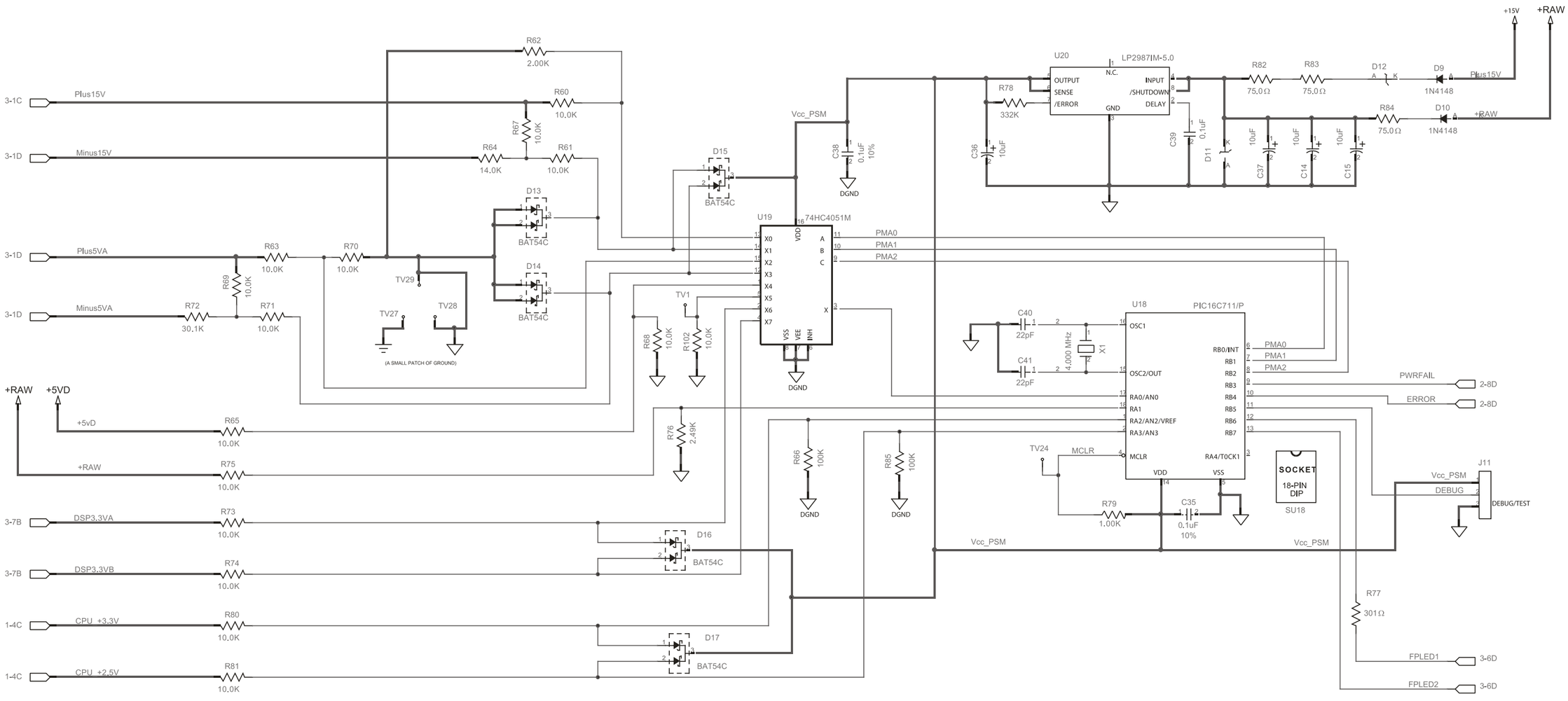
Base Board
Parts Locator Drawing
(version 62165.000.09.1)
Sheet 1 of 1



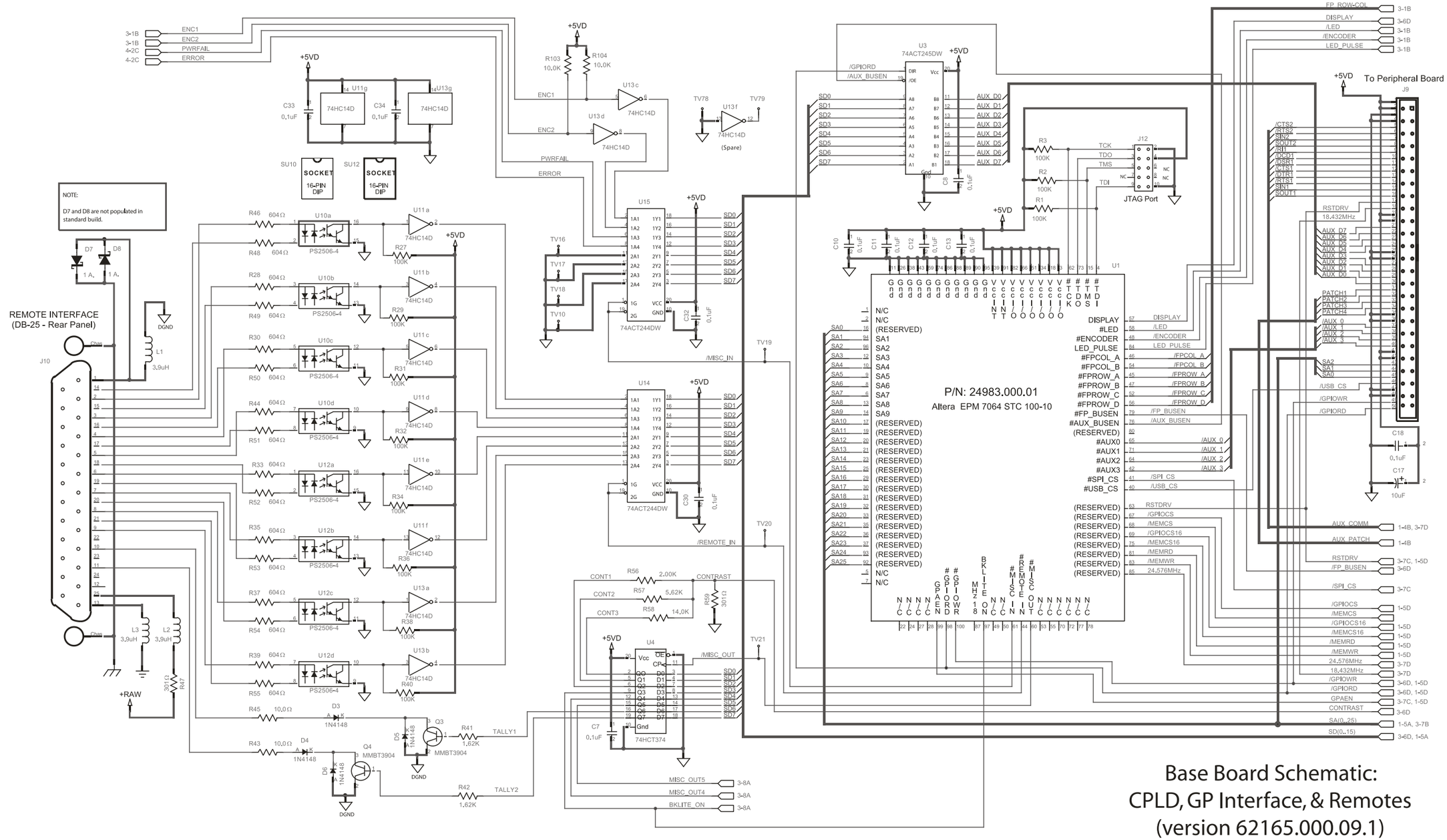
Base Board Schematic:
System Connections
(version 62165.000.09.1)
Sheet 1 of 4

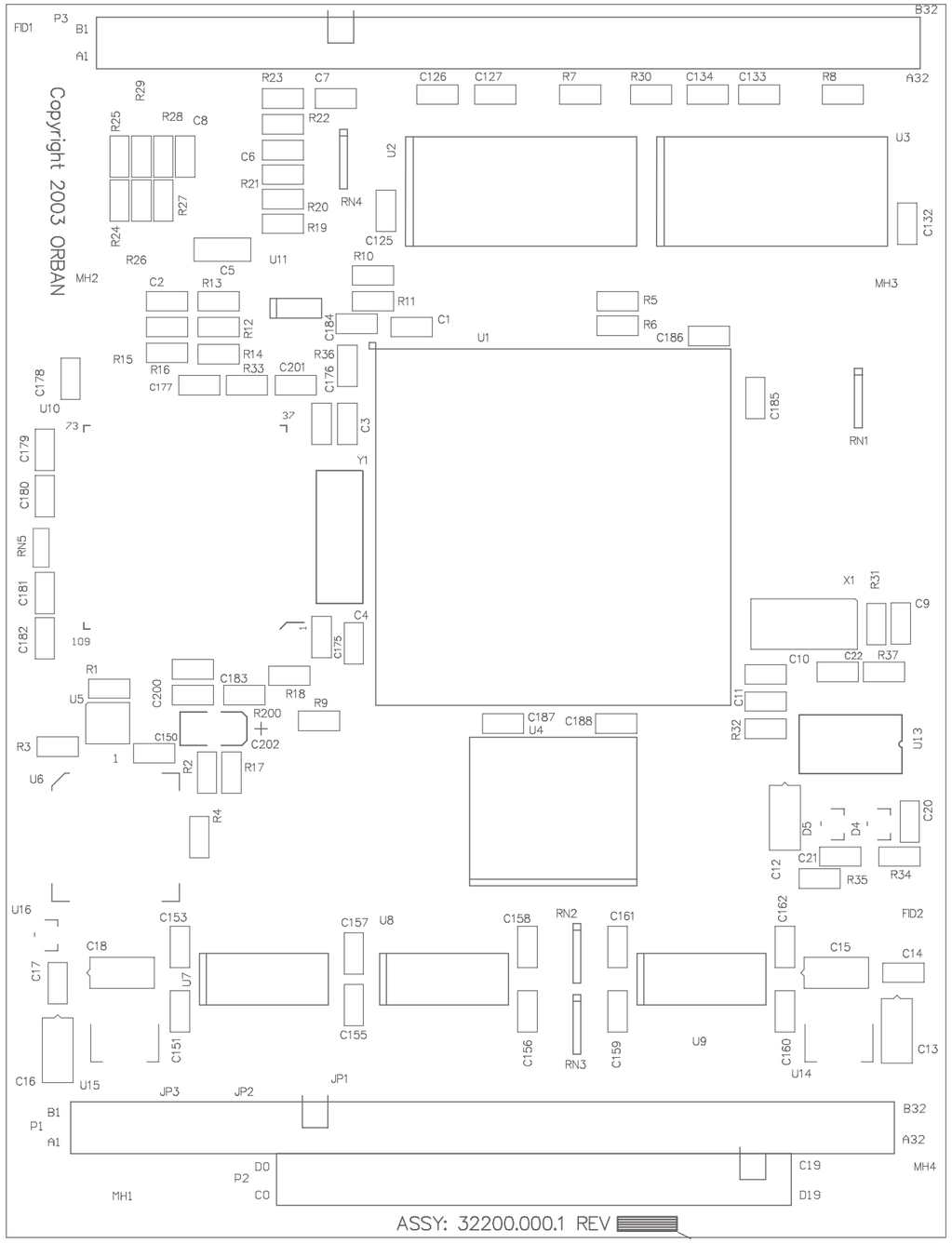


Base Board Schematic:
CPU Module Interface
(version 62165.000.09.1)
Sheet 2 of 4

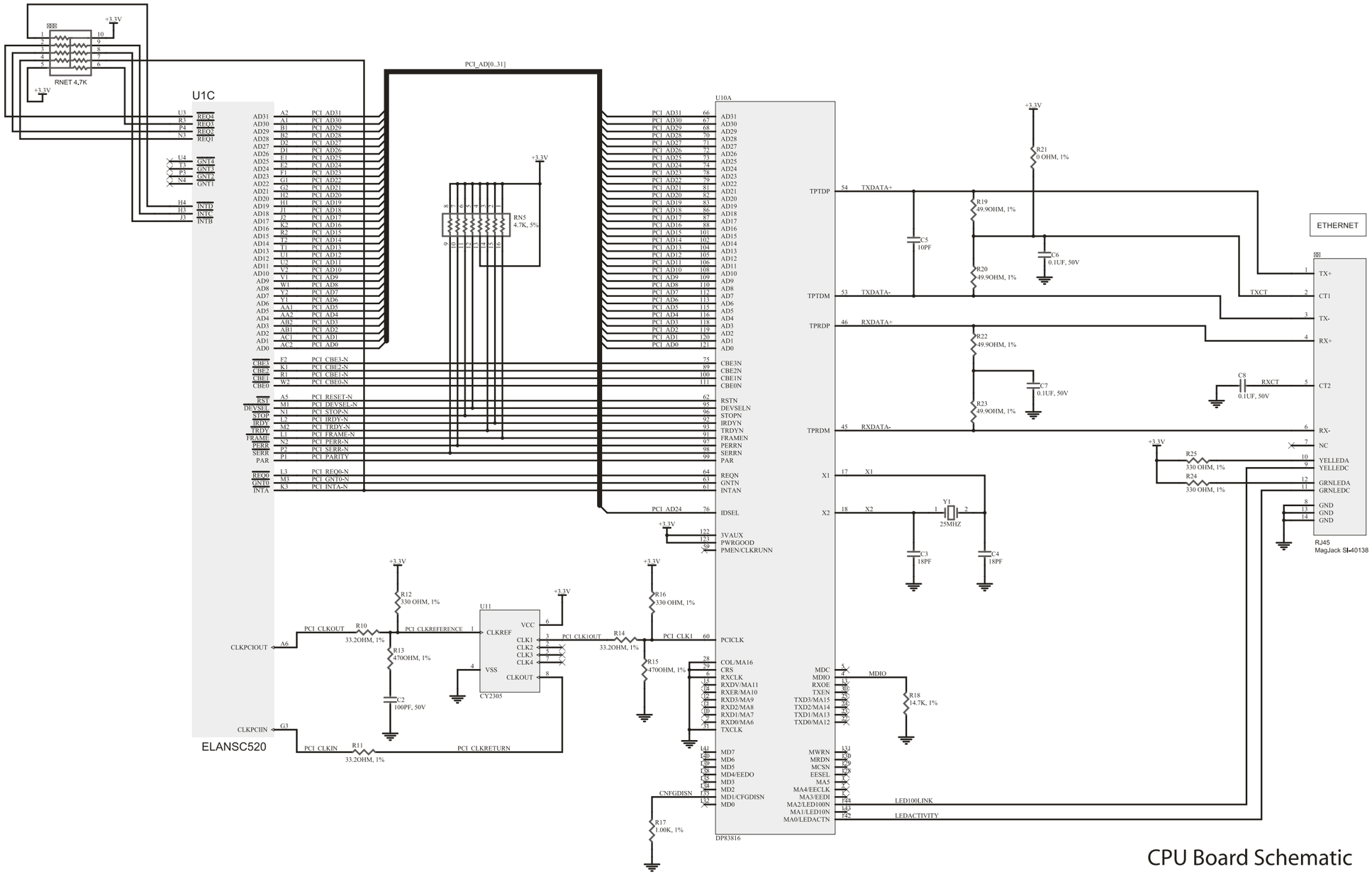


Base Board Schematic:
Power Supply Monitor
(version 62165.000.09.1)
Sheet 3 of 4

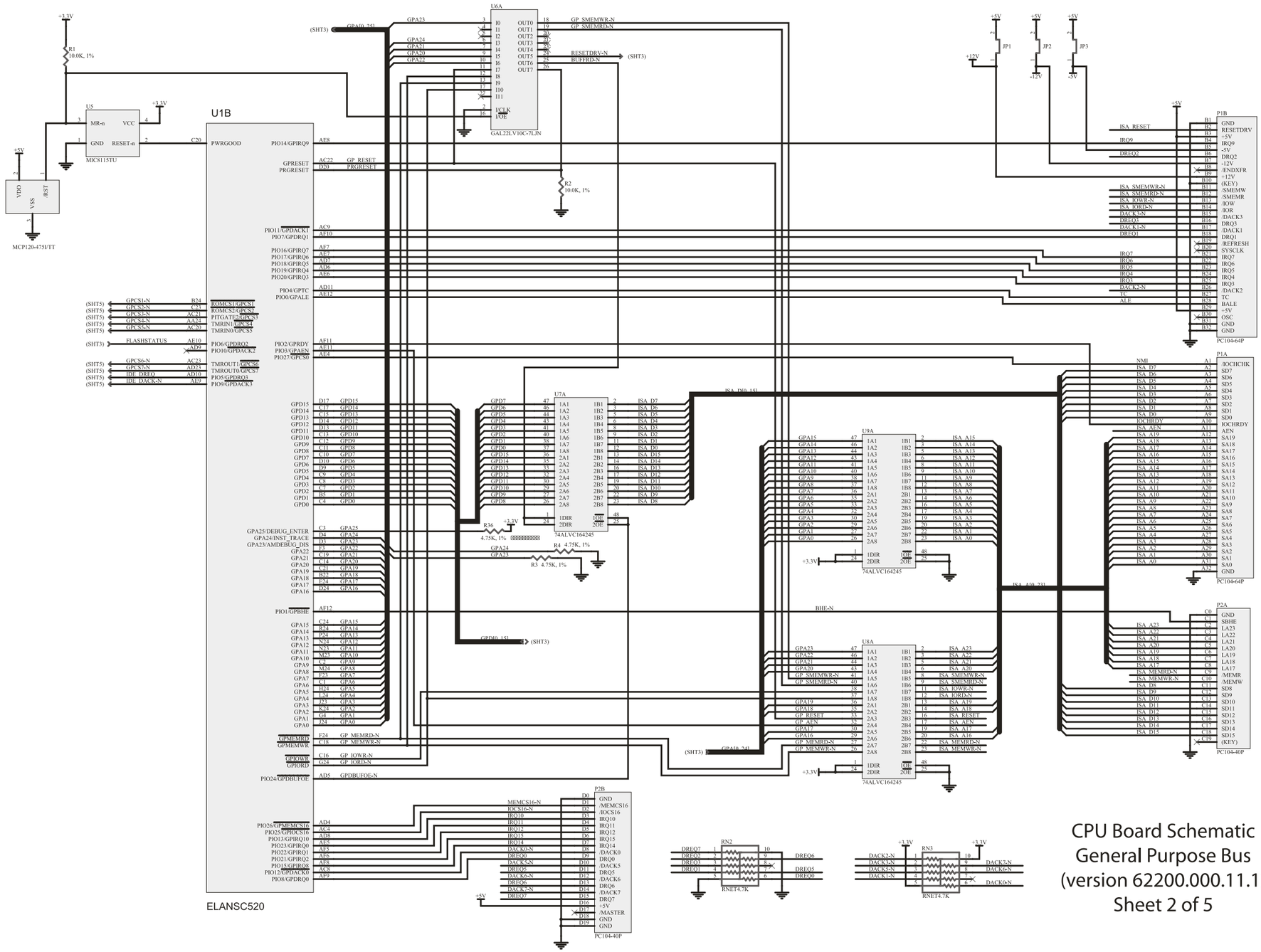

REMOTE INTERFACE
(DB-25 - Rear Panel)
NOTE:
D7 and D8 are not populated in standard build.
P/N: 24983.000.01
Altera EPM 7064 STC 100-10
JTAG Port
To Peripheral Board
Base Board Schematic:
CPLD, GP Interface, & Remotes
(version 62165.000.09.1)
Sheet 4 of 4

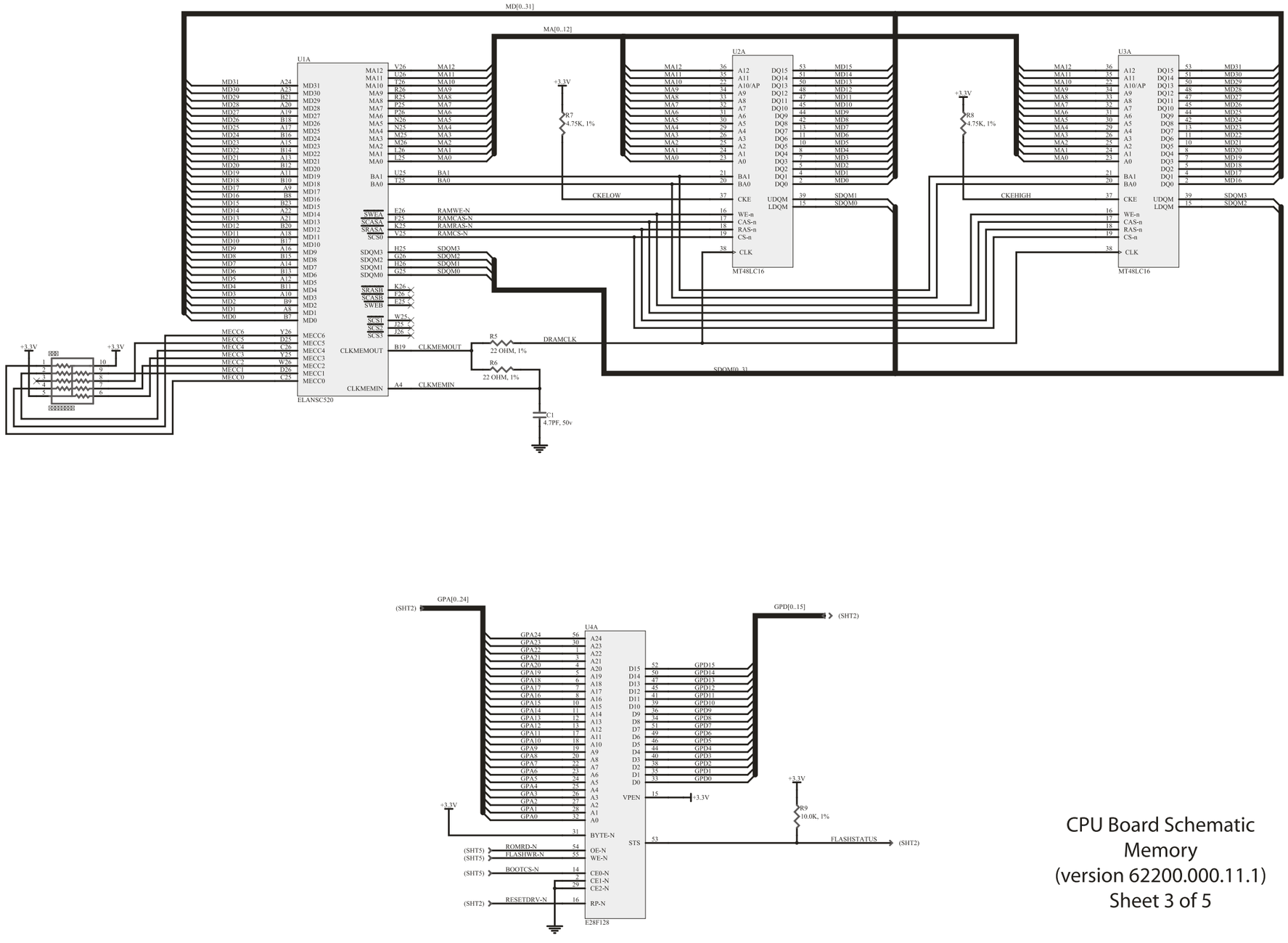


CPU Board
Parts Locator Drawing
(version 62200.000.11.1)
Sheet 1 of 1

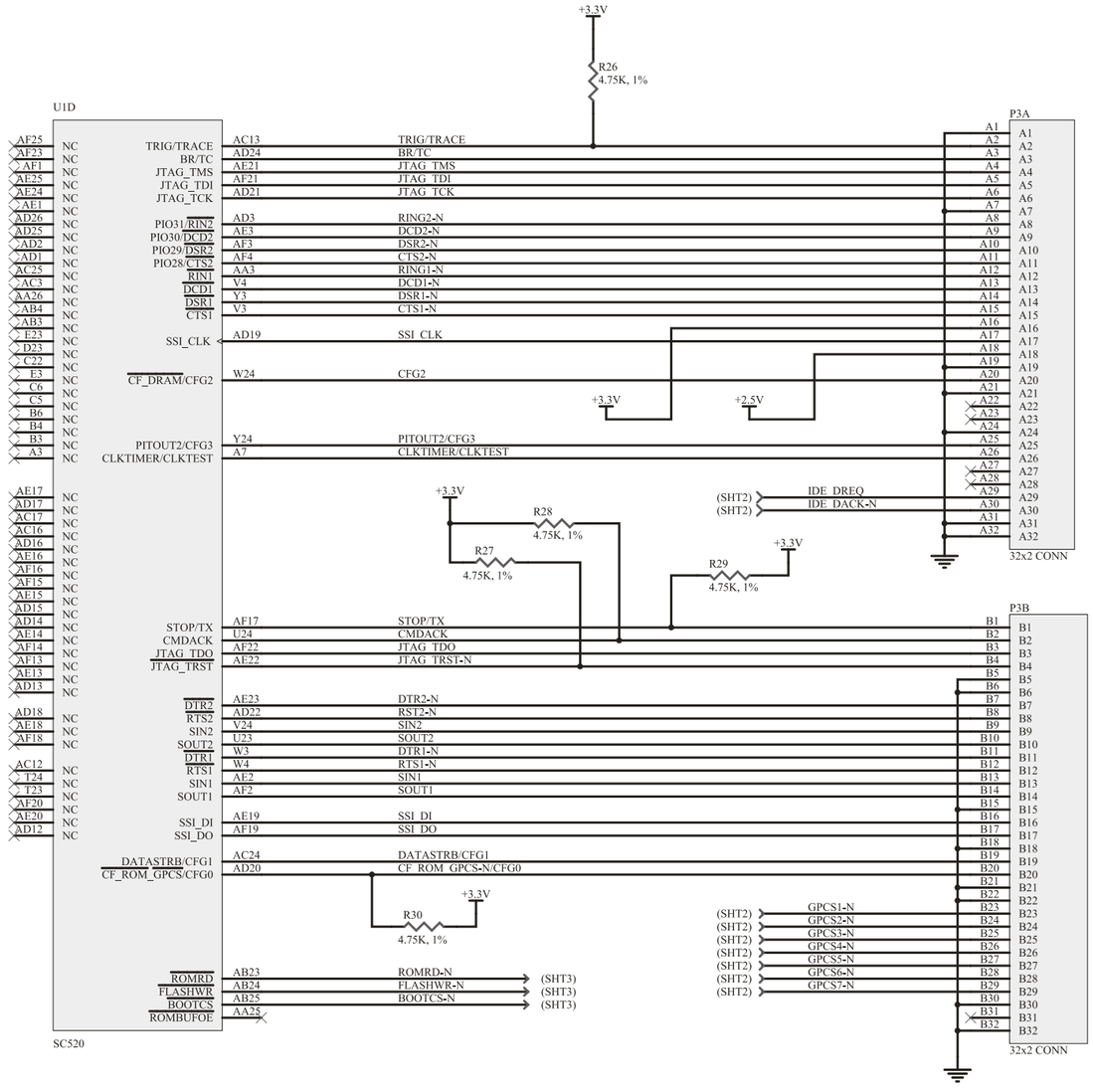


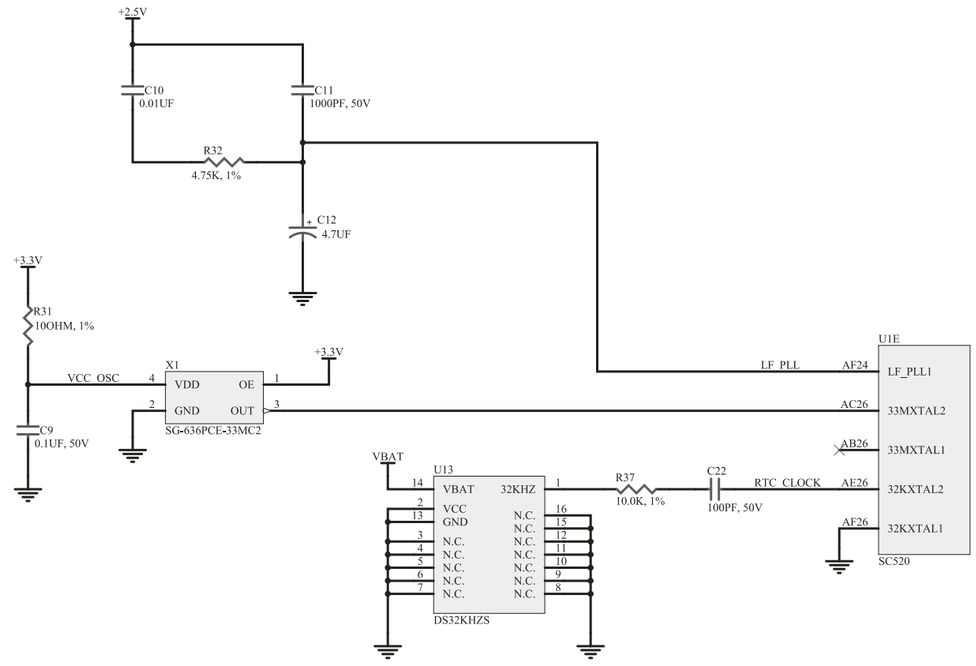
CPU Board Schematic
Ethernet
(version 62200.000.11.1)
Sheet 1 of 5


U1B
ELANSC520
U6A
GAL22LV10C-7LJN
U7A
74ALVC164245
U9A
74ALVC164245
U8A
74ALVC164245
U5
MIC8115TU
MCP120-475I/TT
P1B
PC104-64P
P1A
PC104-64P
P2A
PC104-40P
P2B
PC104-40P
RN2
RNET4.7K
RN3
RNET4.7K
JP1
JP2
JP3
R1 10.0K, 1%
R2 10.0K, 1%
R3 4.75K, 1%
R4 4.75K, 1%
R36 4.75K, 1%
GP_SMEMWR-N
GP_SMEMRD-N
RESETDRV-N
BUFFRD-N
GP_RESET
PRGRESET
BHE-N
ISA_D0_15
ISA_A0_23
GPA0_23
GPD0_15
CPU Board Schematic
General Purpose Bus
(version 62200.000.11.1)
Sheet 2 of 5

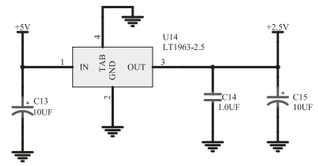


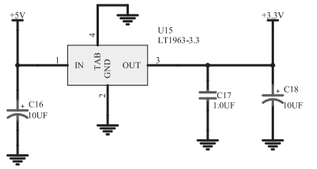
CPU Board Schematic
Memory
(version 62200.000.11.1)
Sheet 3 of 5

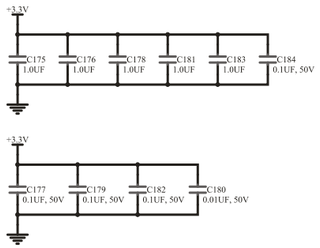


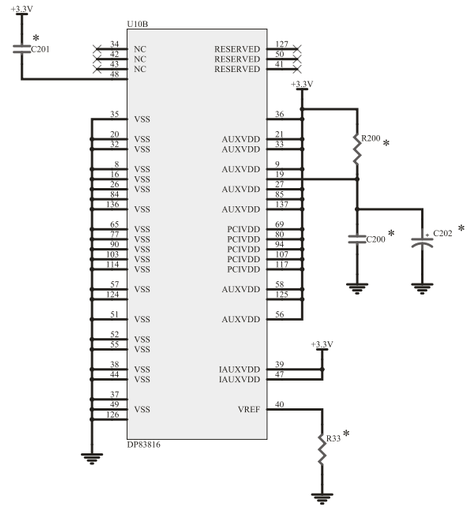


CPU Board Schematic
Miscellaneous Functions
(version 62200.000.11.1)
Sheet 4 of 5

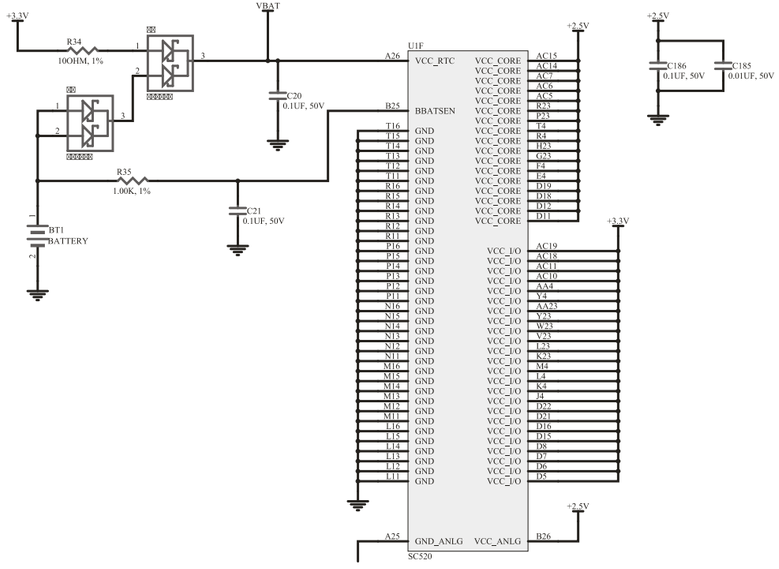


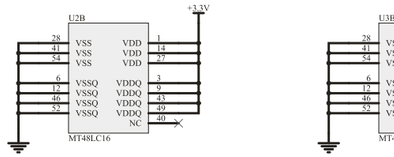


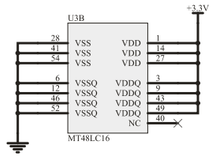


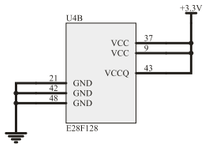


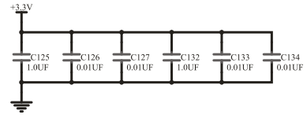
COMPONENT VALUE TABLE FOR DP83815/DP83816		
COMPONENT	DP83815 VALUE	DP83816 VALUE
R33	9.31K, 1%, 0805 20130.931.01.1	10.0K, 1%, 0805 20131.100.01.1
R200	0 OHM, 1%, 0805 20128.000.01.1	NO STUFF
C200	0.01UF, 0805 21146.310.01.1	0.1UF, 0805 21139.000.01.1
C201	NO STUFF	NO STUFF
C202	NO STUFF	10.0 UF, 3528 21325.610.01.1

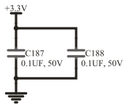


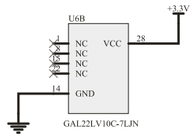


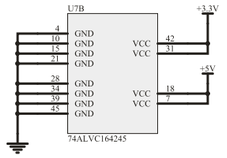


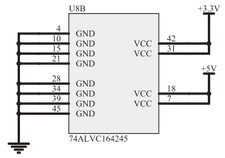


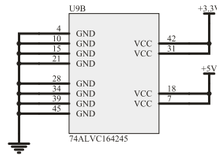


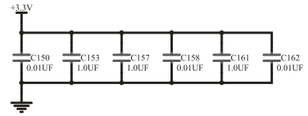


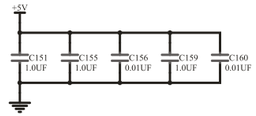




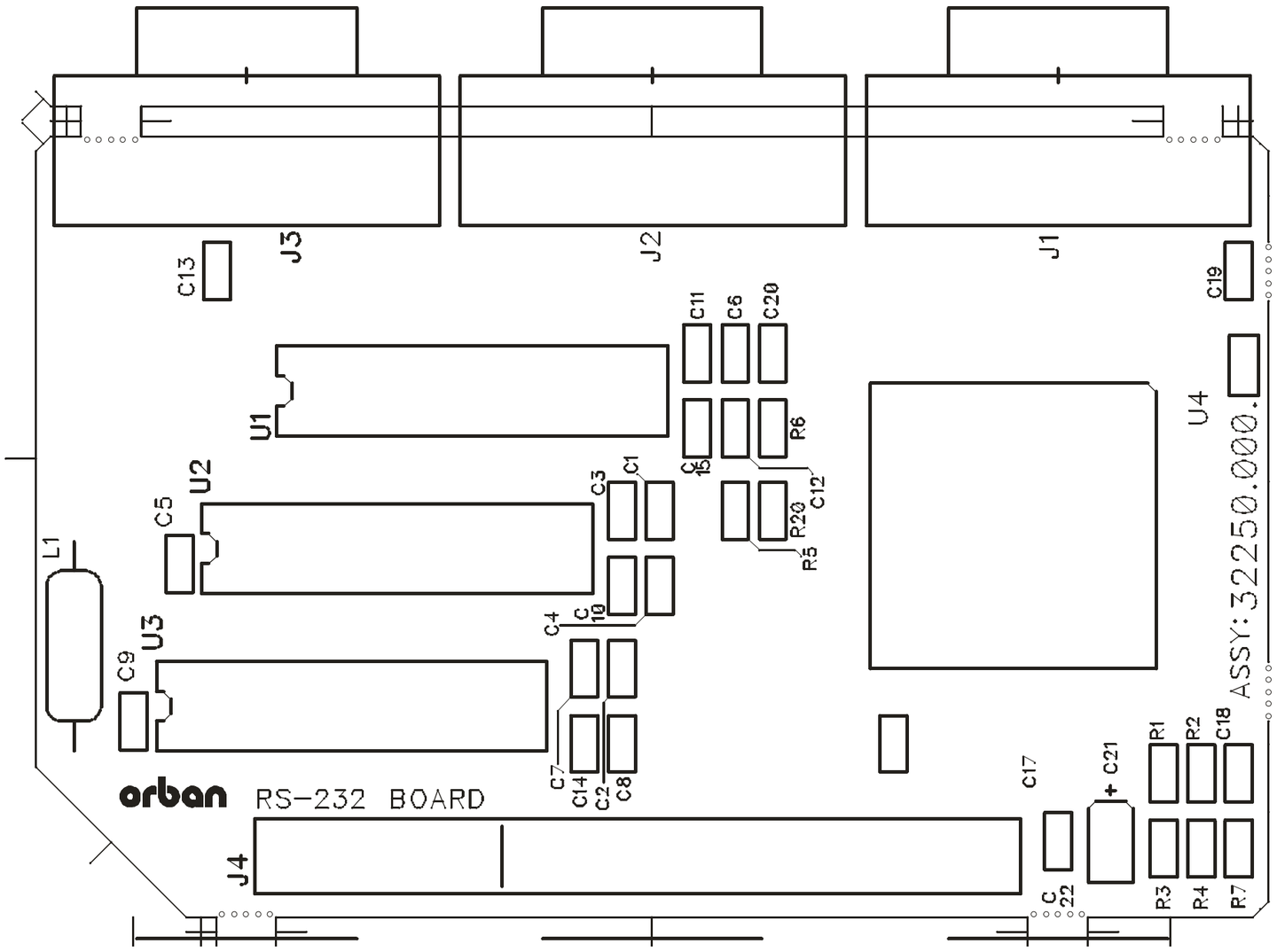




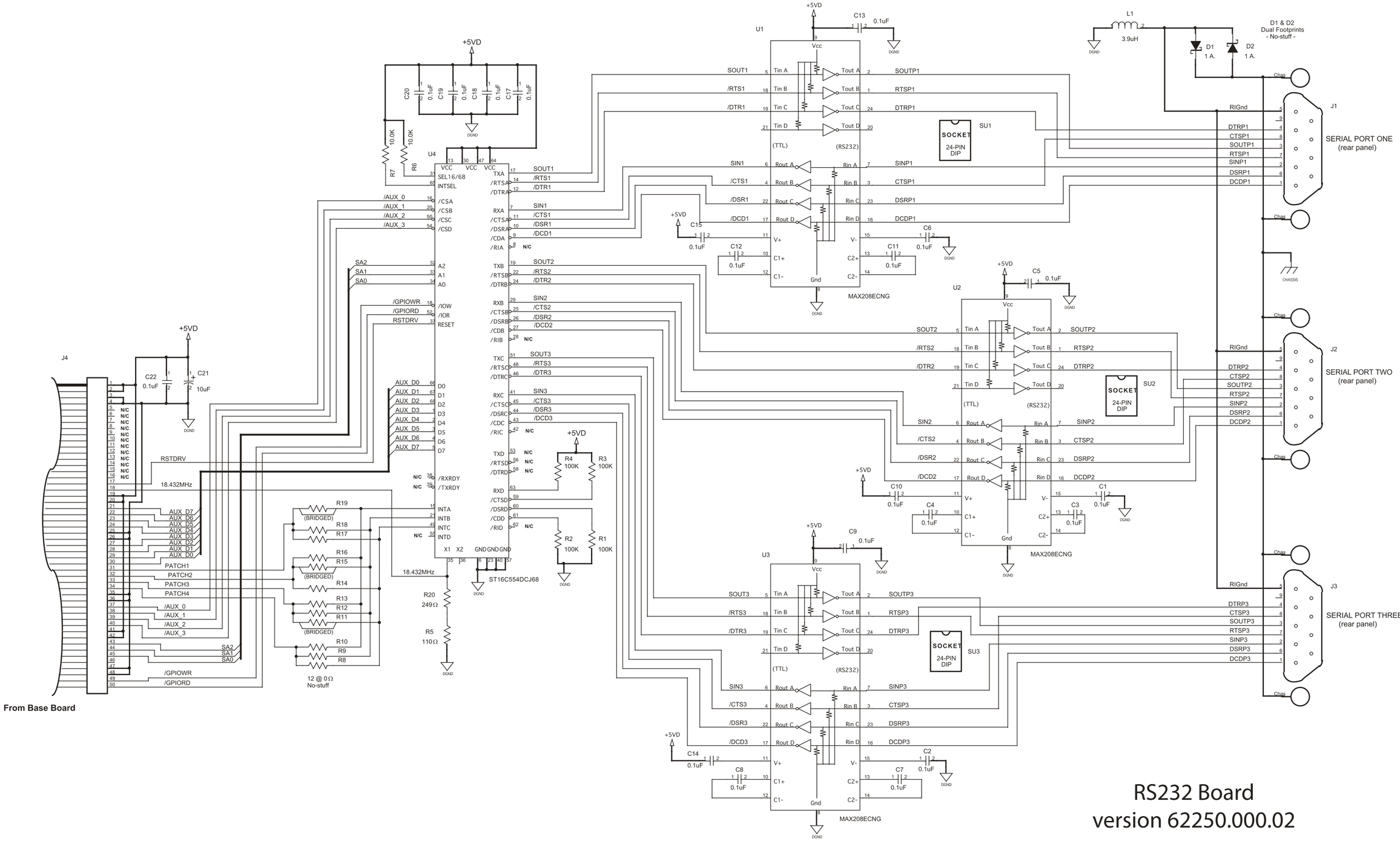


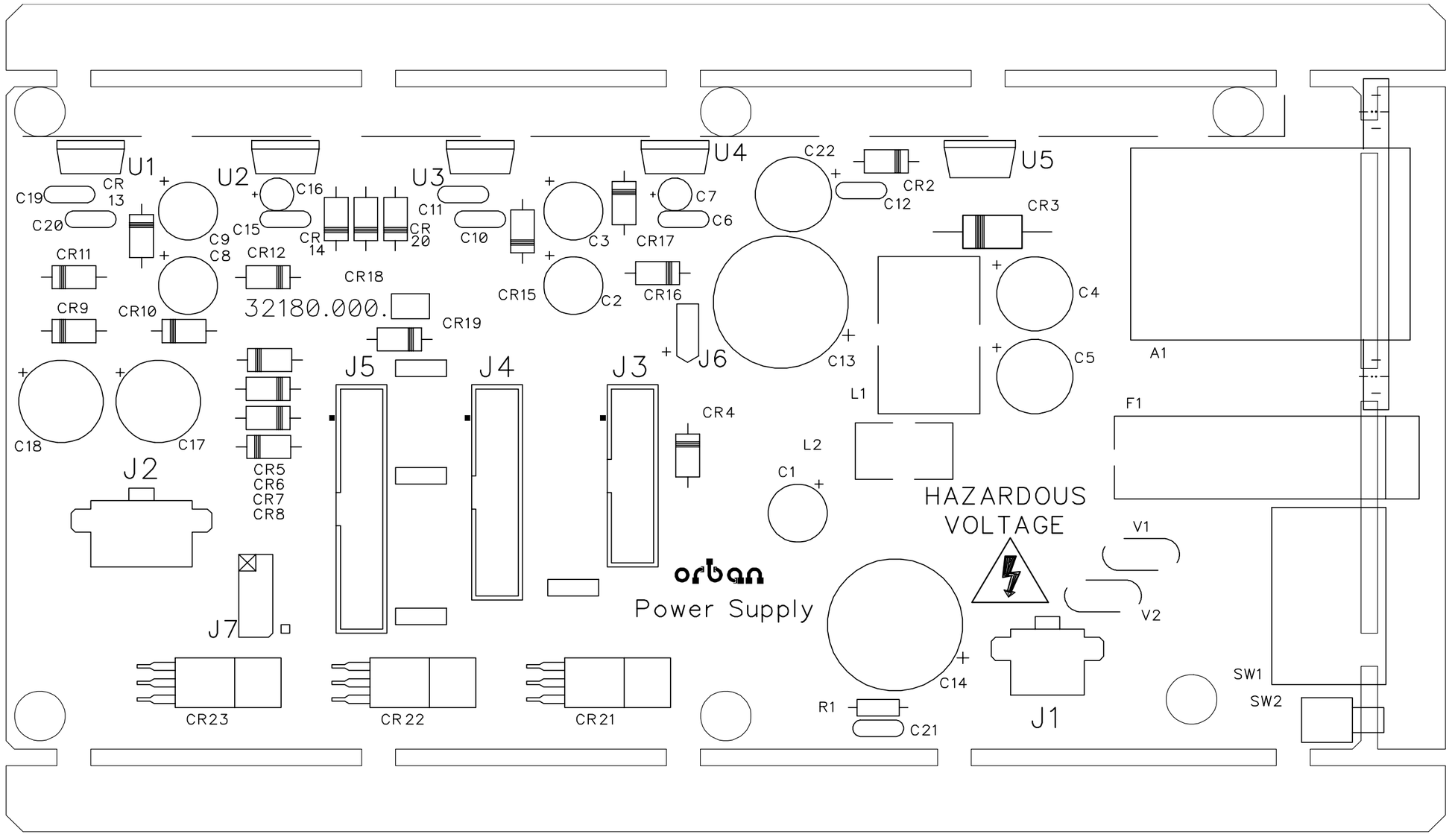


CPU Board Schematic
Power and Ground
(version 62200.000.11.1)
Sheet 5 of 5

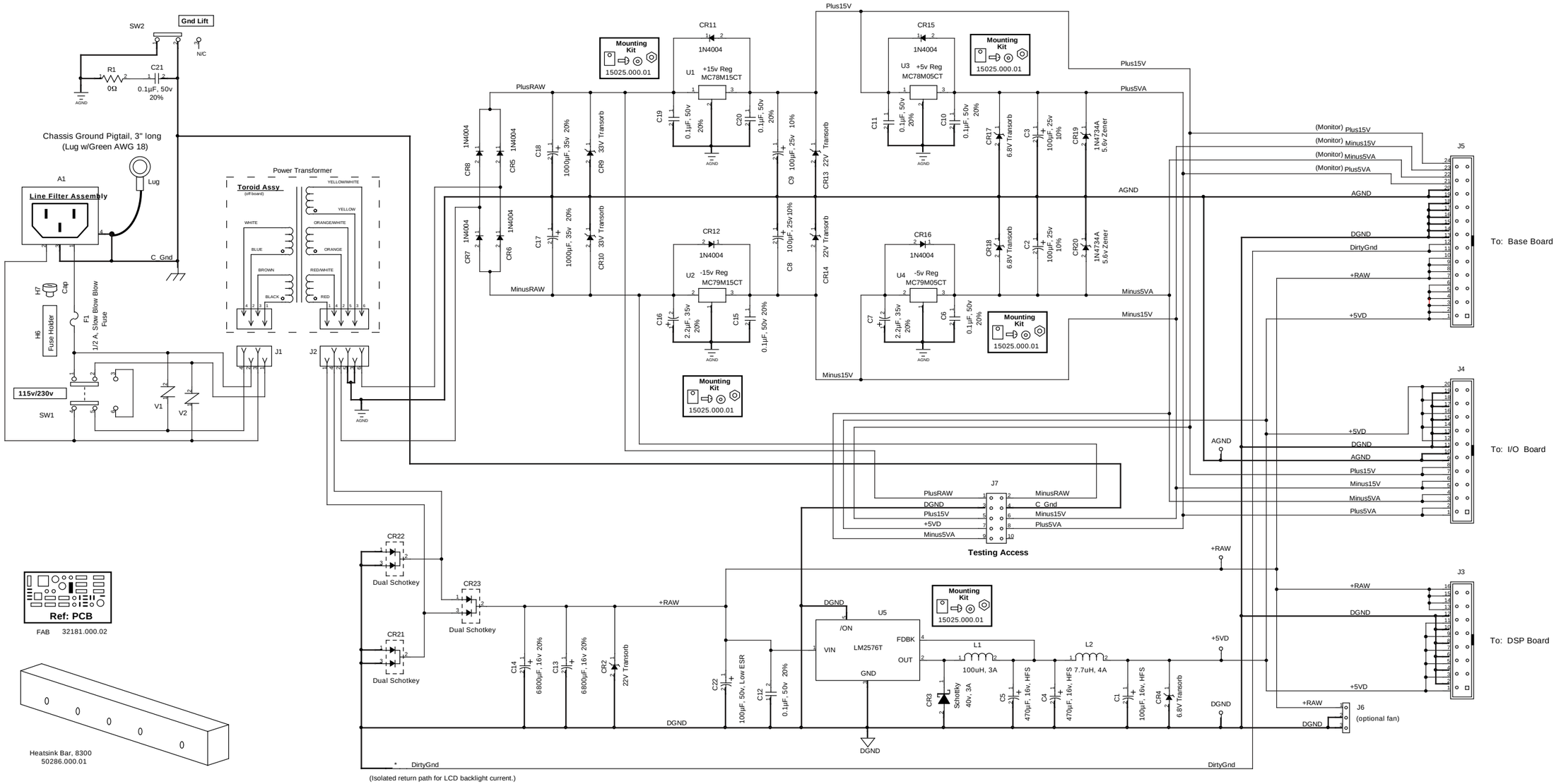


RS232 Board Parts Locator Drawing
(for schematic 62250.000.02)


From Base Board
J4
+5VD
C22 0.1uF
C21 10uF
DGND
N/C
RSTDRV
18.432MHz
AUX_D7
AUX_D6
AUX_D5
AUX_D4
AUX_D3
AUX_D2
AUX_D1
AUX_D0
PATCH1
PATCH2
PATCH3
PATCH4
/AUX_0
/AUX_1
/AUX_2
/AUX_3
SA2
SA1
SA0
/GPIOWR
/GPIORD
C20 0.1uF
C19 0.1uF
C18 0.1uF
C17 0.1uF
10.0K
R7
R6
U4
VCC
SEL16/68
INTSEL
/CSA
/CSB
/CSC
/CSD
A2
A1
A0
/IOW
/IOR
RESET
D0
D1
D2
D3
D4
D5
D6
D7
/RXRDY
/TXRDY
INTA
INTB
INTC
INTD
X1
X2
GND
TXA
/RTSA
/DTRA
RXA
/CTSA
/DSRA
/CDA
/RIA
TXB
/RTSB
/DTRB
RXB
/CTSB
/DSRB
/CDB
/RIB
TXC
/RTSC
/DTRC
RXC
/CTSC
/DSRC
/CDC
/RIC
TXD
/RTSD
/DTRD
RXD
/CTSD
/DSRD
/CDD
/RID
SOUT1
/RTS1
/DTR1
SIN1
/CTS1
/DSR1
/DCD1
SOUT2
/RTS2
/DTR2
SIN2
/CTS2
/DSR2
/DCD2
SOUT3
/RTS3
/DTR3
SIN3
/CTS3
/DSR3
/DCD3
ST16C554DCJ68
R19 (BRIDGED)
R18
R17
R16
R15
R14
R13
R12
R11
R10
R9
R8
12 @ 0Ω
No-stuff
R20 249Ω
R5 110Ω
R4 100K
R3 100K
R2 100K
R1 100K
U1
Vcc
Tin A
Tin B
Tin C
Tin D
Tout A
Tout B
Tout C
Tout D
(TTL)
(RS232)
Rout A
Rout B
Rout C
Rout D
Rin A
Rin B
Rin C
Rin D
V+
V-
C1+
C1-
C2+
C2-
Gnd
MAX208ECNG
C13 0.1uF
C15 0.1uF
C12 0.1uF
C11 0.1uF
C6 0.1uF
SOUTP1
RTSP1
DTRP1
SINP1
CTSP1
DSRP1
DCDP1
SOCKET
24-PIN DIP
SU1
L1
3.9uH
D1 & D2
Dual Footprints
- No-stuff -
D1 1 A.
D2 1 A.
Chas
RIGnd
J1
SERIAL PORT ONE
(rear panel)
CHASSIS
U2
C5 0.1uF
C10 0.1uF
C4 0.1uF
C3 0.1uF
C1 0.1uF
SOUTP2
RTSP2
DTRP2
SINP2
CTSP2
DSRP2
DCDP2
SU2
J2
SERIAL PORT TWO
(rear panel)
U3
C9 0.1uF
C14 0.1uF
C8 0.1uF
C7 0.1uF
C2 0.1uF
SOUTP3
RTSP3
DTRP3
SINP3
CTSP3
DSRP3
DCDP3
SU3
J3
SERIAL PORT THREE
(rear panel)
RS232 Board
version 62250.000.02



POWER SUPP LY PARTS LOC ATOR

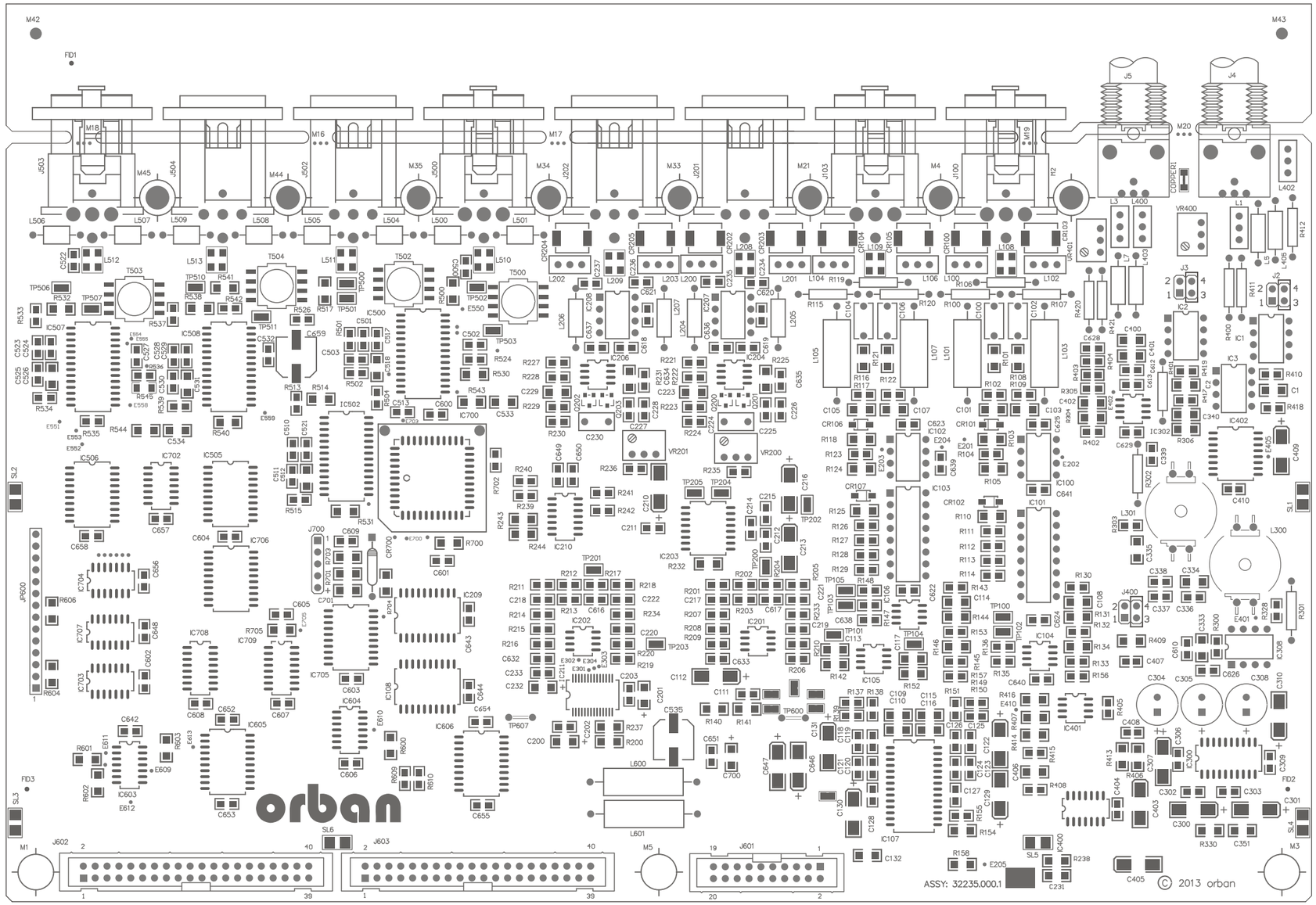


Ref: PCB

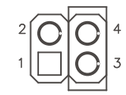
FAB 32181.000.02

Heatsink Bar, 8300
50286.000.01

POWER SUPPLY

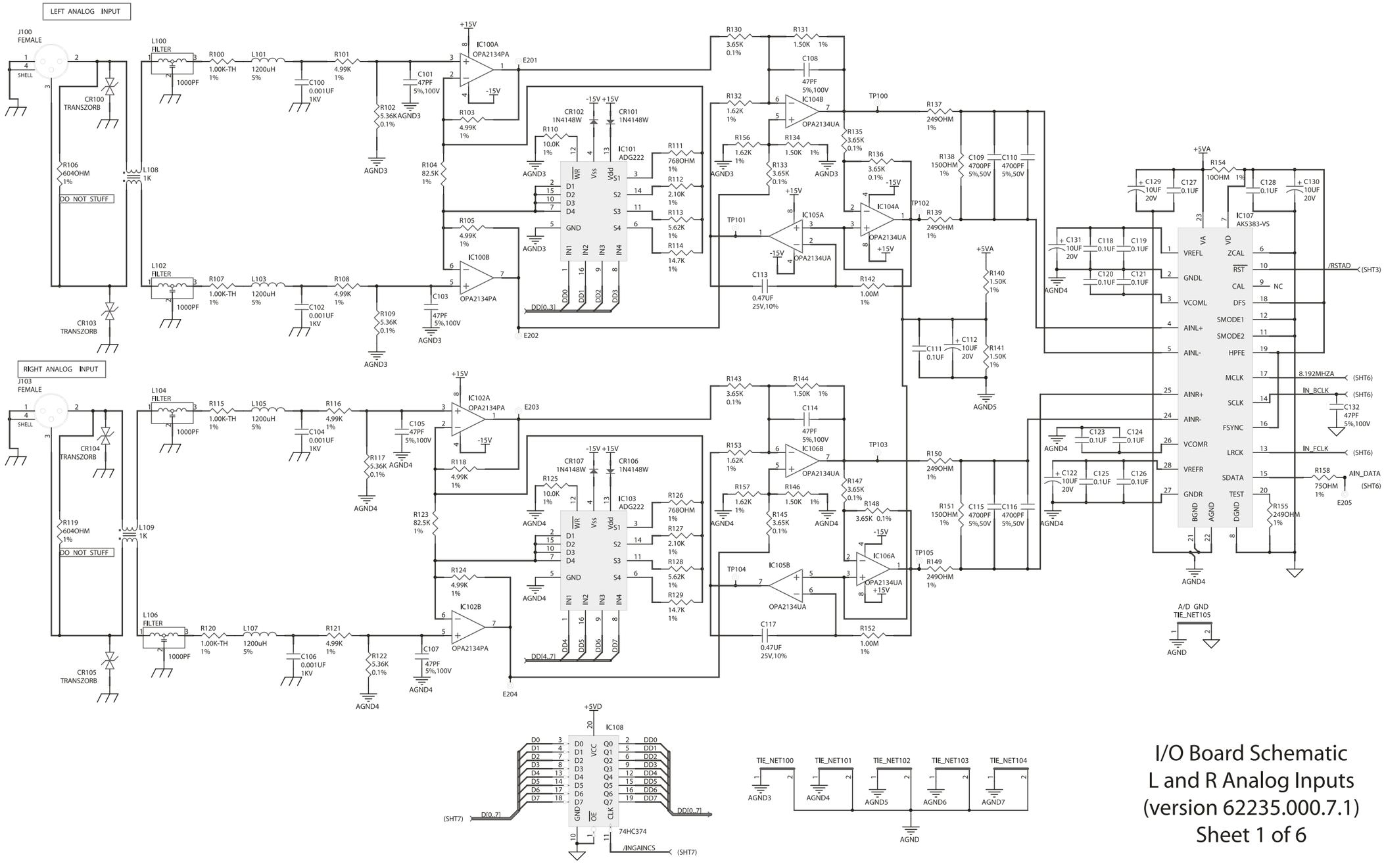


JUMPER DIAGRAM
J2, J3, J400

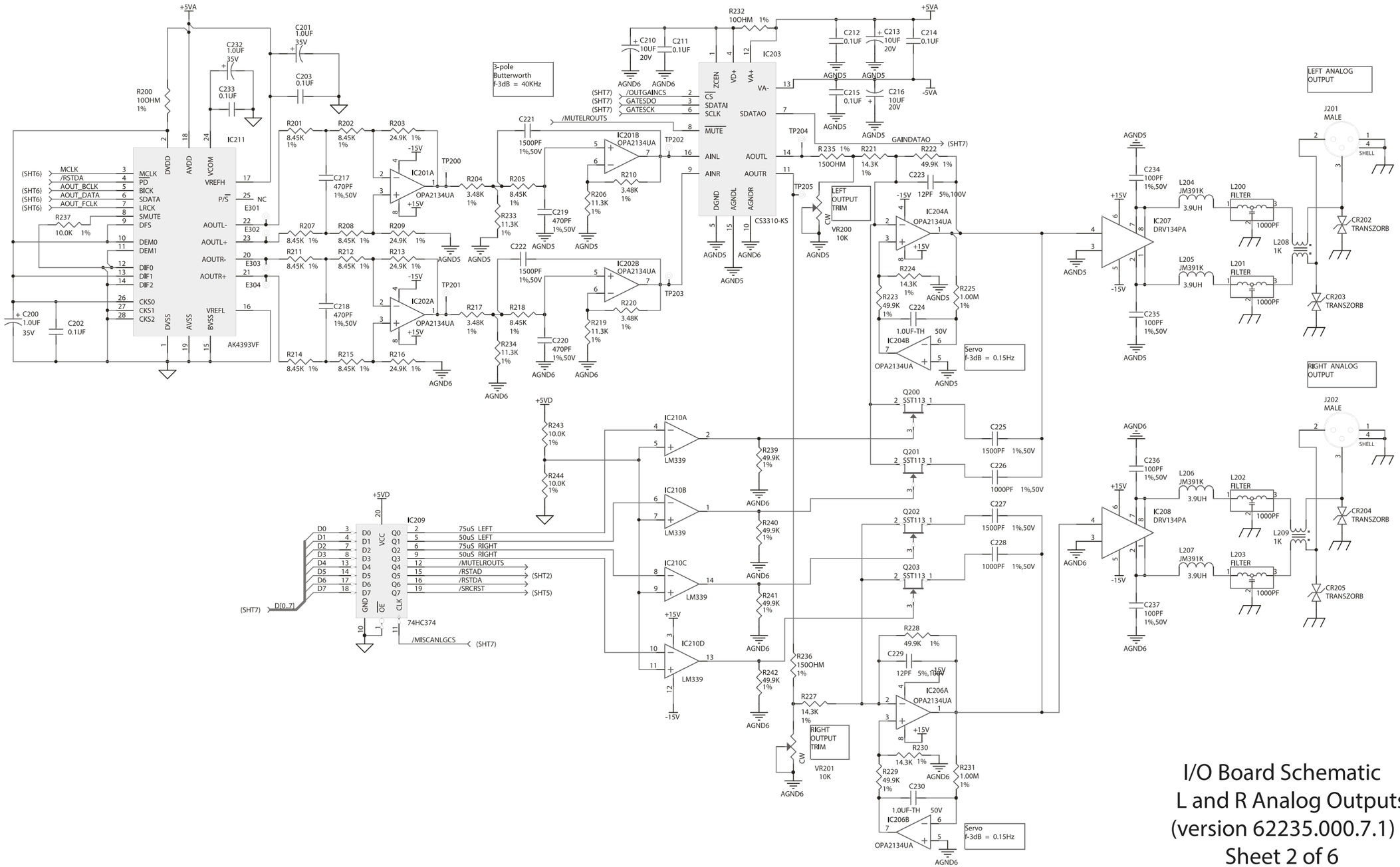


PLACE JUMPER RECEPTICLES
IN SECOND POSITION

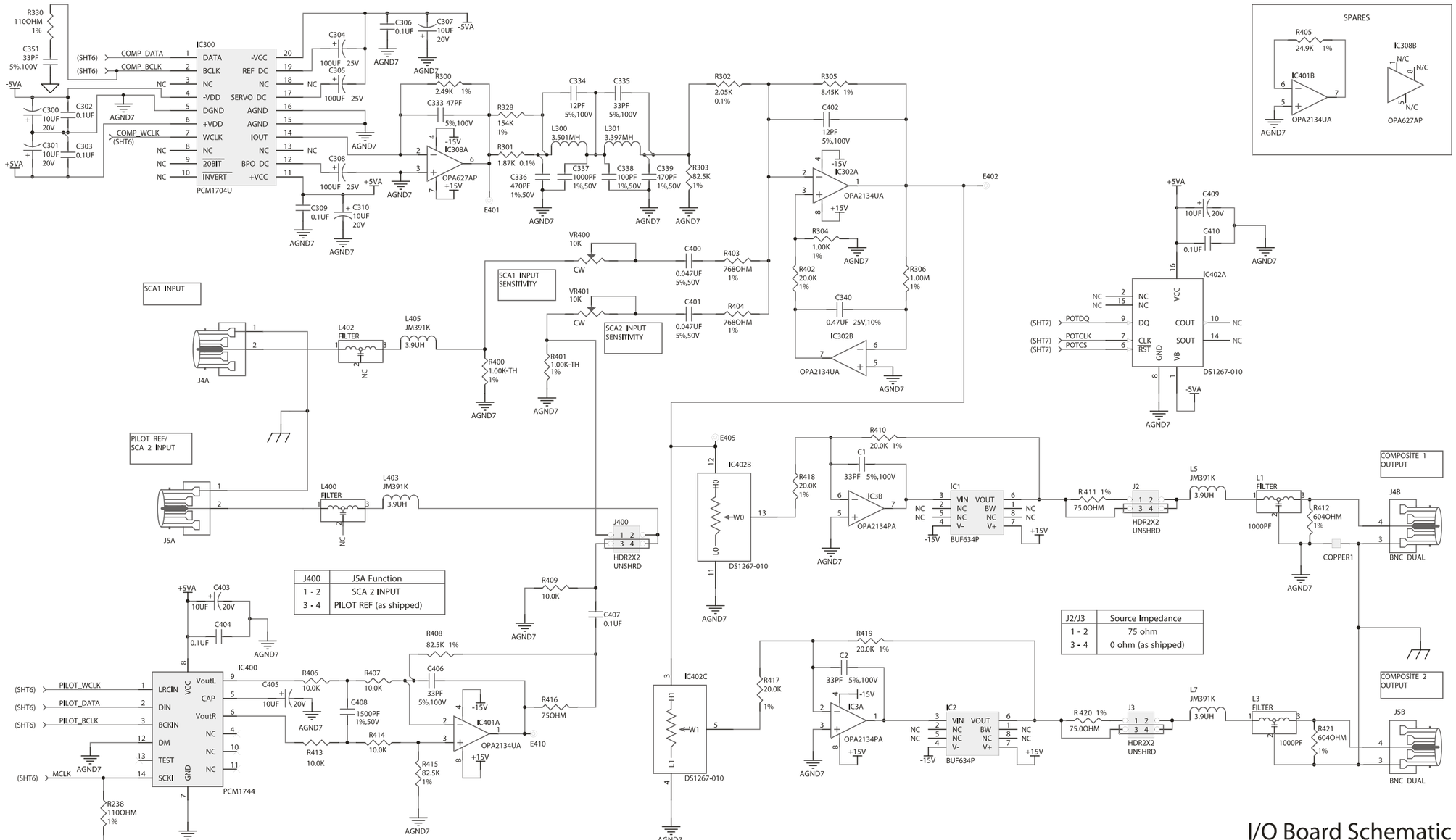
I/O Board
Parts Locator Diagram
(version 62235.000.10.1)
Sheet 1 of 1



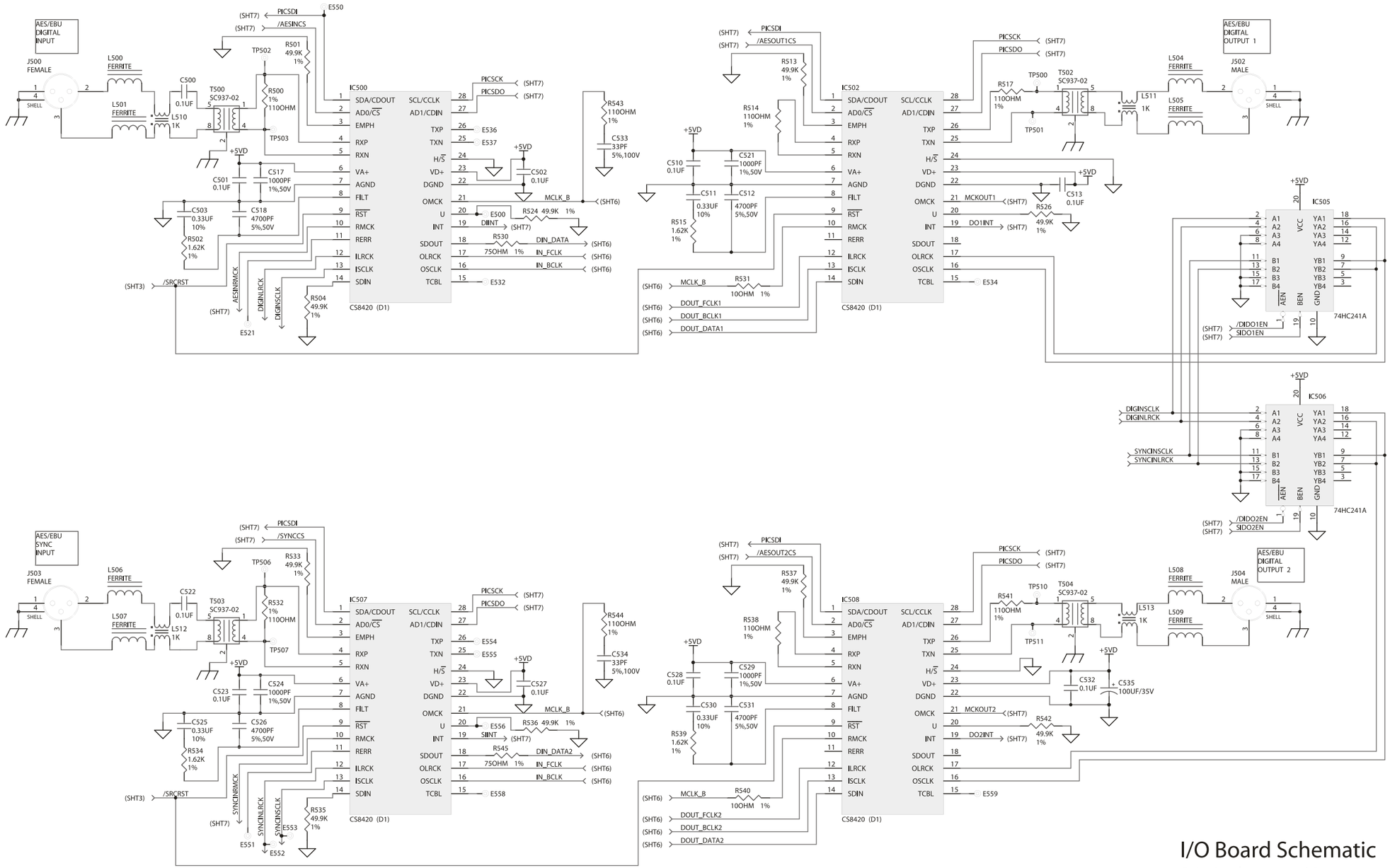
I/O Board Schematic
L and R Analog Inputs
(version 62235.000.7.1)
Sheet 1 of 6



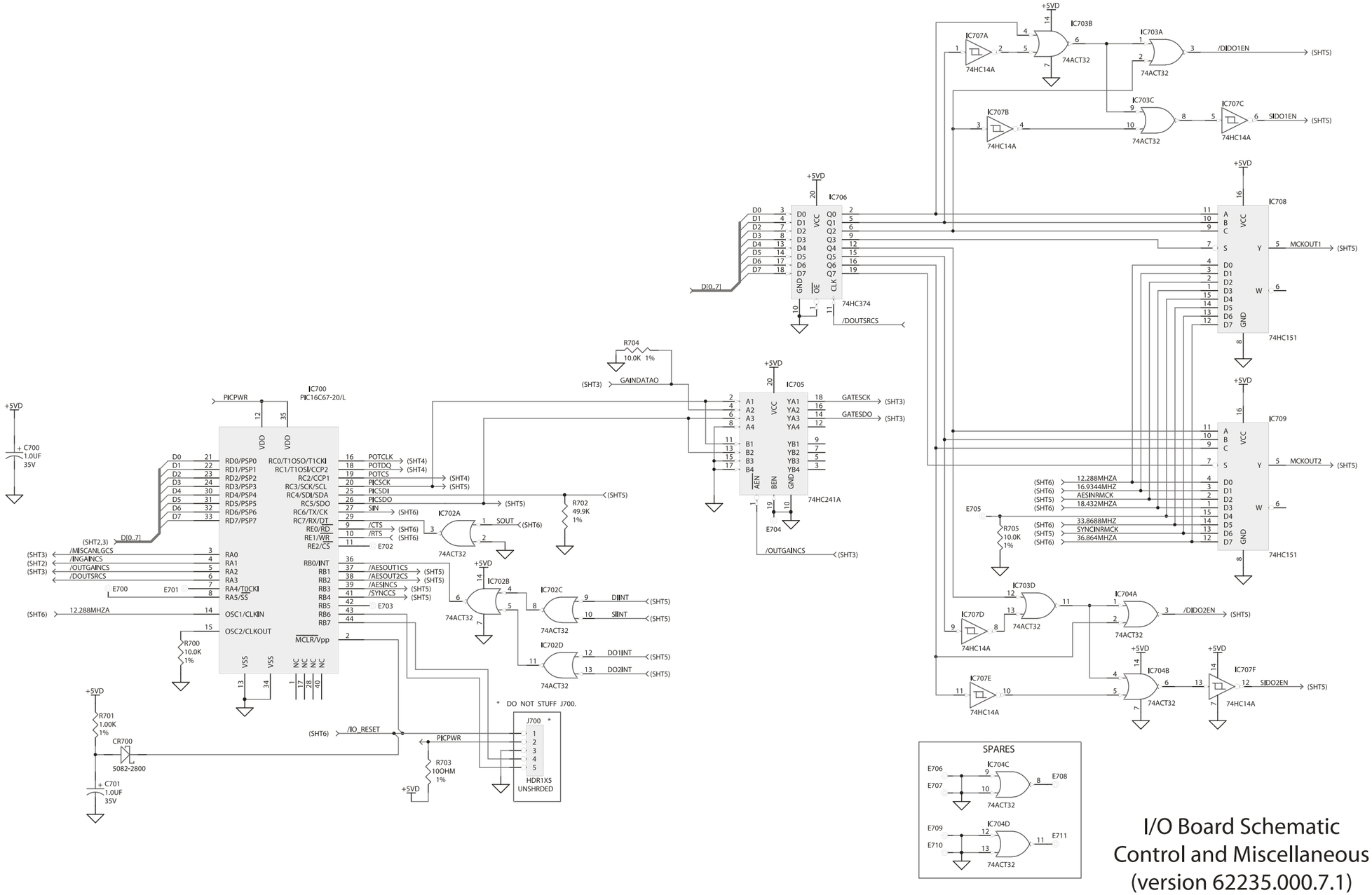
I/O Board Schematic
L and R Analog Outputs
(version 62235.000.7.1)
Sheet 2 of 6



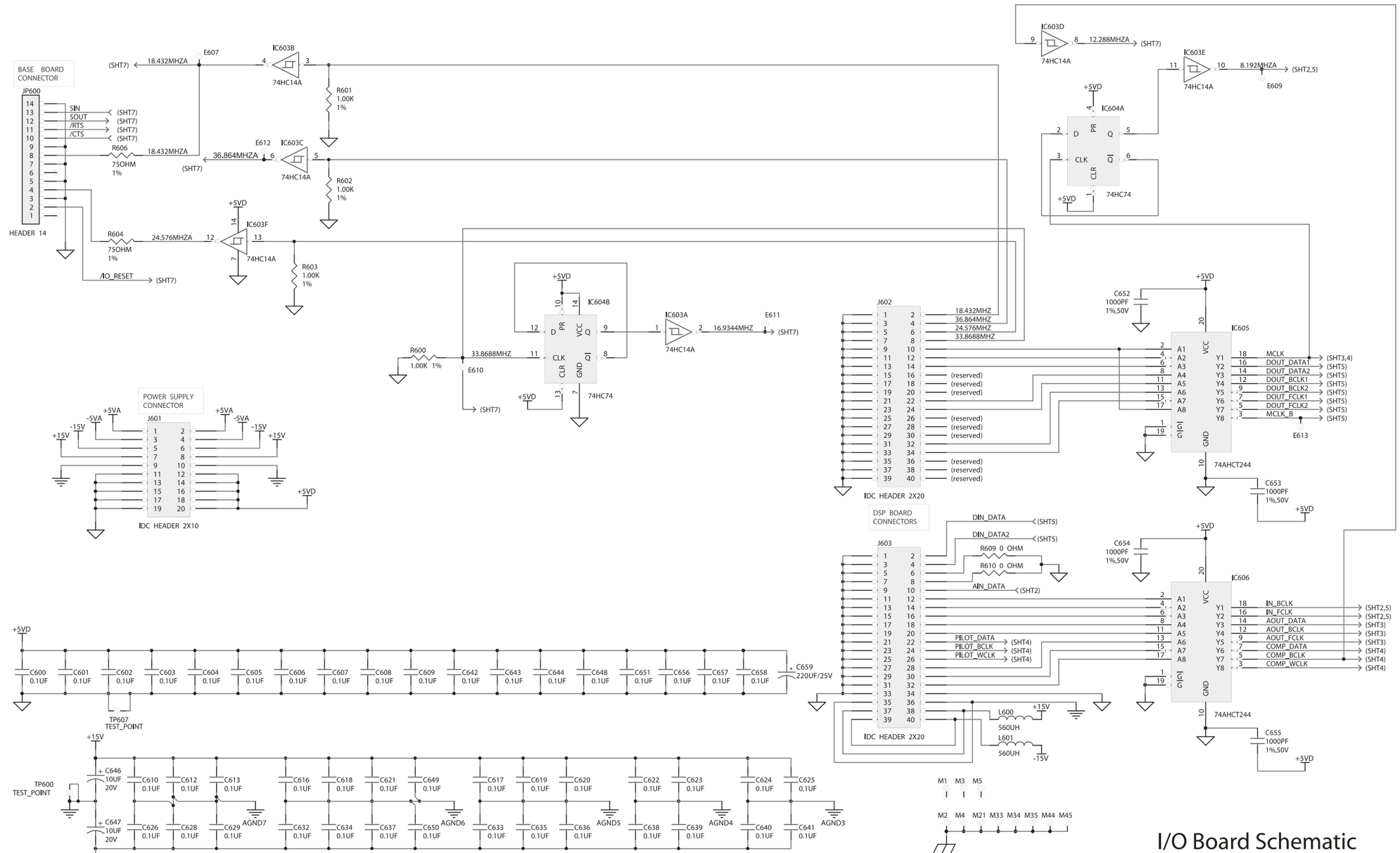
I/O Board Schematic
Composite and SCA
(version 62235.000.7.1)
Sheet 3 of 6



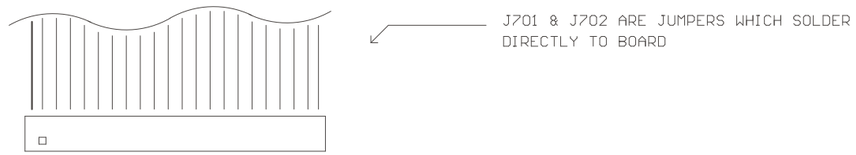
I/O Board Schematic
Digital Input/Output
(version 62235.000.7.1)
Sheet 4 of 6



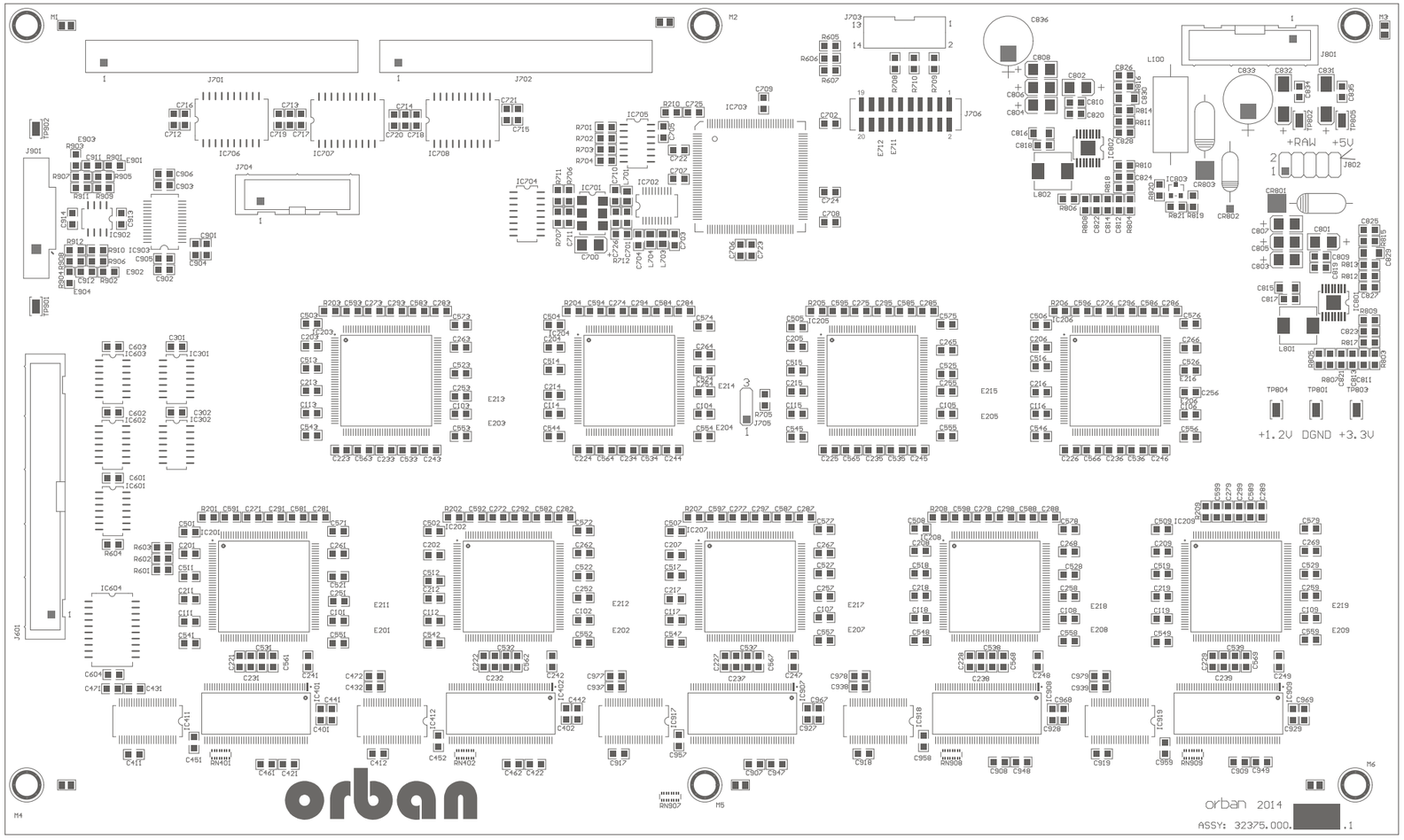
I/O Board Schematic
Control and Miscellaneous
(version 62235.000.7.1)
Sheet 5 of 6



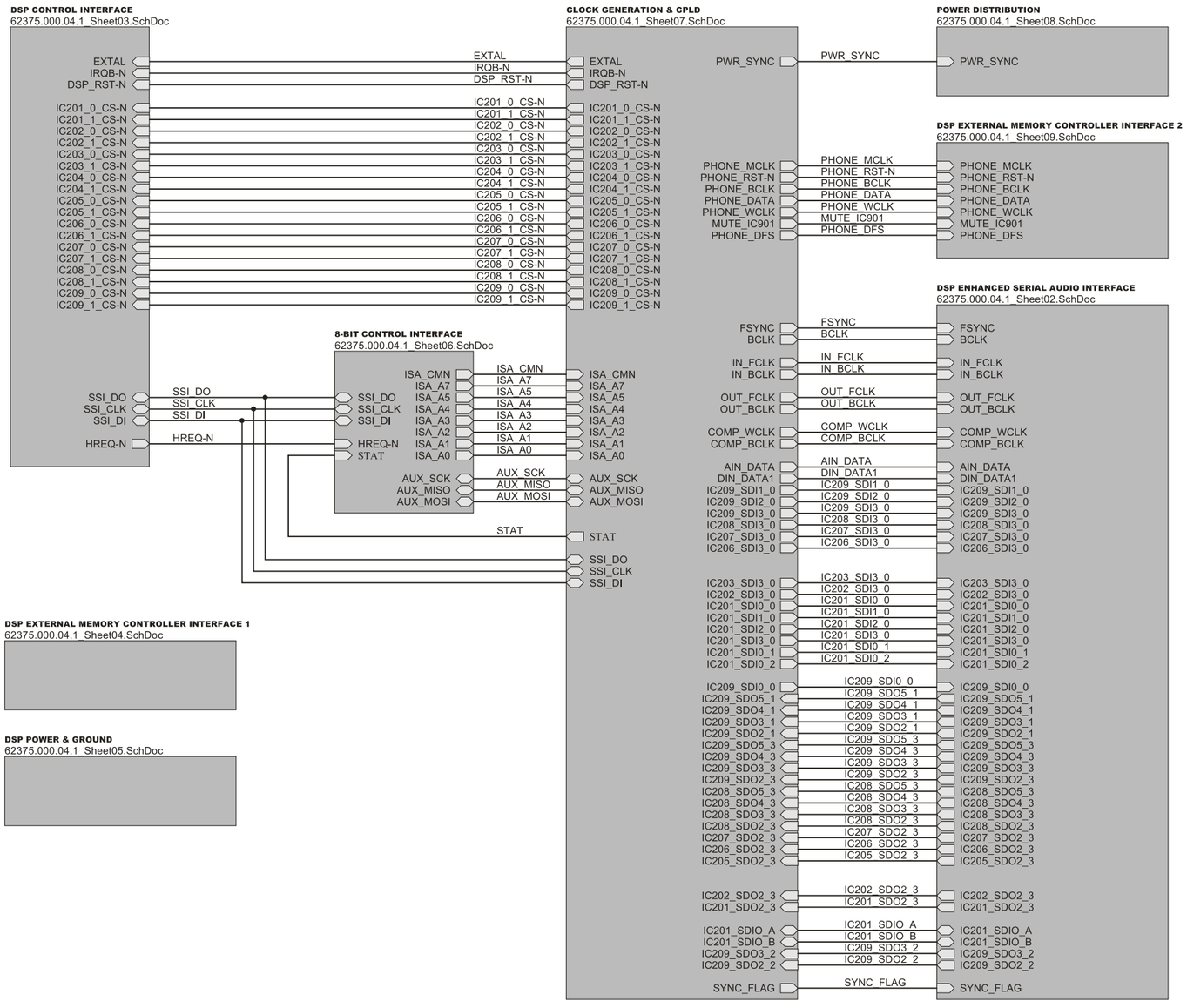
I/O Board Schematic
Interface and Power Distribution
(version 62235.000.7.1)
Sheet 6 of 6



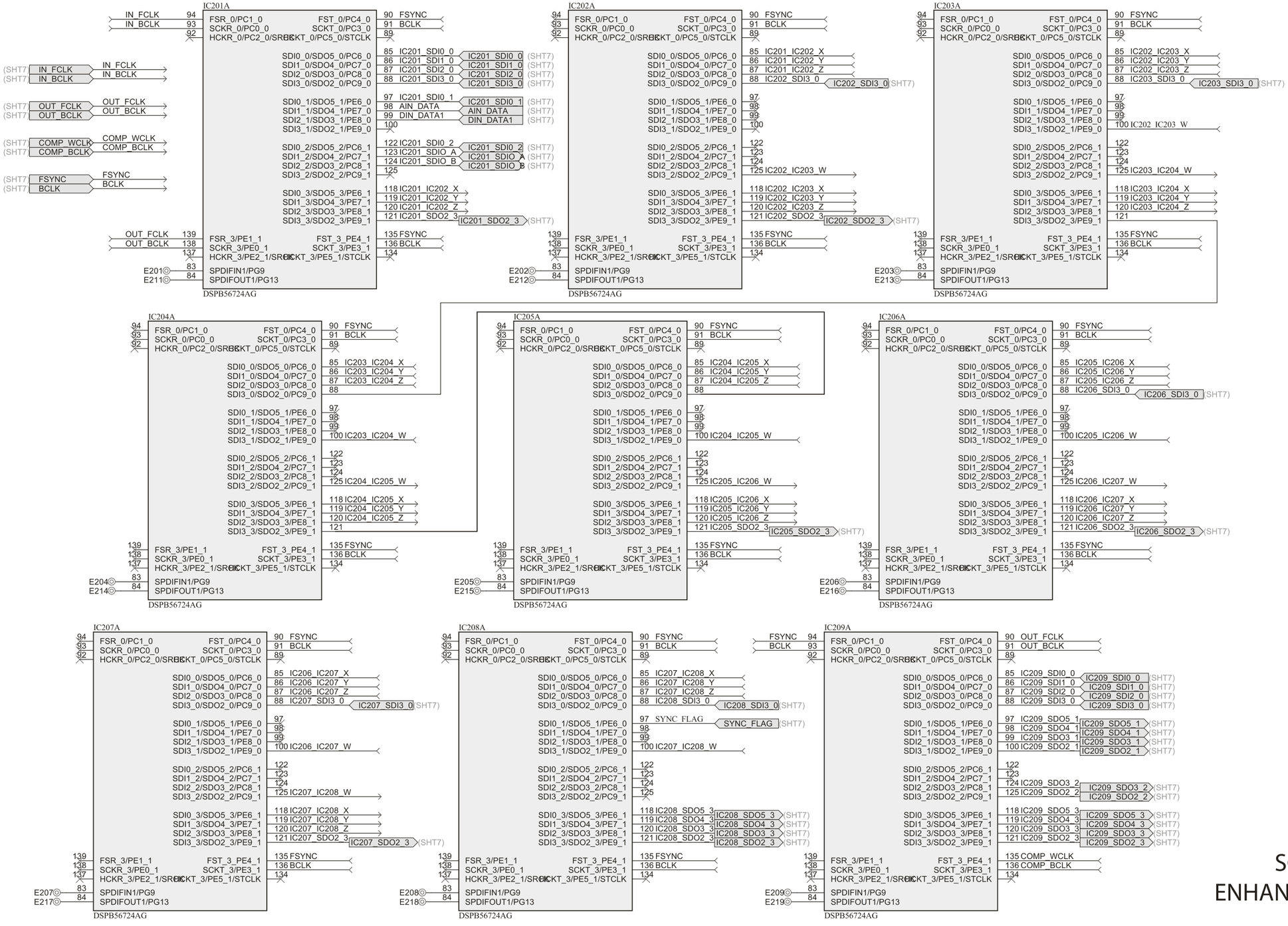
J701 & J702 ARE JUMPERS WHICH SOLDER DIRECTLY TO BOARD



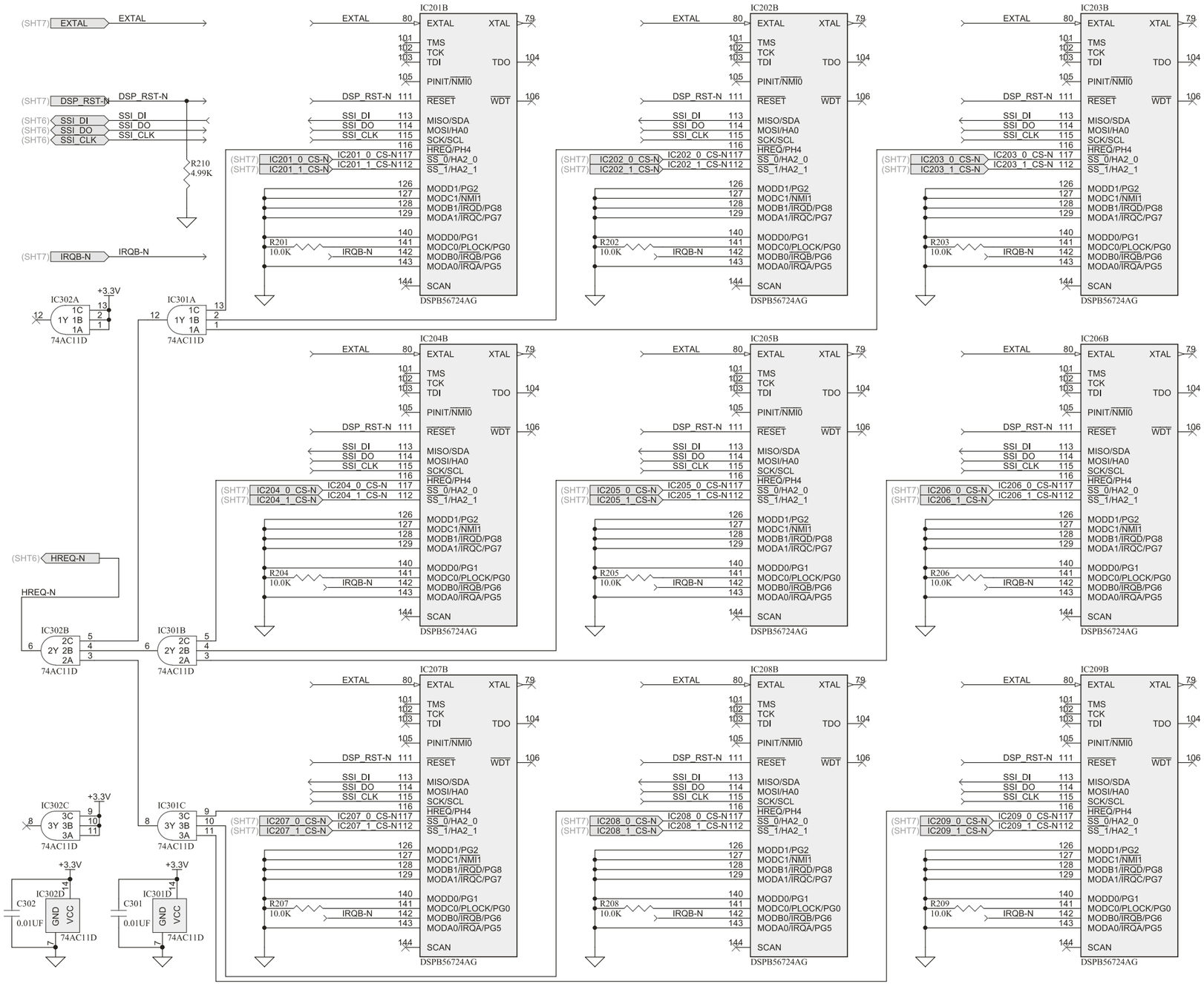
DSP Board
Parts Locator Diagram
(version 62375.000.04.1)
Sheet 1 of 1



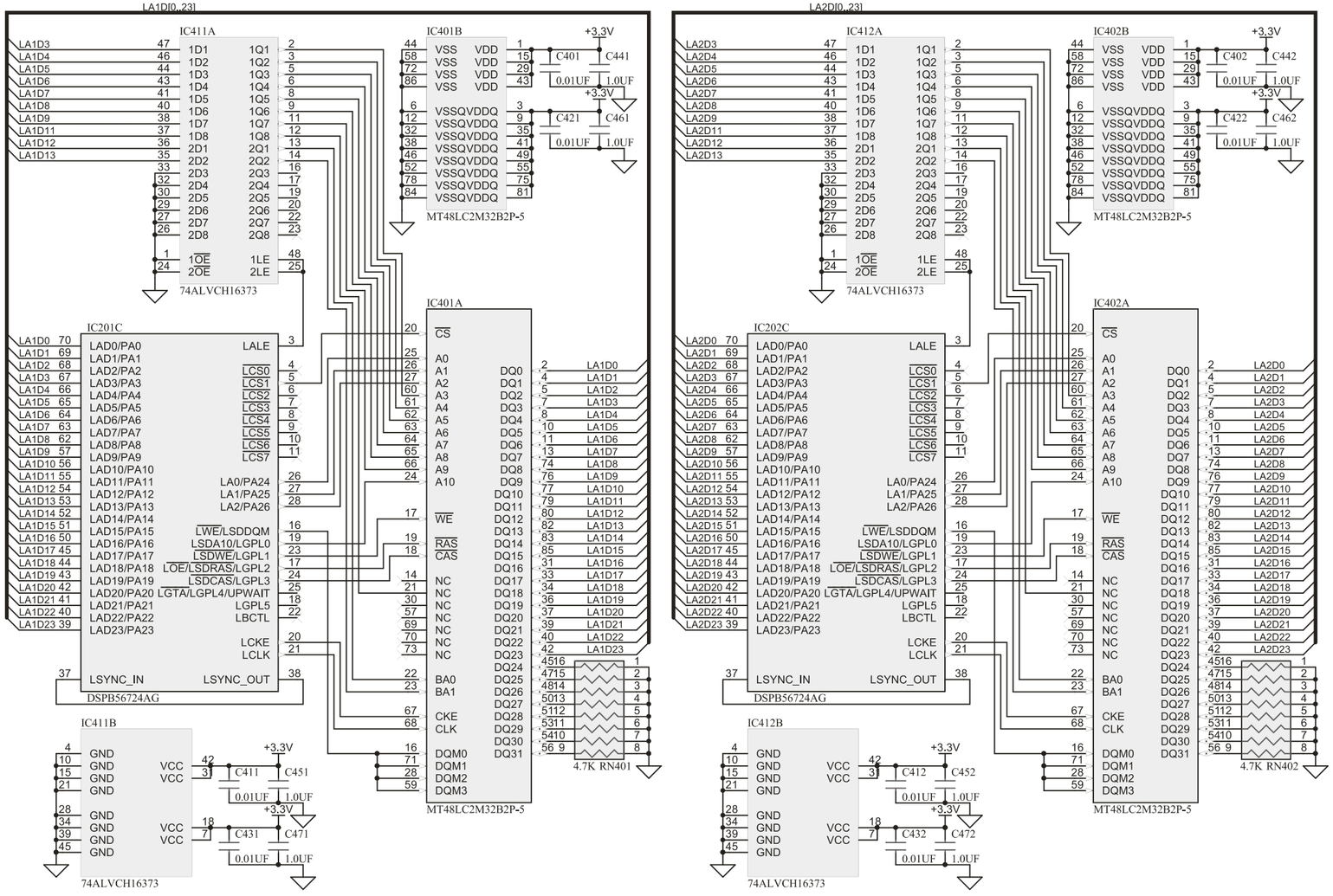
DSP Board Schematic
Interconnects (version 62375.000.04.1) Sheet 1 of 9

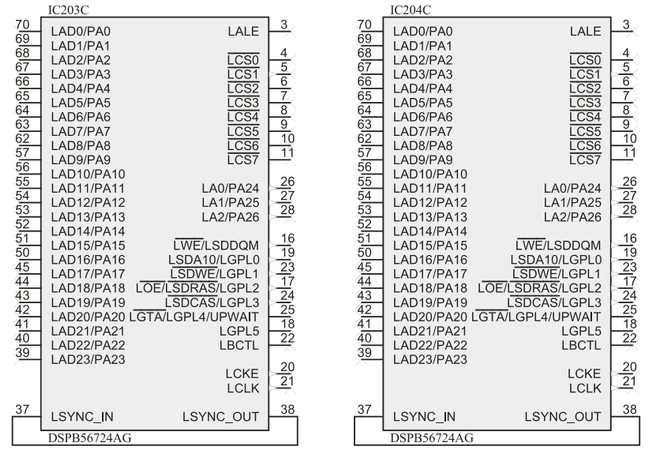


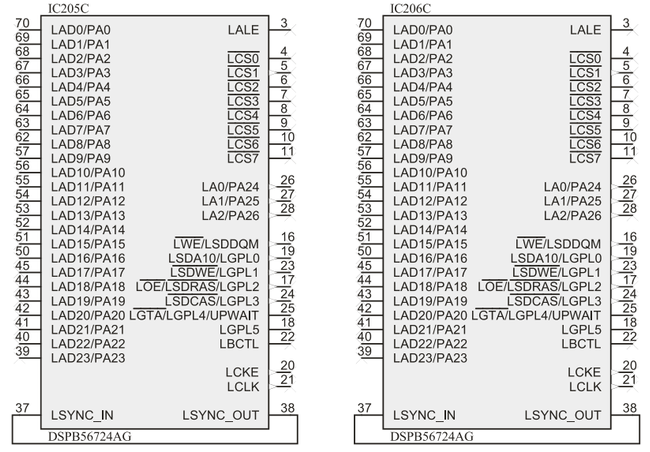
DSP BOARD
SCHEMATIC 2 OF 9
ENHANCED SERIAL INTERFACE
62375.000.xx.1



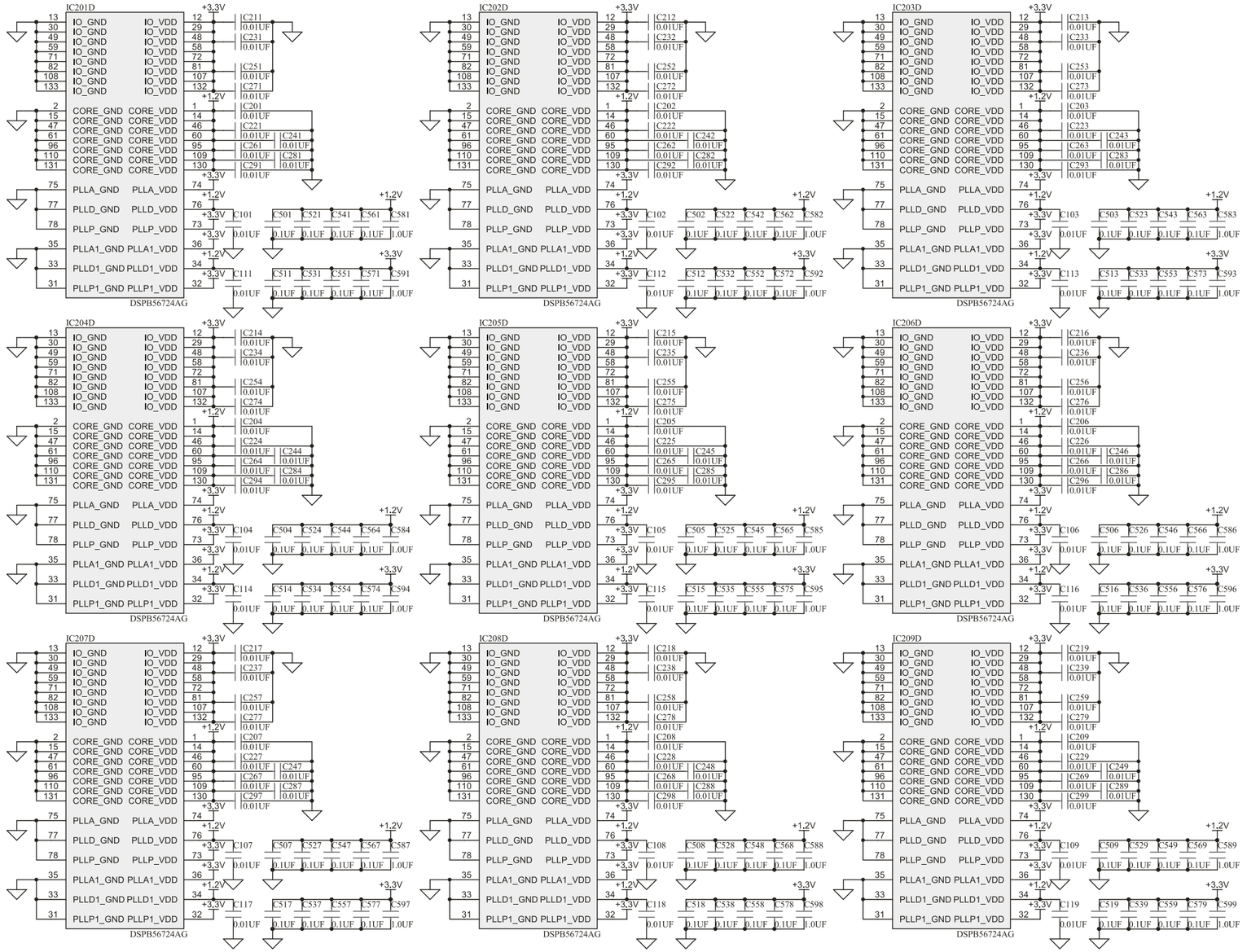
DSP Board Schematic
DSP Control Interface
(version 62375.000.04.1)
Sheet 3 of 9



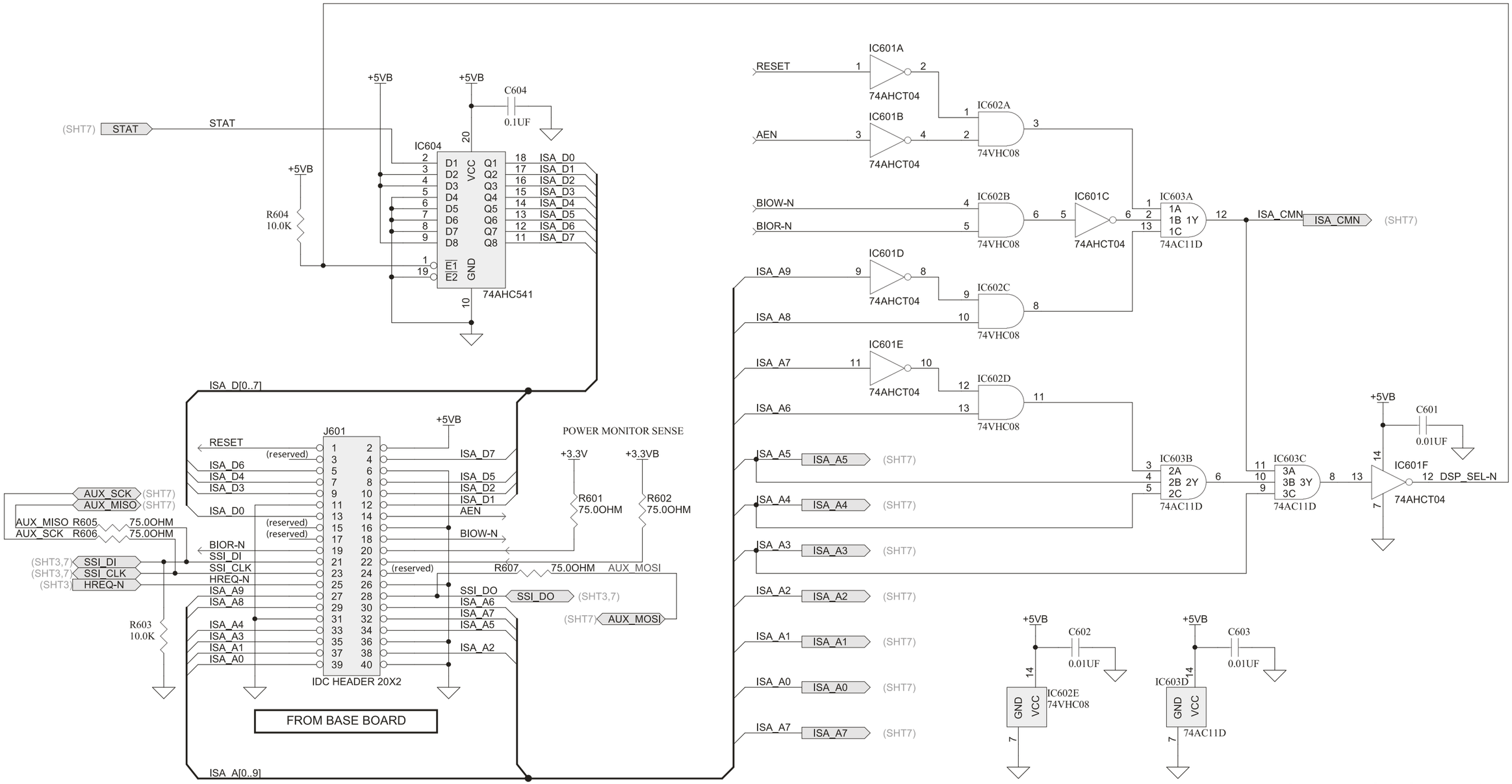




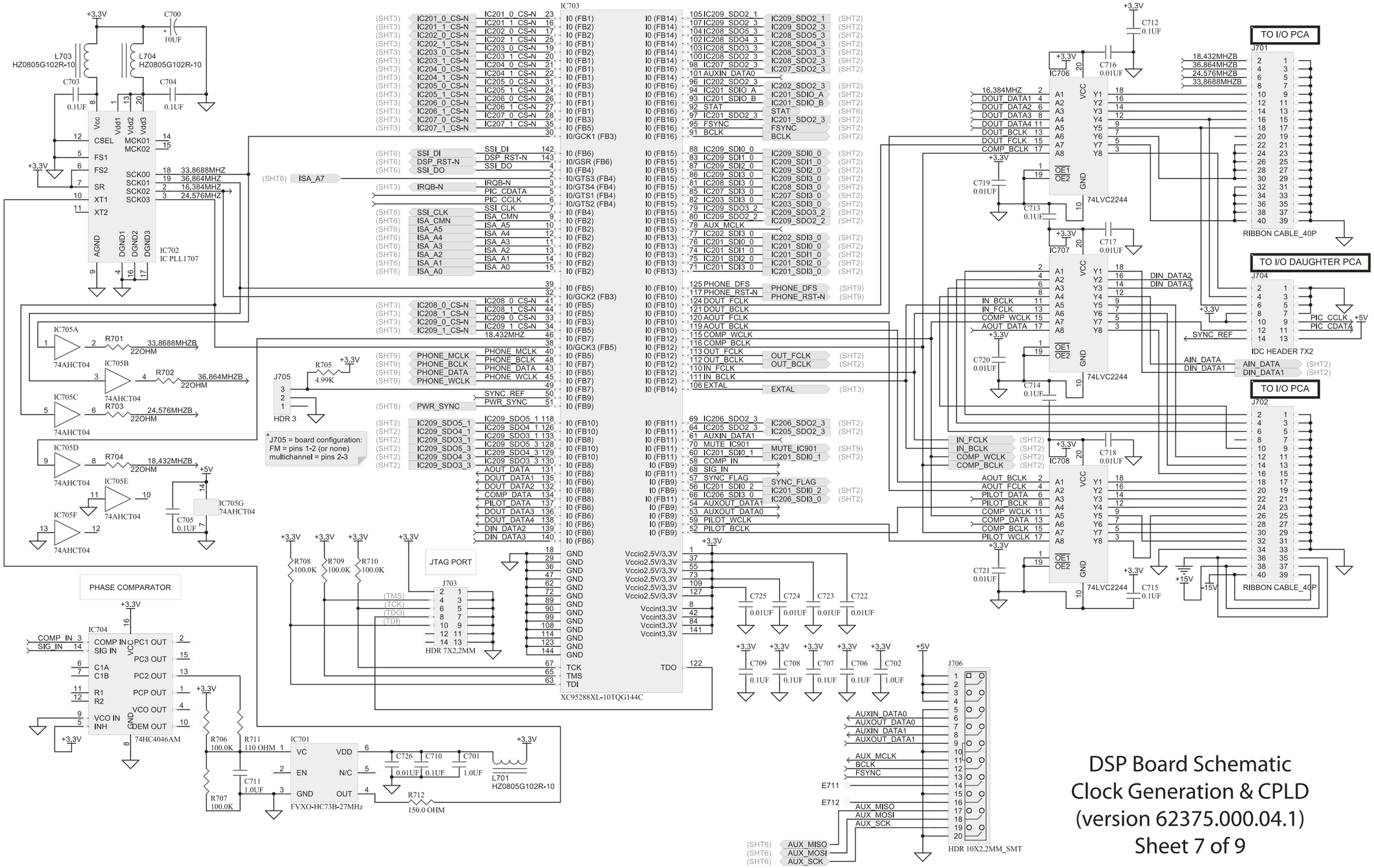
DSP Board Schematic
External Memory Controller Interface
(version 62375.000.04.1)
Sheet 4 of 9

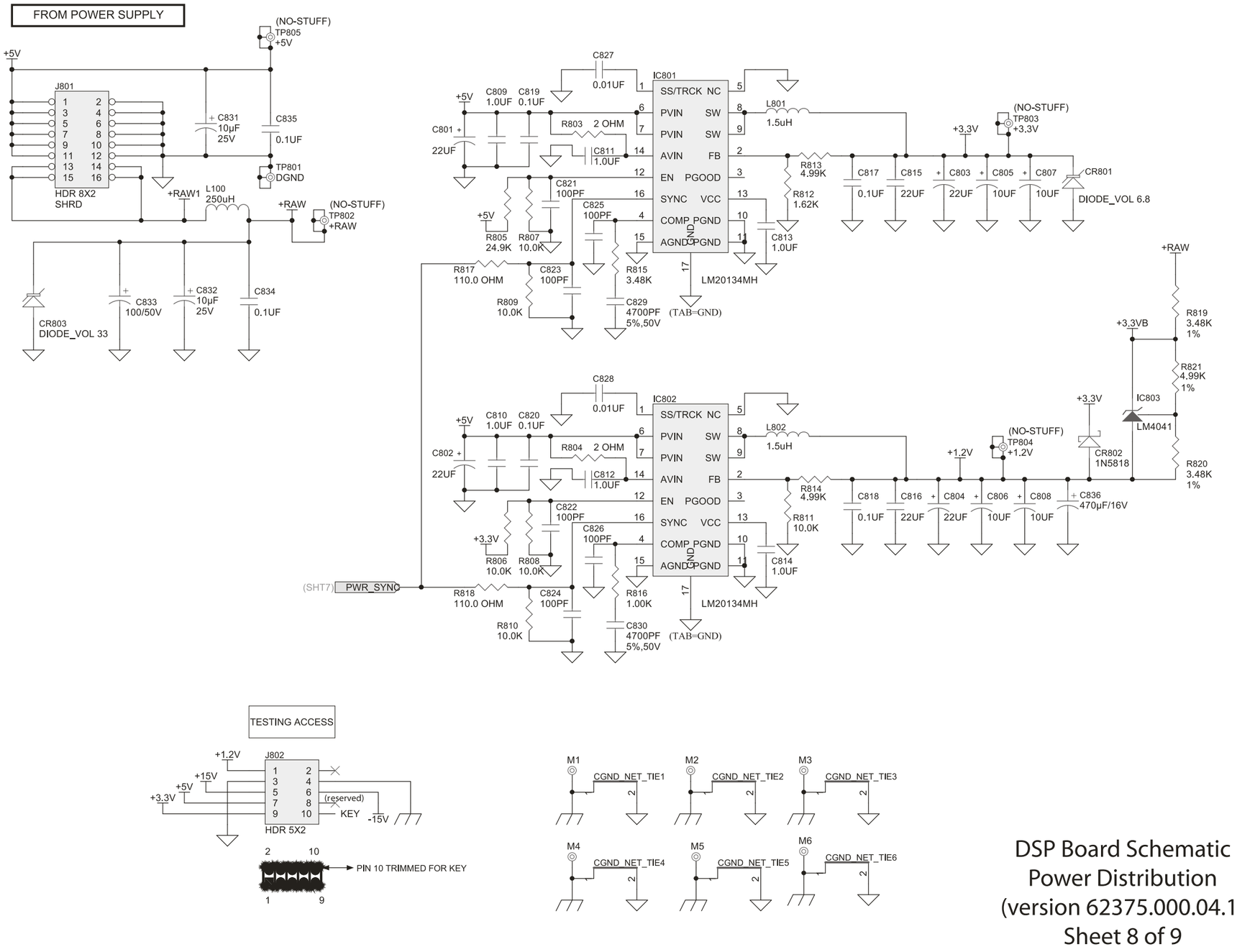


DSP Board Schematic
Power and Ground
(version 62375.000.04.1)
Sheet 5 of 9

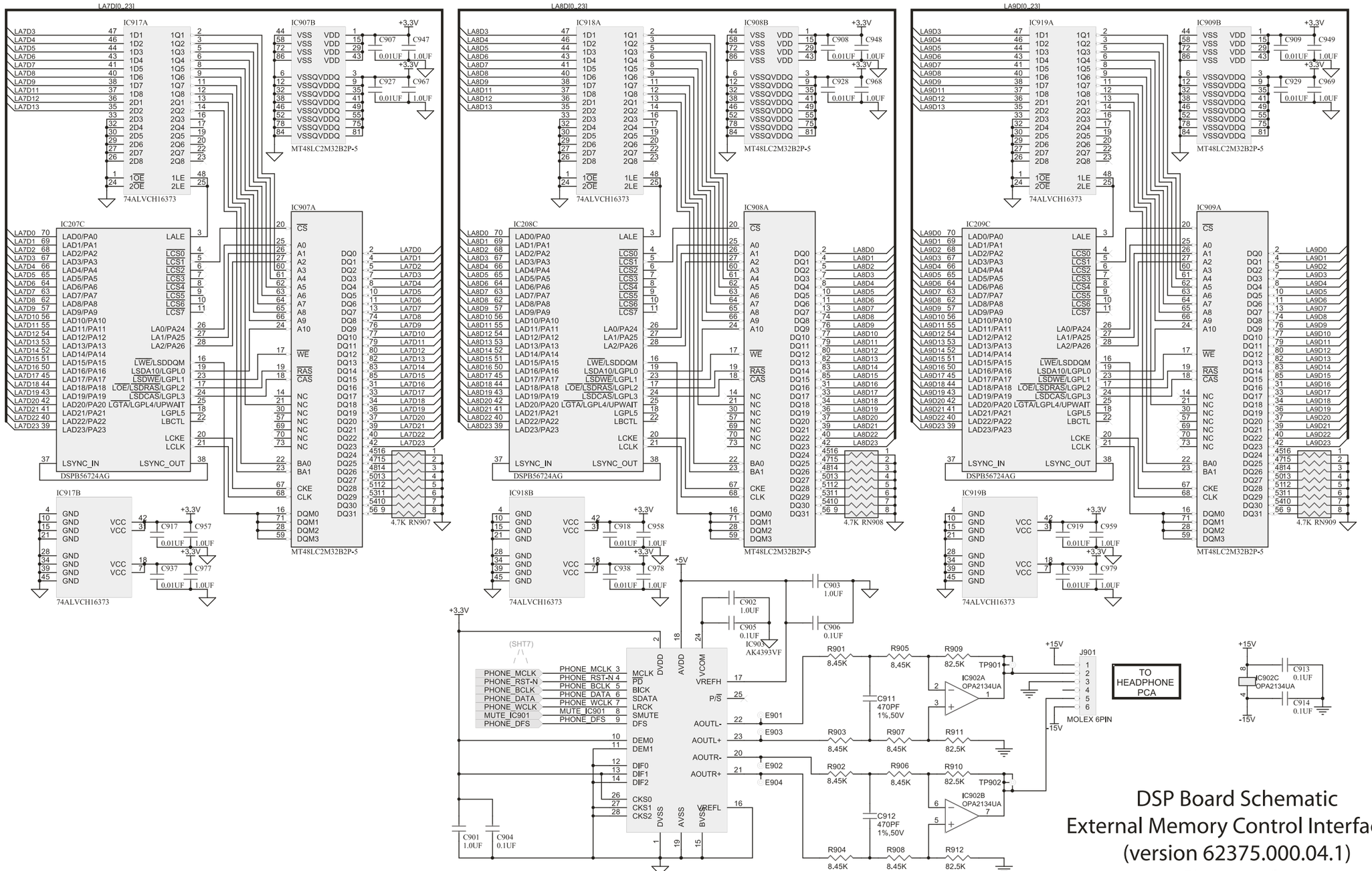


DSP Board Schematic
86xx 8-Bit Control Interface
(version 62375.000.04.1)
Sheet 6 of 9

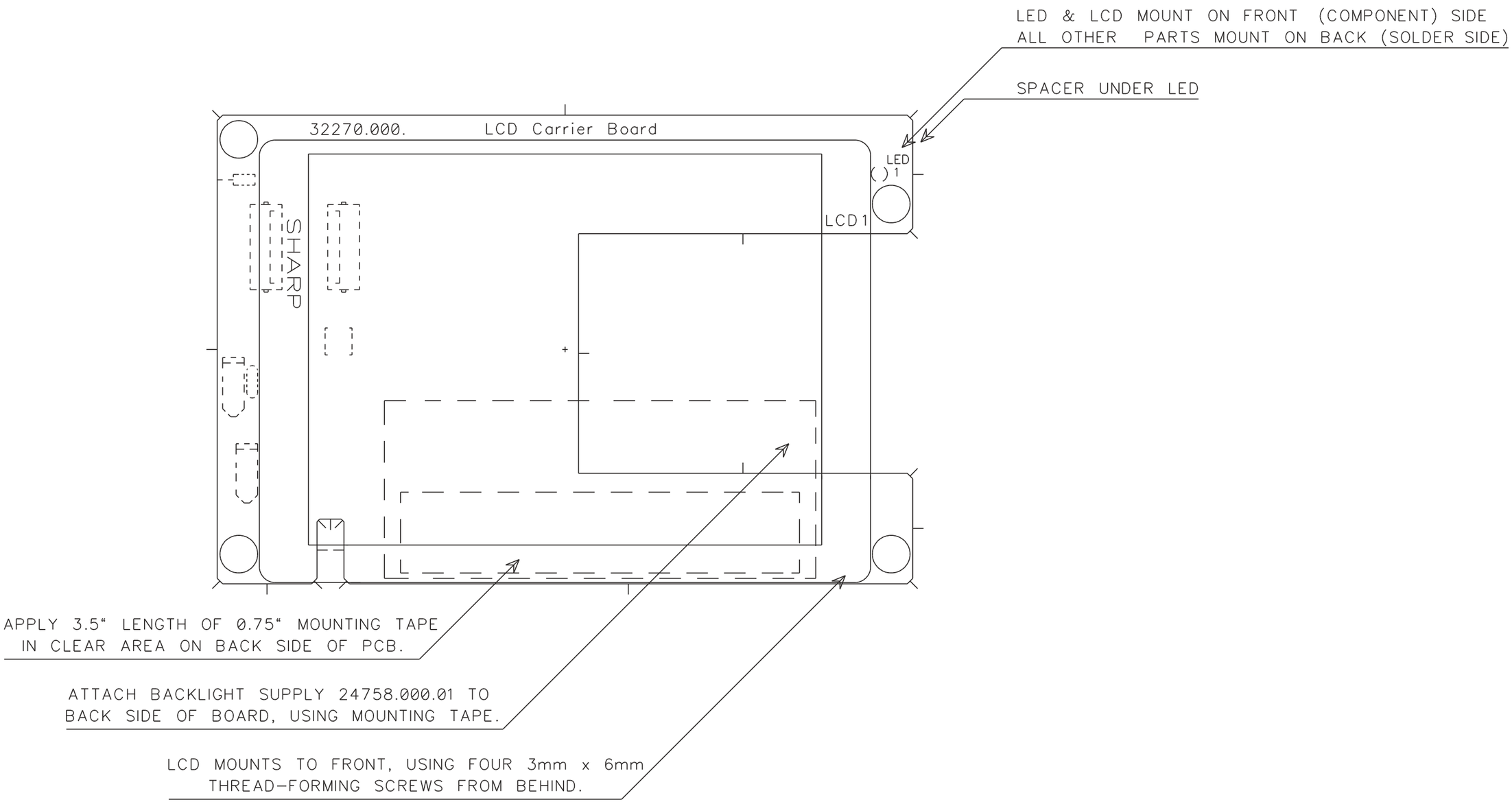

IC703
XC95288XL-10TQG144C
IC702
IC PLL1707
IC704
74HC4046AM
IC701
FVXO-HC73B-27MHz
PHASE COMPARATOR
JTAG PORT
HDR 3
HDR 7X2,2MM
HDR 10X2,2MM_SMT
RIBBON CABLE_40P
IDC HEADER 7X2
TO I/O PCA
TO I/O DAUGHTER PCA
74LVC2244
74AHCT04
J705 = board configuration:
FM = pins 1-2 (or none)
multichannel = pins 2-3
DSP Board Schematic
Clock Generation & CPLD
(version 62375.000.04.1)
Sheet 7 of 9



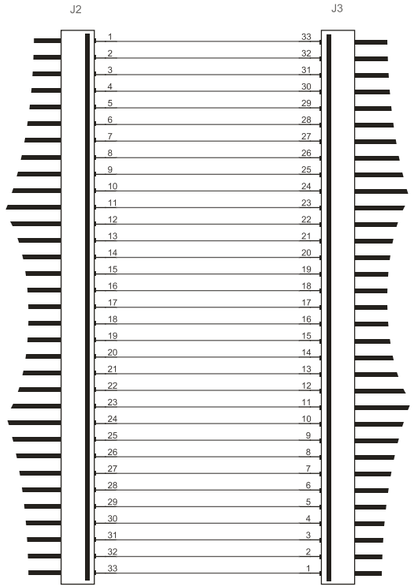
DSP Board Schematic
Power Distribution
(version 62375.000.04.1)
Sheet 8 of 9



DSP Board Schematic
External Memory Control Interface 2
(version 62375.000.04.1)
Sheet 9 of 9

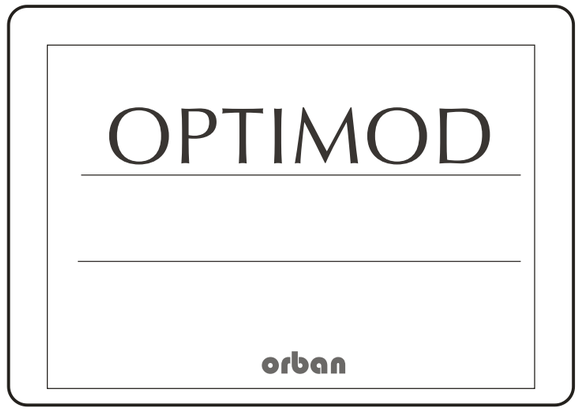


LCD CARRIER PARTS LOCATOR
32270.000



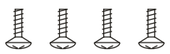
Flex Jumper

LCD1



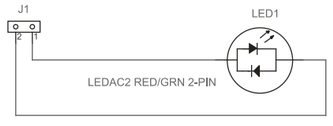
Sharp LCY-99073B-17

Revision History:				62270.000.02	
ECO#	DATE	REV	DESCRIPTION	DONE	CHECKED
3174	03/08/04	01	First Cut Released	WRS	
3200	08/27/04	02	LCD Rotation, add ZIFs, Move LED Connector	WRS	

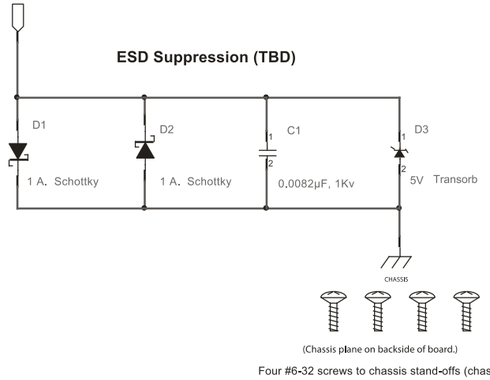


Four M-3 screws to LCD frame
(copper plane on front, under the LCD).

(Mounts From Backside)



Power Supply Monitor LED

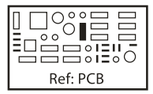


(Chassis plane on backside of board.)

Four #6-32 screws to chassis stand-offs (chassis ground).

Backlight Supply
SIPF-200A

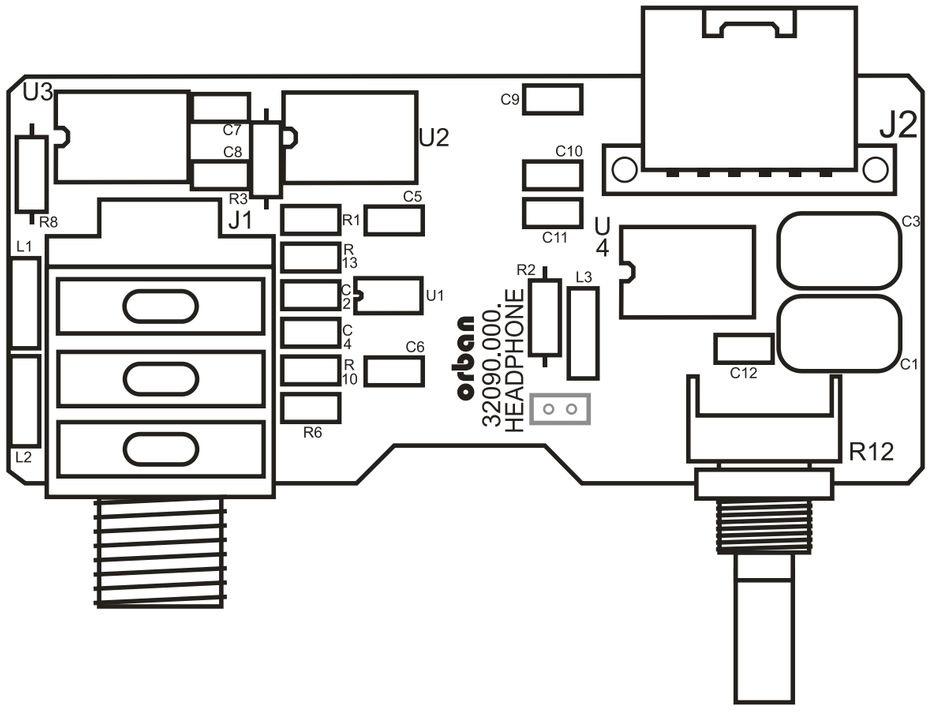
(Backlight power supply module mounts to REAR
of Carrier Board with double-stick tape.)



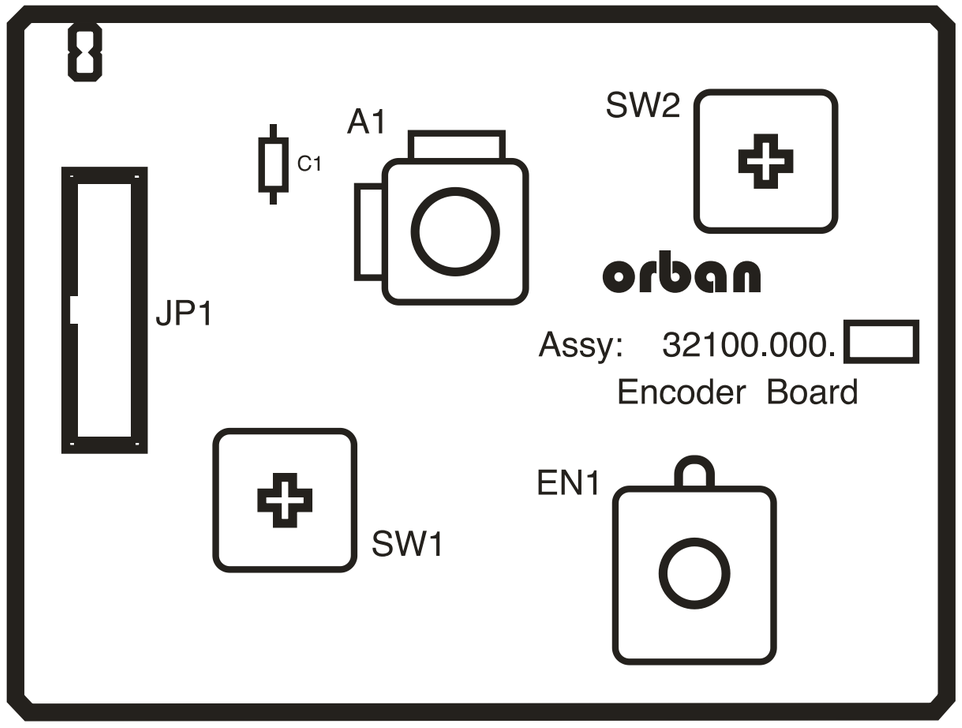
FAB 32271.000.02

* NOTE: All Components Except LED & LCD Mount
To Backside Of PCB.

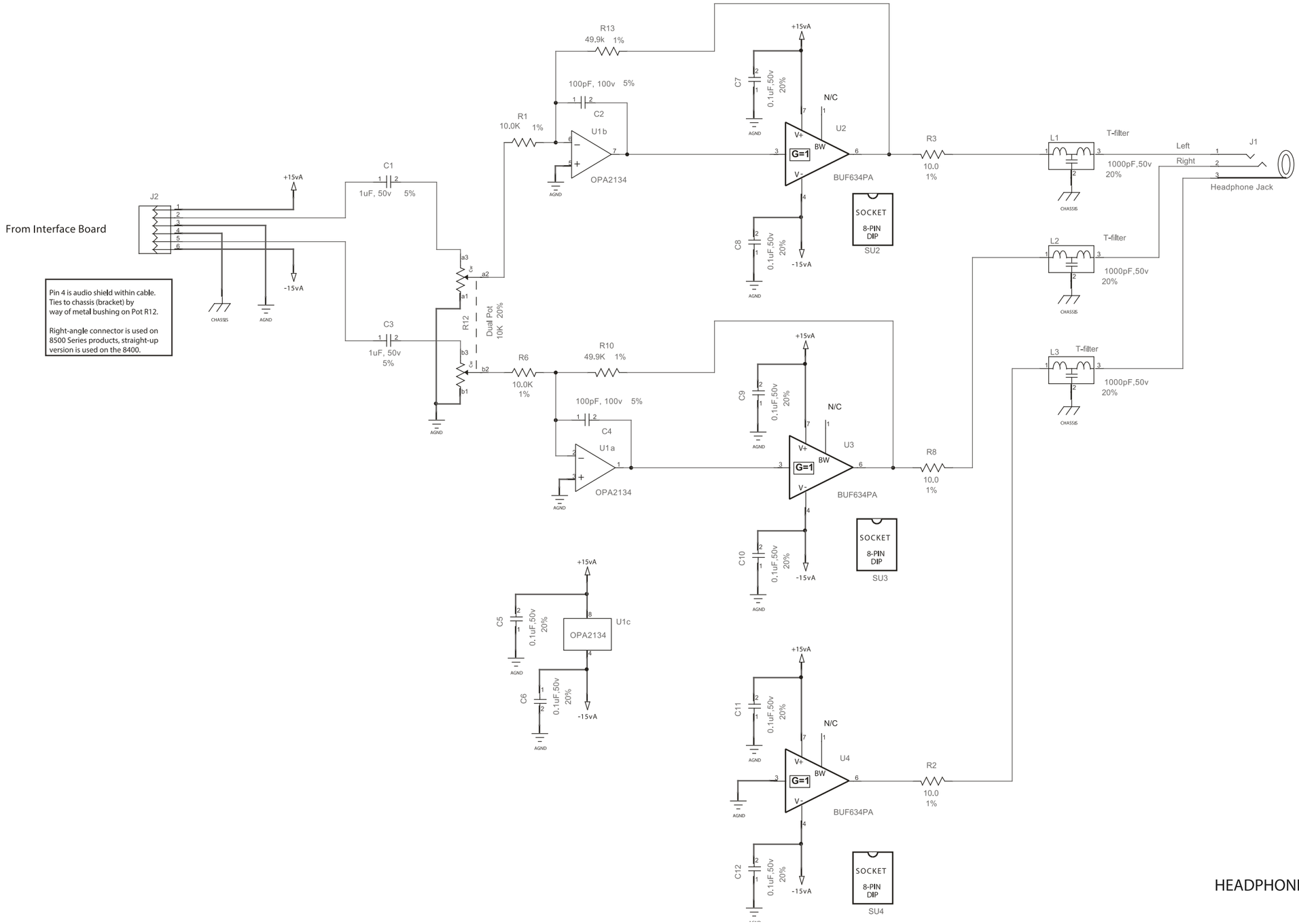
LCD CARRIER BOARD SCHEMATIC
1 of 1



HEADPHONE BOARD PARTS LOCATOR

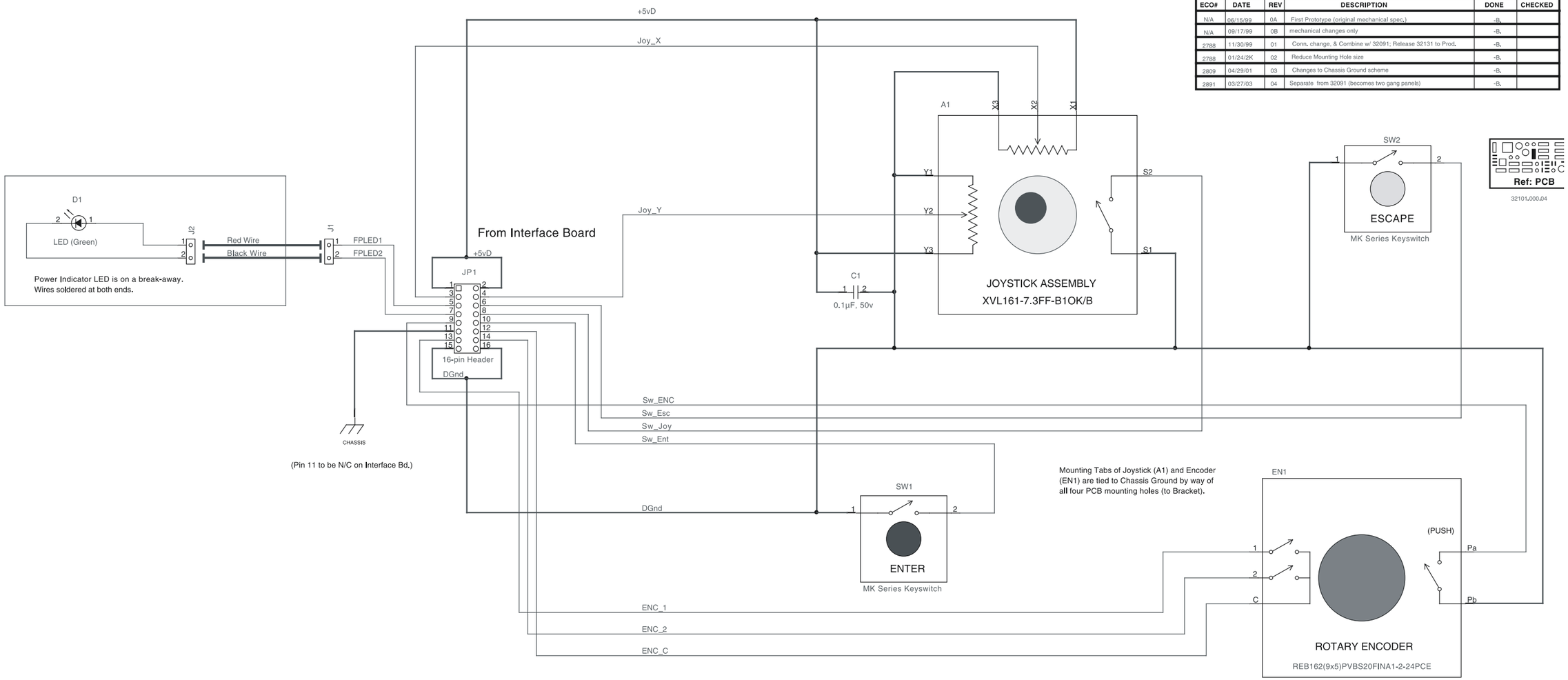


ENCODER BOARD PARTS LOCATOR

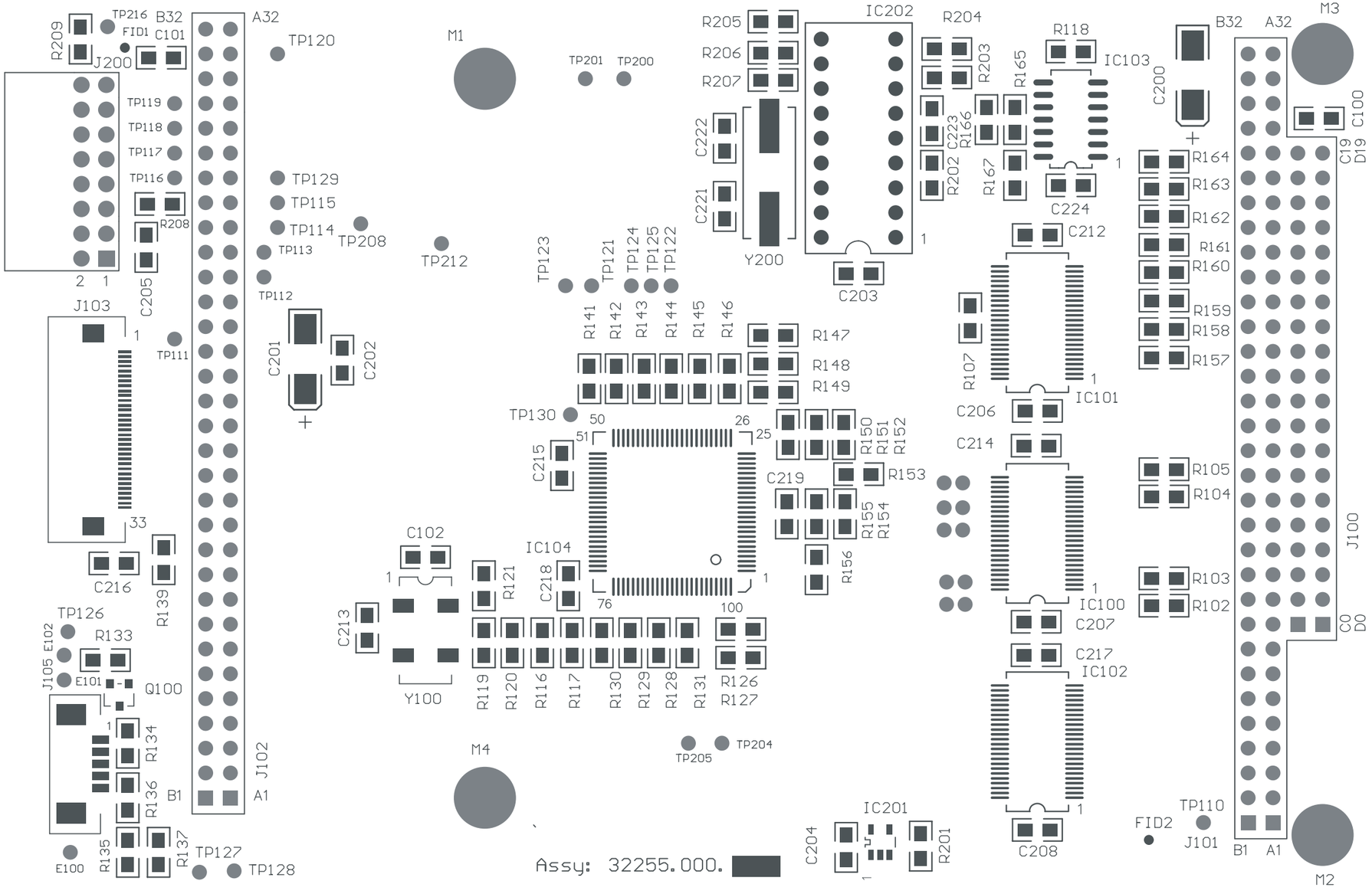


HEADPHONE BOARD SCHEMATIC
1 of 1

Revision History:				62100.000.04	
ECO#	DATE	REV	DESCRIPTION	DONE	CHECKED
N/A	06/15/99	0A	First Prototype (original mechanical spec.)	-B.	
N/A	09/17/99	0B	mechanical changes only	-B.	
2788	11/30/99	01	Conn. change, & Combine w/ 32091; Release 32131 to Prod.	-B.	
2788	01/24/2K	02	Reduce Mounting Hole size	-B.	
2809	04/29/01	03	Changes to Chassis Ground scheme	-B.	
2891	03/27/03	04	Separate from 32091 (becomes two gang panels)	-B.	



ENCODER BOARD SCHEMATIC
1 of 1

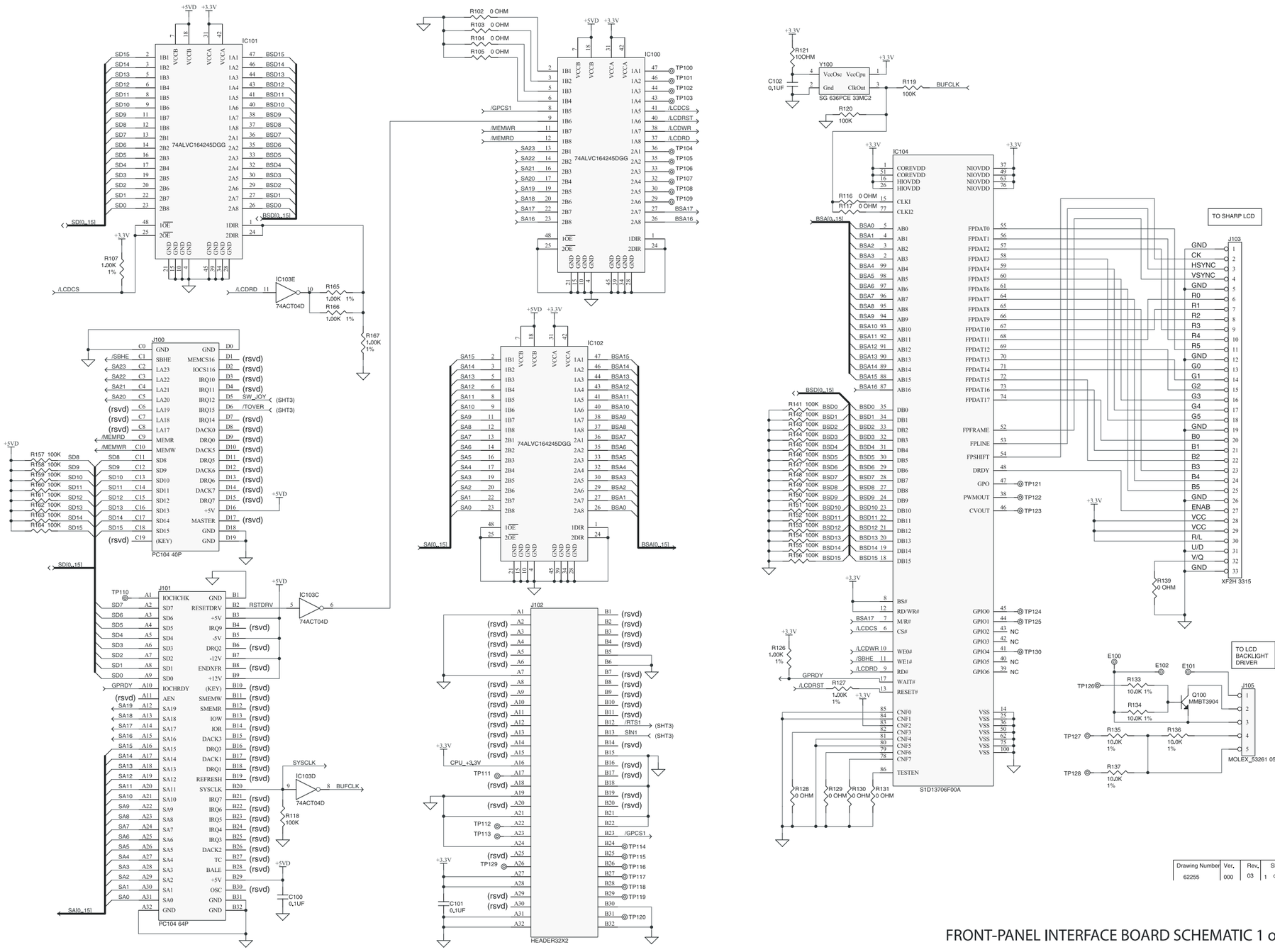


3. MARK ASSEMBLY REVISION LEVEL IN SPACE PROVIDED.
2. REFERENCE SCHEMATIC DRAWING NO. 62255.000
1. SQUARE PADS INDICATE PIN 1 OF CONNECTORS, CATHODE OF DIODES, POS. SIDE OF CAPS. PIN 1 OF IC'S.

NOTES: (UNLESS OTHERWISE SPECIFIED)

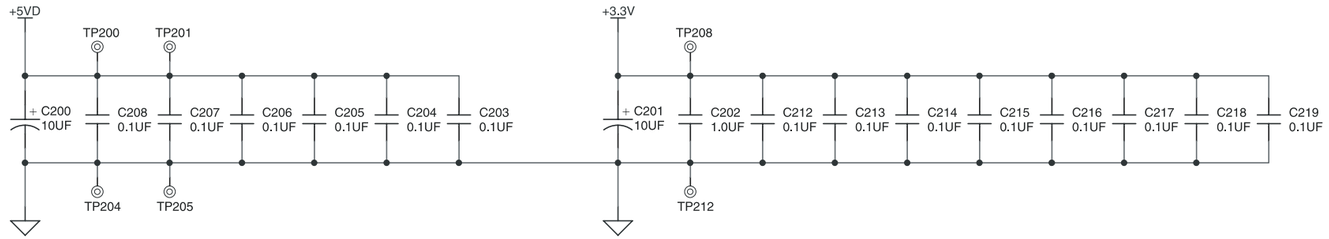
SIZE	DWG NO.	VER	REV	SHEET
C	32255	000	03	1 OF 1

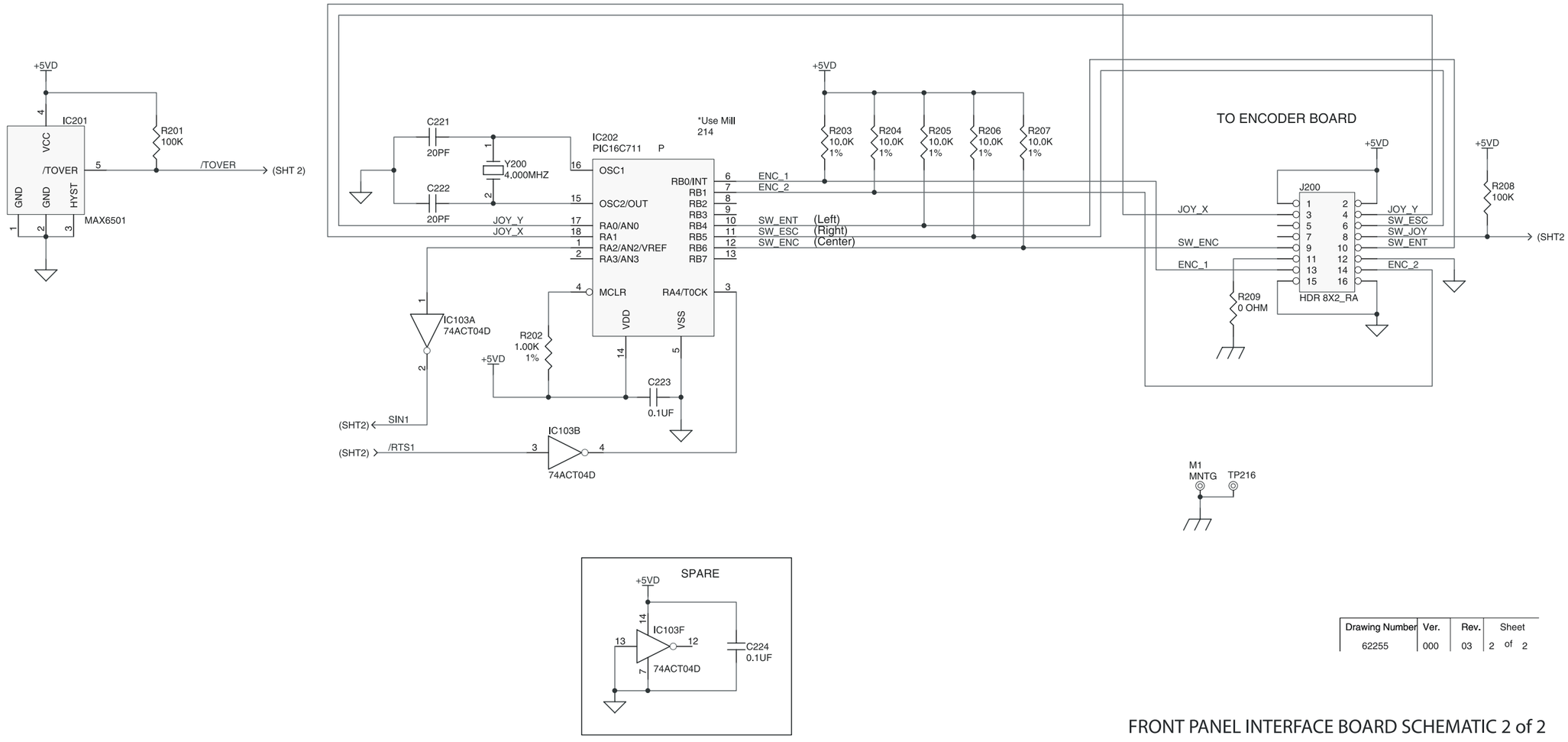
FRONT-PANEL INTERFACE BOARD PARTS LOCATOR DRAWING



Drawing Number	Ver.	Rev.	Sheet
62255	000	03	1 of 2

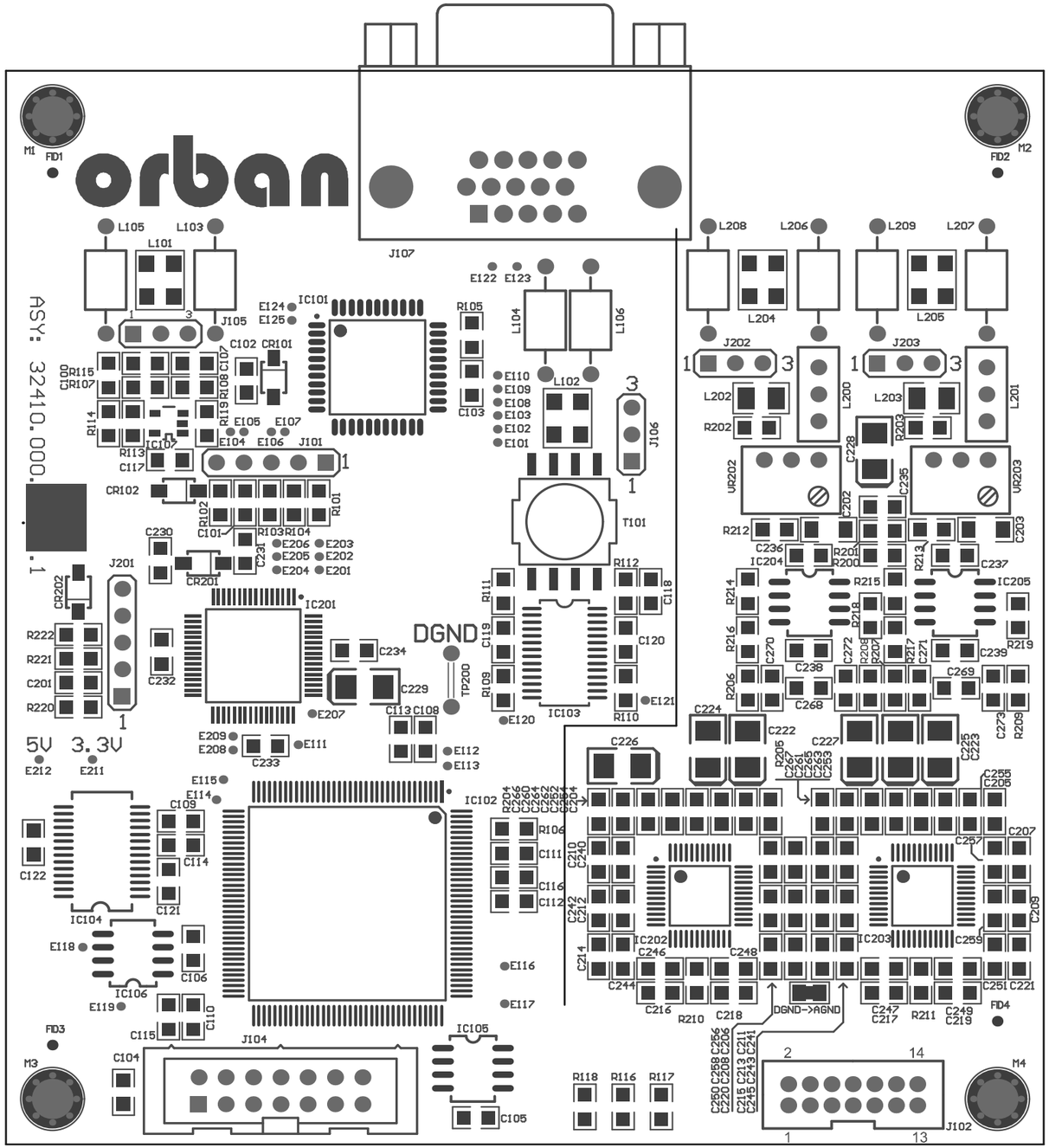
FRONT-PANEL INTERFACE BOARD SCHEMATIC 1 of 2



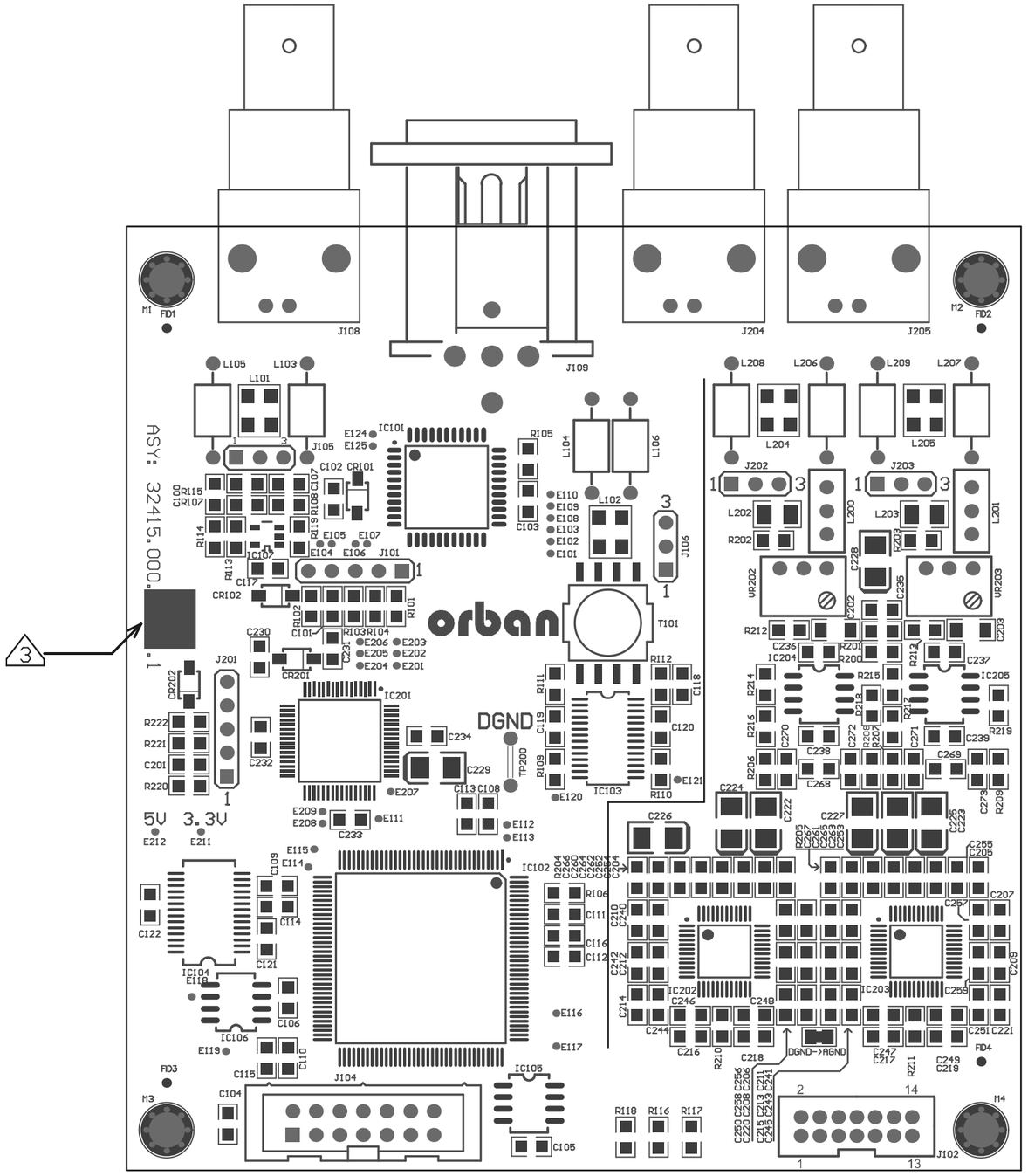


Drawing Number	Ver.	Rev.	Sheet
62255	000	03	2 of 2

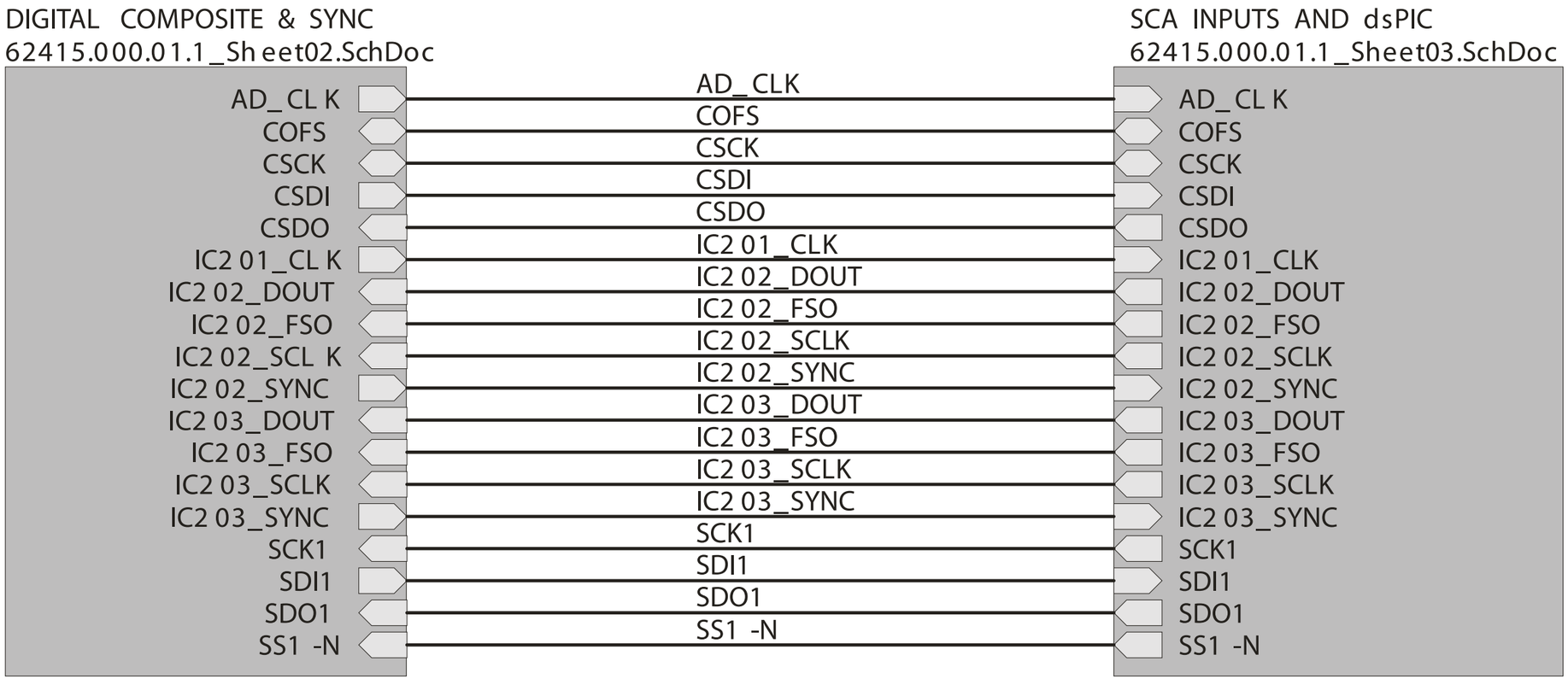
FRONT PANEL INTERFACE BOARD SCHEMATIC 2 of 2



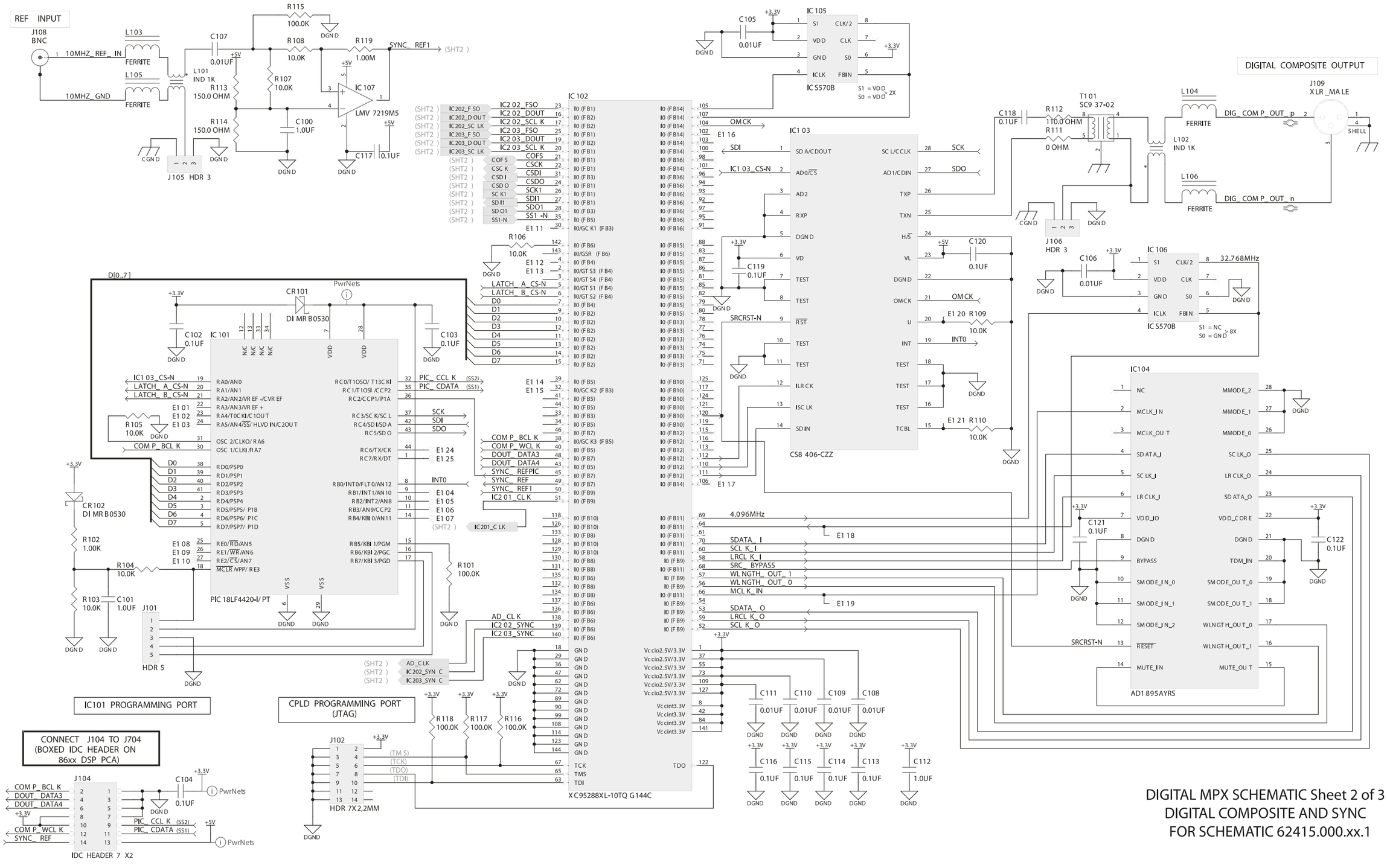
DIGITAL MPX RETROFIT PARTS LOCATOR DIAGRAM
FOR SCHEMATIC 62415.000.xx.1



DIGITAL MPX PARTS LOCATOR DIAGRAM
FOR SCHEMATIC 62415.000.xx.1



DIGITAL MPX SCHEMATIC Sheet 1 of 3
INTERCONNECTS
FOR SCHEMATIC 62415.000.xx.1



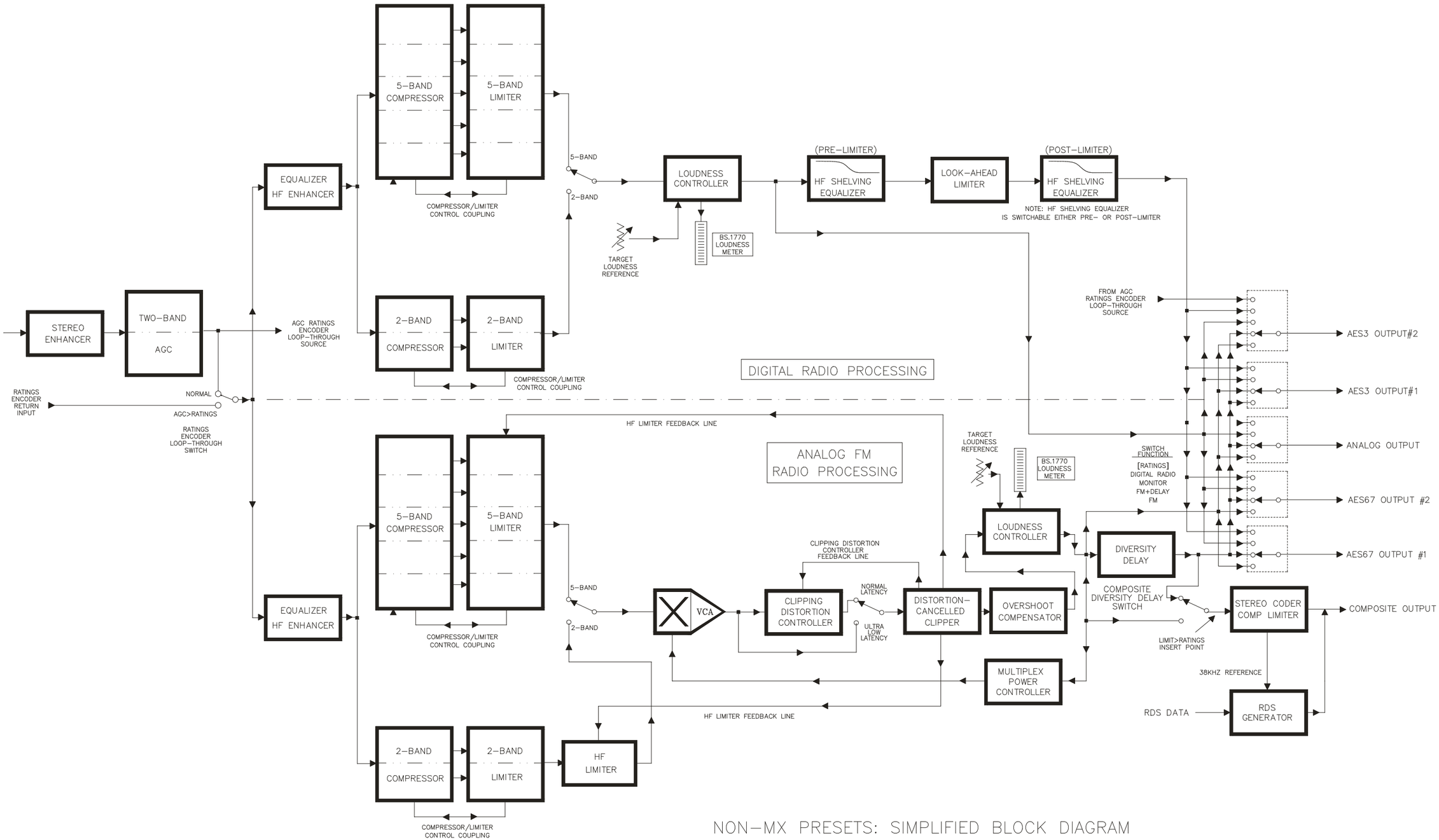
DIGITAL MPX SCHEMATIC Sheet 2 of 3
DIGITAL COMPOSITE AND SYNC
FOR SCHEMATIC 62415.000.xx.1

SCA INPUT 1

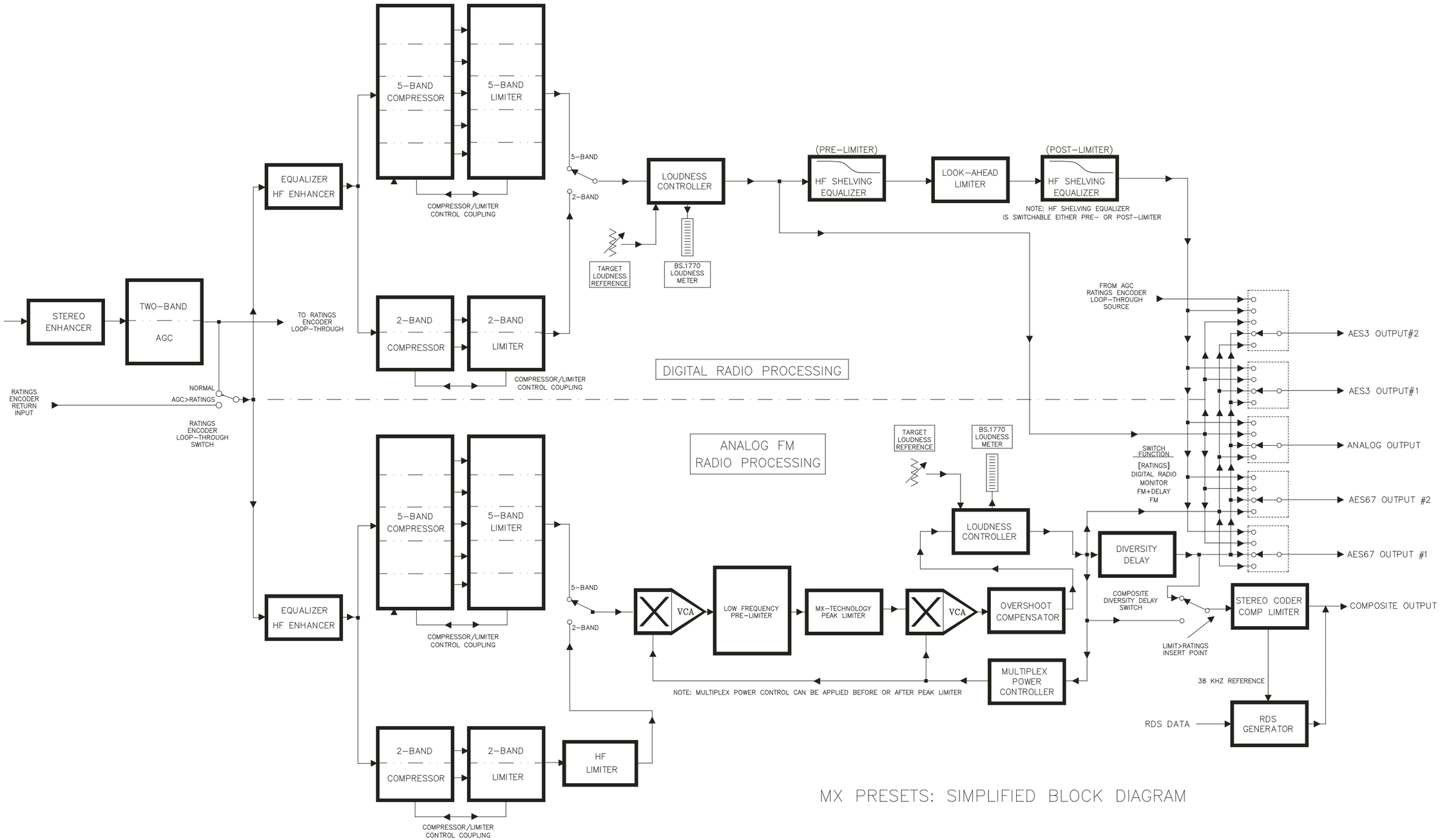
SCA INPUT 2

IC201 PROGRAMMING PORT

DIGITAL MPX SCHEMATIC Sheet 3 of 3
SCA INPUTS AND dsPIC
FOR SCHEMATIC 62415.000.xx.1



NON–MX PRESETS: SIMPLIFIED BLOCK DIAGRAM



MX PRESETS: SIMPLIFIED BLOCK DIAGRAM