Operating Manual

OPTIMOD 8400

Digital Audio Processor

Manual for Version 3.0 and Higher Software

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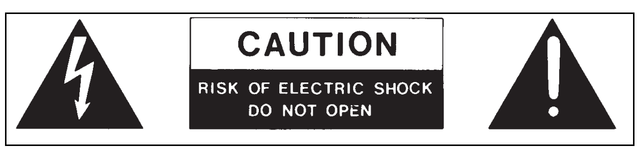
IMPORTANT NOTE: Refer to the unit's rear panel for your Model #.

Model Number:	Description:
8400	OPTIMOD 8400, Stereo Encoder, Digital I/O, Protection Structure, Two-Band Structure, Multi-Band Structure, 115V (for 90-130V operation) or 230V (for 200-250V operation), switchable to 50µs or 75µs.
8400HD FM	OPTIMOD 8400HD FM, Stereo Encoder, Digital I/O, Digital Radio and Internet Streaming Processing, Protection Structure, Two-Band Structure, Multi-Band Structure, 115V (for 90-130V operation) or 230V (for 200-250V operation), switchable to 50µs or 75µs.

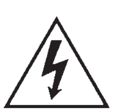
MANUAL:

Part Number:	Description:
96107-300-01	8400 Operating Manual



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.

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.

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).

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

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

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

(4) 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) 510/351-3500.

Operating Manual

OPTIMOD

8400

Digital Audio Processor

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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 item. A much smaller static discharge is likely to completely 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, many damaging discharges will not even be noticed.

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

OPTIMOD 8400 is protected by U.S. patents 4,208,548; 4,460,871; **5,737,434; 6,337,999;** 6,434,241. Other patents pending.

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P/N: 96107-300-01

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Section 1 Introduction

8400 OPTIMOD-FM Digital Audio Processor

Orban's all-digital 8400 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 Motorola DSP56362 Digital Signal Processing chips, the processing has cleanliness, quality, and stability over time and temperature that is unmatched by analog processors.

OPTIMOD-FM 8400 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 8200ST OPTIMOD-Studio Compressor/Limiter/HF Limiter/Clipper. The 8200ST can be adjusted so that it substitutes for the broadband AGC circuitry in OPTIMOD-FM, which is then defeated.

OPTIMOD-FM 8400 is available in two main configurations—the 8400 has a full-featured front panel, while the 8400/PD has a blank front panel and must be controlled by Orban's PC Remote application running on Microsoft Windows 98 or later. Both units have identical sound and features except for the difference in their front panels. Both units run the same software.

Both the 8400 and 8400/PD can accept Orban's HD ("High-Definition Digital Radio") hardware plug-in, 8400HD. 8400HD adds an extra AES/EBU output to drive the digital channel in the iBiquity system. This output provides look-ahead peak limiting that operates in parallel with the FM peak limiting. The look-ahead limiting is optimized to make the most of the limited bit-rate codec used in the iBiquity system's digital channel. By eschewing any clipping, the HD output prevents the codec from wasting precious bits encoding clipping distortion products, allowing the codec to instead use its entire bit budget to encode the desired program material.

15 kHz band-limiting on the digital output also optimizes the operation of the low bit-rate codec. By not wasting bits encoding the 15-20 kHz frequency range that few radio listeners can hear, the codec instead provides higher quality encoding of the crucial 20-15,000 Hz band.

8400HD also adds a second analog output to the 8400 system. This can be configured to emit the 8400's low-latency monitor signal, or to emit the signal that is processed for the HD digital channel.

User-Friendly Interface

- A large (quarter-VGA) **color liquid crystal display** (LCD) makes setup, adjustment and programming of OPTIMOD-FM 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 (8400 only; 8400PD has no display).

- 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 (8400 only).

Absolute Control of Peak Modulation

- The 8400 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.

- The 8400 provides **pre-emphasis limiting** for the internationally used pre-emphasis curves of 50μs and 75μs. Its pre-emphasis control is seldom audibly apparent, producing a clean, open sound with subjective brightness matching the original program.

- The 8400 achieves extremely tight **peak control** at all its outputs—analog left/right, AES/EBU left/right, 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 8400 eliminates the overshoot problems that waste valuable modulation in traditional external encoders.

- The 8400 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 8400's audio outputs and stereo encoder.

Flexible Configuration

- The OPTIMOD-FM Audio Processor is supplied with analog and **AES/EBU digital** inputs and outputs. Both digital input and digital output are equipped with sample-rate converters and can operate at 32 kHz, 44.1 kHz, and 48 kHz sample rates. The pre-emphasis status and output levels are separately adjustable for the analog and digital outputs.

- OPTIMOD-FM has an **internal, DSP-based stereo encoder** (with a proprietary composite processor) to generate the pilot tone stereo baseband signal and control its peak level.

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

- OPTIMOD-FM 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.

- OPTIMOD-FM has a **subcarrier input** that is mixed with the output of OPTIMOD-FM's stereo encoder before application to the composite output connectors.

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

- OPTIMOD-FM precisely **controls the audio bandwidth** to 15 kHz. This prevents overshoots in uncompressed digital links operating at a 32 kHz-sample rate and prevents interference to the pilot tone and RDS (or RBDS) subcarrier.

- OPTIMOD-FM Audio Processor is designed to **meet all applicable international safety 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 both are active simultaneously to permit mute-free switching between them. OPTIMOD-FM realizes its processing structures as a series of high-speed mathematical computations made by Digital Signal Processing (DSP) chips.

- OPTIMOD-FM features two processing structures: **Five-Band** (or Multiband) for a consistent, "processed" sound, free from undesirable side effects, and **Two-Band** 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.

- OPTIMOD-FM **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.

- OPTIMOD-FM **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.

- OPTIMOD-FM's Two-Band processing structure is **phase-linear** to maximize audible transparency.

Controllable

- OPTIMOD-FM **can be remote-controlled** by 5-12V pulses applied to eight programmable, optically isolated ports.

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

- OPTIMOD-FM 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. It also permits simple ASCII strings to trigger preset recall, facilitating interface to automation systems that can emit such strings through an RS232 serial port.

- OPTIMOD-FM can be connected to a TCP/IP **network** through an optional rear-mounted Ethernet PC card.

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

- OPTIMOD-FM's **software can be upgraded** remotely through its Serial 1 port (connected to an external modem) or PC-Card Port (via an optional Ethernet card), or locally (by connecting a Windows® computer to its Serial 1 port through the supplied null modem cable) and running Orban-supplied downloadable upgrade software.

- OPTIMOD-FM's **user presets** can be **saved, recalled, and archived** through its PC Remote software or to a PC-Card memory card plugged into its front PC Card port.

- OPTIMOD-FM Audio Processor contains a built-in **line-up tone generator**, facilitating quick and accurate level setting in any system.

- The 8400 Audio Processor contains a versatile real-time clock, which allows automation of various events (including recalling presets) at pre-programmed times.

Presets in OPTIMOD-FM

There are two distinct kinds of presets in OPTIMOD-FM: 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 more or less processing.

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, which you are free to name as you wish. The Factory Preset remains unchanged.

User Presets

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

You may store up to 64 User Presets. You may enter in any name you wish, up to 20 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, 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 is designed to simultaneously accommodate:

- Digital AES/EBU left/right inputs and outputs.
- Digital AES/EBU sync reference input.

- Analog left/right inputs and outputs.
- Composite stereo outputs.
- Subcarrier (SCA and RDS/RBDS) input.

The 8400HD option adds:

- A second AES/EBU digital output, which carries the HD-processed signal.
- A second stereo pair of analog outputs, switchable between the low-latency "monitor" signal and the HD-processed signal.
- A second AES/EBU sync input. The sample rate of the HD-processed signal can be genlocked to this input, independent of the sample rate of the main (FM-processed) AES/EBU output.

Digital AES/EBU Left/Right Input/Output

The digital input and output conform to the professional AES/EBU standard. They both have sample rate converters to allow operation at 32, 44.1, and 48 kHz sample frequency.

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

OPTIMOD-FM simultaneously accommodates digital and analog inputs and outputs. You select whether OPTIMOD-FM uses the digital or analog input on the Input/Output screen or by remote interface. Both analog and digital outputs are active continuously.

Level control of the AES/EBU input is via software control through the Input/Output screens.

In addition, an AES/EBU sync input can accommodate house sync. It will lock the 8400's output to this sync even if the digital input is asynchronous to house sync.

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 3 on page 2-30 and step 4 on page 2-32.)

Stereo Analog 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 footnote on page 1-12 and refer to Fig. 2-3 on page 2-10). Independent level control of each output is via software in the Input/Output: Composite screen. A ground lift switch is available on the rear panel. This is useful to prevent ground loops between the 8400 and the transmitter.

The second composite output can be reconfigured in software to provide the stereo pilot tone only, which can provide a pilot reference for an RDS subcarrier generator.

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. Their level will not be affected by the composite level control for that output.

> The mixing occurs after D/A conversion and is analog. Subcarriers are not digitized by the 8400.

The gain from the subcarrier input to the composite output is fixed at –20 dB. Therefore, the gain is scaled so that 1.5V peak at the subcarrier input produces 10% subcarrier injection with reference to 3.0Vp-p=100% modulation of the FM carrier. This –20 dB gain minimizes the effect of any noise picked up on the subcarrier input cable while still accommodating most subcarrier generators.

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 that total baseband peak modulation be maintained within specified limits. Thus, the level of the stereo main and subchannel must be reduced when a subcarrier is turned on. The 8400's remote control feature allows you to reduce the stereo main and sub-channel level by connecting an on/off signal from your subcarrier generator (See page 2-9). You define the amount of reduction in % on the Input/Output screen (See page 2-32). See page 2-51 for information on programming the remote control.

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 8400:

- Recall any Factory Preset, User Preset, Test Mode state (Bypass or Tone), or exit from a Test Mode to the previous processing preset.

- Switch the stereo encoder to stereo, mono from left, mono from right, or mono from sum audio input.

- Switch the 8400 to use either the analog input or the digital input.

- Reduce the stereo main and subchannel modulation to compensate for transmitter overshoot and subcarrier inputs (SCAs).

  The remote control of overshoot compensation and SCA modulation (see page 2-51) is not latching. You must supply a continuous current to the programmed remote input to hold the gain at its compensated level. Use the status outputs of your transmitter and/or SCA generators to provide the switching signal so the compensation will automatically follow the transmitter and/or subcarrier generator on the air.

- Reset the 8400's internal clock to the nearest hour or to midnight.

  The functions of the eight inputs can be re-configured by the user via System Setup Network/Remote. For example, if you are not using the stereo encoder, the three inputs ordinarily dedicated to controlling the state of the stereo encoder can instead be re-configured to call three additional presets. See page 2-51 for information on programming the remote control interface.

Computer Interface

On the rear panel of the 8400 is a serial port and a PC-Card port for interfacing to IBM-compatible PCs. These computer interfaces support remote control and metering, and downloading software upgrades.

Each 8400 package ships with 8400 PC Remote software, a program for any IBM-compatible PC with VGA graphics or higher (running Microsoft Windows 98 or higher). 8400 PC permits you to adjust any 8400 preset by remote control, or to do most anything else that you can do from the 8400's front panel controls. The program displays all of the 8400's LCD meters on the computer screen to aid remote adjustment.

RS-232 Serial Port (Serial 1)

8400 PC Remote can communicate via modem or direct connection between the computer and the 8400 through their RS-232 serial ports.

RS-232 Serial Port (Serial 2)

A computer (running a simple ASCII terminal program like Hyperterminal®) can communicate with the 8400 through direct cable connection between their RS-232 serial ports. This connection can administer communications and security, and can recall presets.

PC-Card Port (for Ethernet Cards)

An optional Ethernet PC card can be plugged into the PC-Card Port on the rear panel of the 8400 to connect to any Ethernet-based network that supports the TCP/IP protocol.

The Ethernet card is supplied standard on 8400/PD units.

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 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 therefore 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 pre-emphasis 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 ±10° 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 pre-emphasis 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 pre-emphasis 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 8400, 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 8400 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 waveform — a short length of coaxial cable. If this is impossible, the next best arrangement is to feed the 8400's AES/EBU digital output through an all-digital, uncompressed path to the transmitter's exciter.

Use the 8400'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 8400'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 AES/EBU digital link is available to the transmitter, this is also an excellent means of transmission, although it will not pass the effects of the 8400's composite processor (if you are using it). However, if the digital link employs lossy compression, it will disturb peak levels.

If only an audio link is available, use the 8400's left and right audio outputs and feed the audio, without pre-emphasis, directly into the link. If possible, request that any transmitter protection limiters be adjusted for minimum possible action — 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 1-10).

If the transmitter is accessible:
You can achieve the most accurate control of modulation peaks by locating OPTIMOD-FM at the transmitter site and 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 1-14). However, many composite baseband STLs do not control peaks perfectly because of bounce (see page 1-15), and locating OPTIMOD-FM at the transmit-

ter site (where it can control peaks just prior to the transmitter's RF exciter) is thus likely to maximize loudness.

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 8400's internal stereo encoder to feed the output of the encoder directly to the baseband input of the exciter through less than 100 feet (30 meters) of coaxial cable.

100 feet of coaxial cable (assuming 30pF/foot capacitance) will reduce measured separation at 15 kHz (worst case) to approximately 60 dB. This separation is comfortably above the separation (approximately 20 dB) that starts to cause perceptible changes in the stereo image.[1]

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. However, because it accepts audio in left/right form, the 8218 will not let you exploit the 8400'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 Orban's CIT25 Composite Isolation Transformer 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. Its instruction manual contains complete information on its installation and application.

If a separate stereo encoder must be used, feed the encoder directly from the 8400's left and right analog outputs. If possible, bypass the pre-emphasis network and the input low-pass filters in the encoder so that they cannot introduce spurious peaks. Because of their special design, OPTIMOD-FM's pre-emphasis network and low-pass filters perform the same functions while retaining tight peak control.

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[1] 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, broa dBand 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.]

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 8400's input, (2) they can pass the 8400's peak-controlled analog or digital left and right audio outputs, or (3) they can pass the 8400'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. Such processing 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 8400 to the transmitter. For example, the MPEG Layer 2 algorithm can increase peak levels up to 4 dB at 160kB/sec by adding large amounts of quantization noise to the signal. While the desired program material may psychoacoustically mask this noise, it is nevertheless large enough to affect peak levels severely. For any lossy compression system the higher the data rate, the less the peak levels will be corrupted by added noise, so use the highest data rate practical in your system.

It is practical (though not ideal) to use lossy data reduction to pass unprocessed audio to the 8400'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 8400 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 8400's multiband processing will not unmask quantization noise. This is also true of any lossy data reduction applied in the studio (such as hard disk digital delivery systems).

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

Some links may use straightforward PCM (pulse-code modulation) without lossy data reduction. If you connect to these through an AES/EBU digital interface, these can be very transparent provided they do not truncate the digital words produced by the devices driving their inputs. Because the 8400's output is tightly band-limited to 15 kHz, it can be passed without additional overshoot by 32, 44.1 or 48 kHz links equally well.

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 AES/EBU input, you must drive its analog input from the 8400'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 pre-emphasis.

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 8400's AES/EBU 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, and 48 kHz.

Composite Baseband Microwave STLs

The composite baseband microwave STL carries the standard pilot-tone stereo baseband, and is therefore fed from 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 8400's stereo encoder to drive the composite STL transmitter.

Dual Microwave STLs

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

Lack of transparency in the path will cause overshoot. Unless carefully designed, dual microwave STLs can introduce non-constant group delay in the audio spectrum, distorting peak levels when used to pass processed audio. Nevertheless, in a system using a microwave STL, the 8400 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 8400 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 8200ST Compressor/Limiter/HF Limiter/Clipper or an Orban Transmission Limiter protects the link from overload.

If the 8400 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 8400.

Some microwave links can be modified such that the deviation from linear phase is less than ±10° 20-15 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 8400 is usually located at the main FM transmitter and is driven by the microwave receivers. The microwave transmitters at the studio are protected from overload by one of Orban's Studio Level Control Systems, such as the 8200ST. These units also perform the gain riding function ordinarily executed by the AGC section of the 8400's processing, and optimize the signal-to-noise ratio obtainable from the dual-microwave link.

If the STL microwave uses pre-emphasis, its input pre-emphasis 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 pre-emphasized output, we strongly recommend that the microwave STL's pre-emphasis be defeated, and pre-emphasis performed in the Studio Level Control System. This frees the system from potential overshoot. (The Orban 8200ST can be easily 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 2 dB, 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. Even the best landlines tend to slightly veil audio quality, due to line equalizer characteristics and phase shifts. 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) Studio Chassis with the 8400

If you have an OPTIMOD-FM 8100A1 (or 8100A or 8100A/1) installation that uses an Orban 8100AST (or 8100A/ST) Studio Chassis at the studio to protect an STL (with the main 8100A, 8100A1 or 8100A/1 chassis at the transmitter), you may wish to continue to use the Studio Chassis to protect the STL when you install the 8400 at the transmitter.

If you are keeping your analog OPTIMOD-FM as a standby processor, you will probably want to use the Studio Chassis to drive both the 8400 and the 8100A1 (also called 8100A/1) transmitter chassis in parallel. This is usually practical, although 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 8400, the Studio Chassis 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 Studio Chassis 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 8400's remote control feature offers the means to reduce the peak level of the 8400's audio output as necessary. This way, you can still use the 8400's line-up tone to adjust the steady-state deviation to ±75 kHz. Yet, the reduced peak level of the audio emitted from the 8400 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.

Using Lossy Data Reduction in the Studio

Many stations are now using lossy data reduction algorithms like MPEG-1 Layer 2 to increase the storage time of digital playback media. In addition, source material 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, "gurgling," or other interference. Psychoacoustic calculations are used to ensure that the added noise is masked by the desired program material and not heard. Cascading several stages of such processing can raise the added quantization noise above the threshold of masking, such that it is heard. In addition, 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 14 dB. 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 easily read the peak level. Fig. 1-1 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, and so that all devices operate with the same input and output levels. This facilitates patching devices in and out without recalibration.

For line-up, the studio engineer uses a calibration tone at a studio standard level, commonly called line-up level, reference level, or operating level. Metering at the studio is by a VU meter or PPM (Peak Program Meter). As discussed above, the VU or PPM indication under-indicates the true peak level. Most modern studio audio devices have a clipping level of no less than +21 dBu, and often +24 dBu or more. So the studio standardizes on a maximum program indication on the meter that is lower than the clipping level, so those peaks that the meter doesn't 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 +8 dBu.

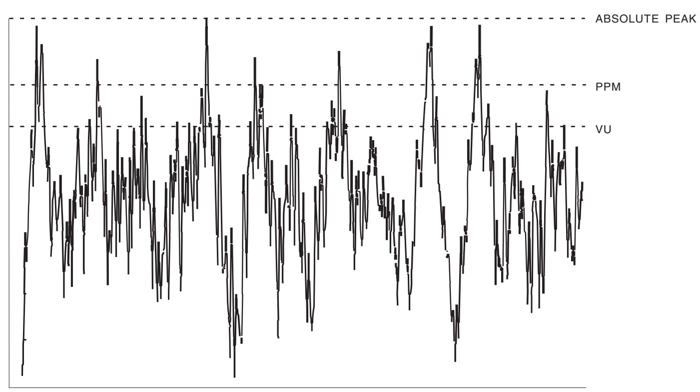


Fig. 1-1: Absolute Peak Level, VU and PPM Reading

For facilities using +4 dBu standard level, instantaneous peaks can reach +18 dBu or higher (particularly if the operator overdrives the console or desk). Older facilities with +8 dBu standard level and equipment that clips at +18 or +21 dBu 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 +4 dBu. Instantaneous peaks will reach +17 dBu or more on voice.

In facilities that use PPMs that indicate level directly in dBu, maximum program and line-up level is often +6 dBu. Instantaneous peaks will reach +11 dBu 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.

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

The meters on the 8400 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 corresponds to the absolute maximum peak level that the 8400 can accept, at the current setting of its Clip Level control. If you are using the AES/EBU digital input, the maximum digital word at the input corresponds to the 0 dB point on the 8400's input meter.

Left/Right Output Level

Left and right output level is shown on a VU-type scale (–10 to +3 dB), where the metering indicates absolute instantaneous peak (much faster than a standard PPM or VU meter). The meter is scaled so that 0 dB is calibrated to the highest left and right peak modulation level, before de-emphasis, that the processing will produce, under any program, processing, or setup condition (except when the processing is switched to Bypass). The meter indication is not affected by the setting of the output level control.

Composite Output Level

The Orban 8400 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 8400'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 it is metered. Therefore, you must mentally add the subcarriers to the meter indication, or refer to an external, calibrated modulation monitor.

Built-in Calibrated Line-up Tones

To facilitate matching the output level of the 8400 to the transmission system that it is driving, the 8400 contains an adjustable test tone oscillator that produces sine waves at 8400'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 in Section 3).

The stereo encoder is calibrated so that 100% left or right modulation will provide 100% modulation of the stereo composite signal, including pilot tone, but excluding any SCA subcarriers.

The pilot tone stereo system has an *interleaving* property, which means that the stereo composite modulation is approximately equal to the *higher* of the left or right channels. Because the pilot tone is phase-synchronous with the stereo subcarrier, the composite modulation will actually increase about 2.7% when the modulation is changed from pure single-channel to L+R modulation while the peak audio level is held constant.

When the 8400's left/right analog output is switched to Flat, a de-emphasis filter is inserted between output of the 8400's audio processing and its line output. Thus, as the frequency of the Test Tone is changed, the level at the 8400's line output will follow the selected de-emphasis curve. In most cases the pre-emphasis filter in the driven equipment will undo the effect of the 8400's internal de-emphasis, so the 8400's output level should be adjusted such that the tone produces 100% modulation of the transmission link as measured after the link's pre-emphasis filter. At 100Hz, switching the de-emphasis out or in will have negligible effect on the level appearing at the 8400'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, with 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.** The delay through the 8400 can be as much as 39 milliseconds (when Soft bass clipping is selected). This delay is likely to be audible as a distinct echo, which most talent finds uncomfortable and distracting. However, the normal delay through the 8400 (from input to FM outputs) is about 20 ms when Hard or Medium bass clipping is selected, as it is in all factory presets other than those with "LL" ("low latency") in their names. A 20 ms delay is comfortable for most talent because they do not hear echoes of their own voices in their headphones. Consequently, customers can ordinarily replace an older processor with the 8400 with no studio wiring changes. Moreover, off-air cueing of remote talent is routine.

A low-delay option is available, which reduces input/FM-output delay to 15 ms. The trade-off for this reduction is slightly lower performance than the 8400's full look-ahead processing offers.

> You can invoke the low-delay mode by setting the BassClipMode control (in the Clippers page of Advanced Control) to LLHard, or by recalling a preset with "LL" as part of its name. ("LL" stands for "low latency.")
>
> 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 15 ms.
>
> 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 8400's analog outputs can be switched to provide a low-delay monitoring feed. This feed has no peak limiting and thus cannot drive a transmitter, but its 5-10ms delay may be more comfortable to talent than the 20 ms delay of the full processing chain because of less bone conduction comb filtering.

> 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 ms and the talent starts to hear slap echo in addition to frequency response colorations.

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 8400 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's always de-emphasized (even if the Analog Output is set with pre-emphasis).

EAS Test

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

1. Run EAS tones and data through the 8400.

Note: Normal 8400 processing may not allow the full modulation level as required by EAS standards. It is therefore necessary to temporarily defeat the 8400's processing during the broadcast of EAS tones and data. Placing the 8400 in its Bypass Test Mode can defeat the processing. The Bypass Gain control allows a fixed gain trim through the 8400. See "Test Modes," on page 3-59 for more information.

2. Place the 8400 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.

3. Place the 8400 in Bypass mode by remote control. Then 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 8400, according to the wiring diagram in Section 2.

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.

H) Place the 8400 in Bypass mode by remote control.

a) Switch the 8400 into Bypass mode by a momentary command from your station's remote control to the input programmed as Bypass.

b) Begin EAS broadcast.

c) When the EAS broadcast is finished, switch the 8400 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 8400 for the duration of the EAS tones and data broadcast.

Warranty, Feedback

The warranty, which can be enjoyed only by the first end-user of record is located on the inside back cover of this manual. Save it for future reference. Details on obtaining factory service are provided in Section 5.

User Feedback Form

We are very interested in your comments about this product. Your suggestions for improvements to either the product or the manual will be carefully reviewed. A postpaid User Feedback Form is provided in the back of this manual for your convenience. If it is missing, please write us at the address printed in the front of the manual, or call or fax our offices at the number listed. We will be happy to hear from you.

Section 2 Installation

Installing the 8400

Allow about 2 hours for installation.

Installation consists of: (1) unpacking and inspecting the 8400, (2) checking the line voltage setting, fuse, and power cord, (3) setting the Ground Lift switch, (4) mounting the 8400 in a rack, (5) connecting inputs, outputs and power, (6) optional connecting of remote control leads and (7) optional connecting of computer interface control leads.

When you have finished installing the 8400, proceed to "Quick Setup," on page 2-23.

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 8400 are:

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

B) Save all packing materials! If you should ever have to ship the 8400 (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) Complete the Registration Card and return it to Orban. (please)

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

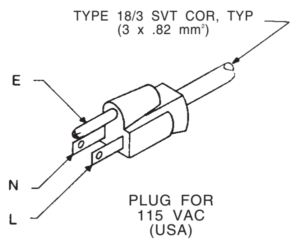
A) DO NOT connect power to the unit yet!

B) Check the Voltage Select switch. This is on the rear panel.

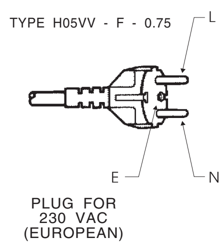
The 8400 is shipped from the factory with the Voltage Select switch set to the 230V position. Check and set the Voltage Select switch to your local voltage requirements. To change the operating voltage, set the Voltage Select to 115V (for 90-130V) or 230V (for 200-250V) as appropriate.

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

The 8400 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-250V Slow-Blow for 115V, or ¼-A-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

Fig. 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), as appropriate to your 8400's Model Number. The green/yellow wire is connected directly to the 8400 chassis.

If you need to change the plug to meet your country's standard and you are qualified to do so, see Fig. 2-1. Otherwise, purchase a new mains cord with the correct line plug attached.

3. Set Ground Lift switch.

The Ground Lift switch is located on the rear panel.

The Ground Lift switch is shipped from the factory in the GROUND position, (to connect the 8400's circuit ground to its chassis ground). If you are using the 8400's composite output to drive an exciter with an unbalanced output, set the switch to LIFT.

This will break most potential ground loops. If you have an installation that does not respond to use of the Ground Lift switch, you can always break a ground loop by using Orban's CIT25 Composite Isolation Transformer. If the CIT25 is in use, the Ground Lift switch will almost always be set to GROUND.

For RFI protection, the 8400's headphone output gets its ground return from *chassis* (not circuit) ground. Therefore, if there is no connection between circuit and chassis ground, headphones will not work. This can occur if the 8400 is removed from the rack and the Ground Lift switch is in the LIFT position. Temporarily set the Ground Lift switch to the GROUND position to cure the problem.

4. Mount the 8400 in a rack.

The 8400 requires three standard rack units (5 inches/12.7 cm).

There should be a good ground connection between the rack and the 8400 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.

5. Connect inputs and outputs.

See the hookup and grounding information on the following pages.

6. Connect remote control interface. (optional)

For a full listing of 8400's extensive remote control provisions, refer to "Remote Control Interface Programming" on page 2-51.

Optically isolated remote control connections are terminated in a type dB-25 male connector located on the rear panel. It is wired according to Fig. 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 ground at pin 17 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. COMMON
2. REMOTE 1+
3. REMOTE 2+
4. REMOTE 3+
5. REMOTE 4+
6. REMOTE 5+
7. REMOTE 6+
8. REMOTE 7+
9. REMOTE 8+
10. TALLY 1
11. TALLY 2
12. N/C
13. POWER COMMON
14. REMOTE -
15. REMOTE -
16. REMOTE 3-
17. REMOTE 4-
18. REMOTE 5-
19. REMOTE 6-
20. REMOTE 7-
21. REMOTE 8-

22 - 24. N/C

25. +12 VDC

REMOTE INTERFACE

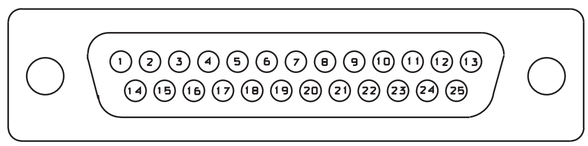


Fig. 2-2: Wiring the 25-pin Remote Interface Connector

7. Connect to a computer

You can connect to a computer running Orban's PC Remote application via the 8400's Serial 1 connector or via an Ethernet network. Ethernet networking requires you to plug an (optional) Ethernet PC card into the 8400's rear PC card slot with 8400 power off. (See **Networking** on page 2-52.)

Because procedures and instructions for connecting to a PC are subject to development and change, we have placed these instructions in a file called `8400_Vxxx_installation.pdf` (where `xxx` represents the version number of the software). You can access this file from the Orban/Optimod 8400 folder in your computer's Start Menu after you have run Orban's PC Remote installer software or version 1.0 or greater of Orban's 8400 software update software. You can use Adobe's .pdf reader application to open and read this file. If you do not have the .pdf reader, it is available for free download from `www.adobe.com`.

See "Installing 8400 PC Remote Control Software" on page 2-56 for more detail.

This file is also available from the /8400/Documentation/Vxxx folder at Orban's ftp site, ftp.orban.com.

You can also connect a computer to the 8400's Serial 2 connector to recall passwords and to administer communications and security, all using simple ASCII strings. See **Administering the 8400 through Serial Port #2** on page 2-46.

8400 Rear Panel

The **Ground Lift Switch** can be set to connect the 8400's circuit ground to its chassis ground (in the GROUND position). In the LIFT position, it breaks that connection. (See **Set Ground Lift switch** on page 2-3.)

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. The fuse must be 1A-250V Slow-Blow for 115V, or 1A-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 8400's Model Number.

An **RS-232 (PC Remote) Computer Interface,** labeled **Serial 1**, is provided to connect the 8400 to IBM PC-compatible computers, directly or via modem, for remote control, metering and software downloads. An additional RS-232 port, label **Serial 2**, is reserved for future development.

A **Remote Interface Connector** is provided to connect the 8400 to your existing transmitter remote control. The 8400 remote control supports user-programmable selection of up to eight optically isolated inputs for any one of the following parameters: recalling any factory- or user presets, tone or bypass modes, selecting stereo encoder modes (stereo, mono-left, mono-right, mono-sum), selecting analog, digital or digital+J.17 input, overshoot compensation, SCA modulation compensation, and clock synchronization. (See **Remote Control Interface Programming** on page 2-51.) The 8400 remote control accepts a dB-25 connector.

The **PC-Card Port (PCMCIA)** accepts an optional Ethernet card for connecting to network cable, or a Modem card to connect to phone lines, used for remote control and metering, or software updates. This card is *not* hot-swappable. **Do not insert or remove the card while the 8400 is powered.**

Digital AES/EBU Input and **Output** are provided to support two-channel AES/EBU-standard digital audio signals through XLR-type connectors. In addition, an **AES/EBU 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.

Two **Composite Baseband Outputs** are provided, each with independent output level control. The second composite output can be reconfigured in software as a Pilot Reference Output useful for RDS (RBDS) subcarrier generators that require an external sync reference. Each output uses a BNC connector.

Two **SCA Inputs** are provided for stations that use additional subcarriers (SCAs). Each input uses a BNC connector.

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

- Nominal input level between –14 dBu and +8 dBu will result in normal operation of the 8400.

  (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 is dependent 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), do not connect the cable shield at the 8400 input—it should be connected at the source end only. In high RF fields (like a transmitter site), also connect the shield to pin 1 of the male XLR-type connector at the 8400 input.

- If the output of the driving unit is unbalanced and does not have separate CHASSIS GROUND and (–) (or LOW) output terminals, connect both the shield and the black wire to the common (–) or ground terminal of the driving unit.

Analog Audio Output

- Electronically balanced and floating outputs simulate a true transformer output. The source impedance is 50Ω. The output is capable of driving loads of 600Ω or higher; the 100% modulation level is adjustable with the Analog Out Level control 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), it should be taken between pin 2 and pin 3 of the XLR-type connector. Connect the LOW pin of the XLR-type connector (#3 or #2, depending on your organization's standards) to 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 8400's output (and at the output of other equipment in the system), connect the cable's shield to the CHASSIS GROUND terminal (pin 1) on the XLR-type connector. 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.

- The 8400HD ("High Definition Digital Radio") option adds a second stereo pair of analog outputs, switchable between the low-latency "monitor" signal and the HD-processed signal.

AES/EBU Digital Input and Output

In a standard 8400, there are two AES/EBU inputs and one AES/EBU output. One input accepts program audio; the other accepts house sync. The program input and output are both equipped with sample rate converters and can operate at 32, 44.1, and 48 kHz.

Per the AES/EBU standard, each digital input or output line carries both the left and right stereo channels.

The digital input clip level is fixed at 0 dB relative to the maximum digital word. The maximum digital input will make the 8400 input meters display 0 dB. The reference level is adjustable using the Digital Reference Level control.

The 8400 is a "multi-rate" system and its internal sample rate is 32 kHz and multiples thereof (up to 512 kHz). The output is strictly band-limited to 16 kHz. Therefore, the output can be passed through a 32 kHz uncompressed link with bit-for-bit transparency. Because sample rate conversion is a phase-linear process that does not add bandwidth, the 8400's output signal will continue to be compatible with 32 kHz links even if it undergoes intermediate sample rate conversions (for example, 32 kHz to 48 kHz to 32 kHz).

The 8400HD option adds:

- A second AES/EBU digital output, which carries the HD-processed signal.
- A second AES/EBU sync input. The sample rate of the HD-processed signal can be genlocked to this input, independent of the sample rate of the main (FM-processed) AES/EBU output.

Composite Output and Subcarrier Input

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 8400's subcarrier inputs.

The output level of each output is independently adjustable in the Input/Output: Composite screen from –12.0 dBu to +12.0 dBu. Output impedance may be configured for 0 or 75Ω by internal jumpers (refer to Maintenance section). Each output can drive up to 75Ω in parallel with 0.047μF before performance deteriorates significantly (see footnote on page 1-12). A Ground Lift switch is available on the rear panel. This is useful to prevent ground loops between the 8400 and the transmitter.

The second composite output can be reconfigured in software to provide the stereo pilot tone only, which can provide a pilot reference for RDS or RBDS subcarrier generators.

Connect the 8400'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 8400 close to the exciter and limit the length of the composite cable to less than 6 feet (1.8m).

We do not recommend that the exciter input be terminated 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 tend to have more accurate amplitude and phase response (and thus, better stereo separation) if the coax is driven by a very low impedance source and is terminated by greater than 1kΩ at the exciter end. This also eases thermal stresses on the output amplifier in the stereo encoder, and can thus extend equipment life.

If the Orban CIT25 Composite Isolation Transformer is used, the exciter *must* present a 1kΩ or greater load to the transformer for proper transformer operation.

Designed to be installed adjacent to each exciter, the CIT25 Composite Isolation Transformer provides ground loop isolation between the 8400 composite output and the exciter's input, and presents the 8400 with a balanced, floating load.

Even when its composite limiter is being heavily used, the 8400 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).

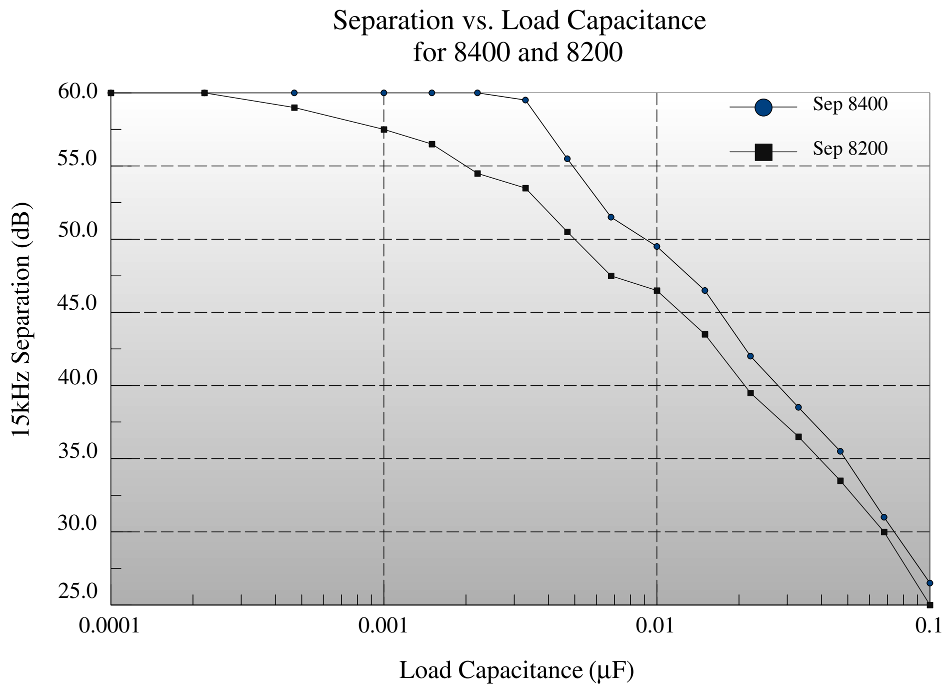


Fig. 2-3: Separation vs. load capacitance for 8400 and Orban stereo encoders using 8200-style line driver (8200, 2200, 8208, and 8218). Test frequency = 15 kHz.

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 20 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 8400's composite output in the analog domain, after D/A conversion of the 8400 stereo encoder's output.

Connect your subcarrier generator(s) to the 8400's subcarrier input(s) with coaxial cable terminated with BNC connectors.

> The subcarrier inputs have 600Ω impedance and are unbalanced. To minimize noise pickup and to be compatible with the output levels available from many subcarrier generators, gain from the subcarrier input to the composite output is fixed at –20 dB. Therefore, the gain is scaled so that 1.5V peak at the subcarrier input produces 10% subcarrier injection with reference to 3.0Vp-p = 100% modulation of the FM carrier. (Because the signals at the subcarrier inputs are not digitized, it was not practical to offer a software-settable gain control for them.)

Grounding

Very often, grounding is approached in a "hit or miss" manner. But 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 8400 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 8400 chassis through the third wire in the power cord. Proper grounding techniques never leave equipment chassis unconnected to power/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 should always be connected by setting the 8400's Ground Lift switch to its GROUND connect position, *except* when the 8400's stereo encoder is driving an **unbalanced exciter input**. (Many older exciters have unbalanced inputs.) This is an unbalanced-to-unbalanced connection, so set the 8400's Ground Lift switch to LIFT to break the ground loop that would otherwise occur.

  Alternately, you can balance and float the exciter input with the Orban CIT25 Composite Isolation Transformer—see page 2-10.

- In high RF fields, the system is usually grounded through the equipment rack in which the 8400 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.

8400 Front Panel

- **Headphone Jack** allows you to monitor the output of the processing through headphones. Headphone impedance should be 75Ω or higher.

  If the HD option is installed, you can switch the headphone feed between the output of the digital-channel processing chain and the signal driving the FM processing's analog output. Otherwise, the headphone jack carries the signal driving the FM processing's analog output, which can be switched between the output as processed for transmission and a low-delay monitor output.

- **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. Your edited preset will automatically be retained without saving even if the 8400 is powered down and will be restored on-air upon power-up. Your edited preset will also appear in the RECALL list of available presets as the name of its parent present prefixed by the letter "M" (for "modified").

However, if you edit another preset, your old edited preset will be lost—the 8400 automatically retains only one "modified" preset. Therefore, it is wise to rename and save any edited preset you wish to keep, using the 8400'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 following meters and indicators are displayed on the 8400's screen:

  - In meters show the peak input level applied to the 8400'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 drastically slowed 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.

    If the Multiband structure is operational, all the meters display gain reduction (G/R) activity. If the Two-Band structure is operational, only the two leftmost meters display G/R activity.

- Out meters display 8400's output in two distinct modes. The meters can be set to display in L/R or L+R/L–R mode.

- 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.

  These meters appear only when the HD option is installed.

- Multiplex Power meter indicates the action of the ITU Multiplex Power controller. It shows how much the MPX Power Controller has reduced the clipper drive, reducing the average power in the processed audio.

  This meter, labeled "P," is displayed on the 8400'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 MPX Power Controller is turned on.

Installation of Studio Level Controller (optional)

If using an Orban 8100AST (or 8100A/ST) Studio Chassis, refer to page 1-16.

[Skip this section if you are not using a studio level controller ahead of the 8400. Continue with "Quick Setup" on page 2-23.]

If you are using Orban 8200ST-Studio Chassis

If the STL uses pre-emphasis, its input pre-emphasis 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 pre-emphasis be defeated (freeing the STL from such potential overshoot), and that the 8200ST be used to provide the necessary pre-emphasis.

If the STL transmitter's pre-emphasis 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.

1. Configure the 8200ST's internal jumpers.

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

B) Refer to Fig. 2-4.

a) Place jumper JA in the "CLIPPER ON" position.

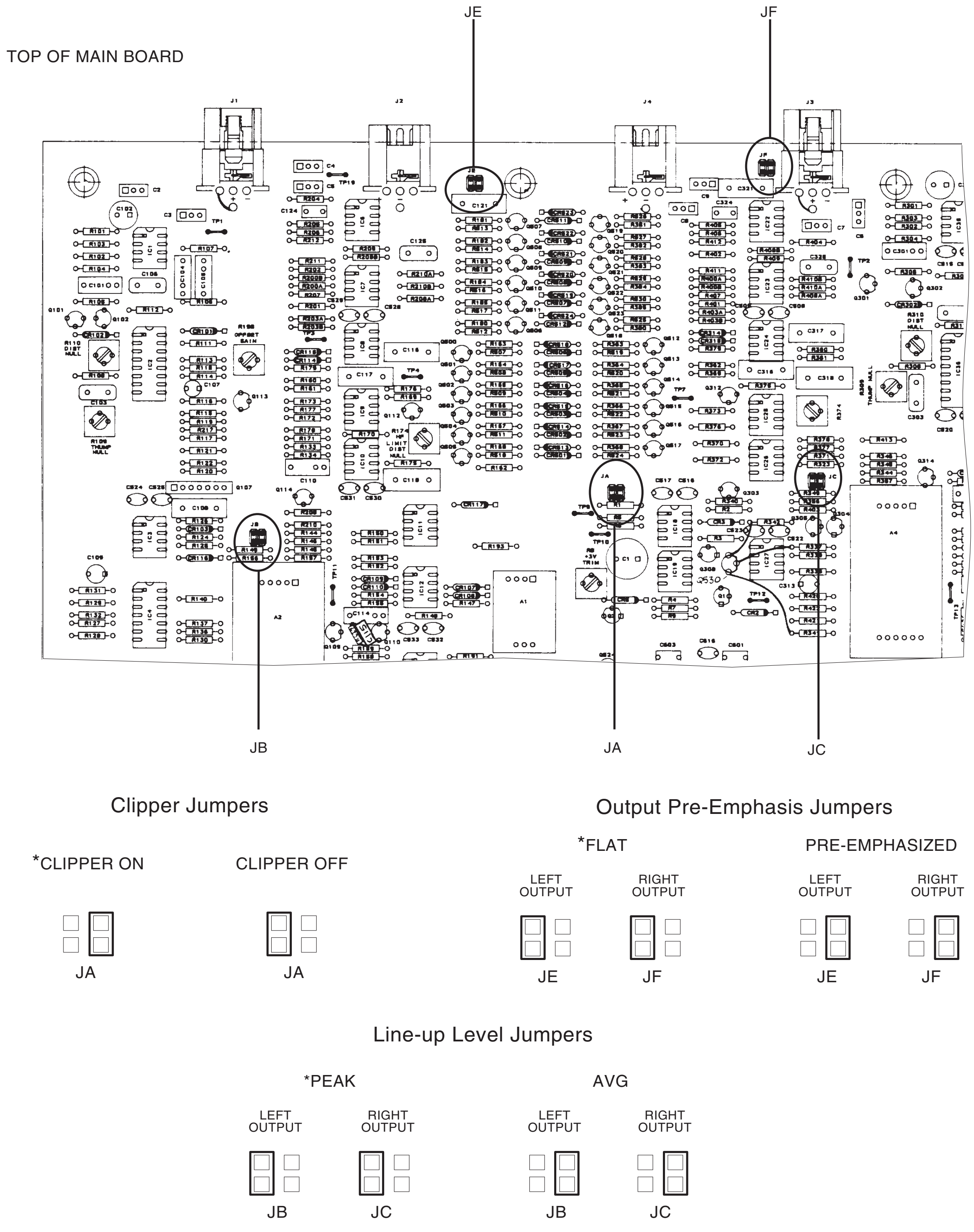


Fig. 2-4: 8200ST Jumper Settings (*Factory Configuration)

b) If you have defeated the STL transmitter's pre-emphasis, place jumpers JE and JF in the "PRE-EMPHASIZED" position.

c) If you cannot defeat the STL transmitter's pre-emphasis, place jumpers JE and JF in the "FLAT" position.

C) Replace the top cover, 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 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 Fig. 2-4.) 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 *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 over-modulation 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 pre-emphasis of the transmission system
L OUT, 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 you are using Orban 464A Co-Operator

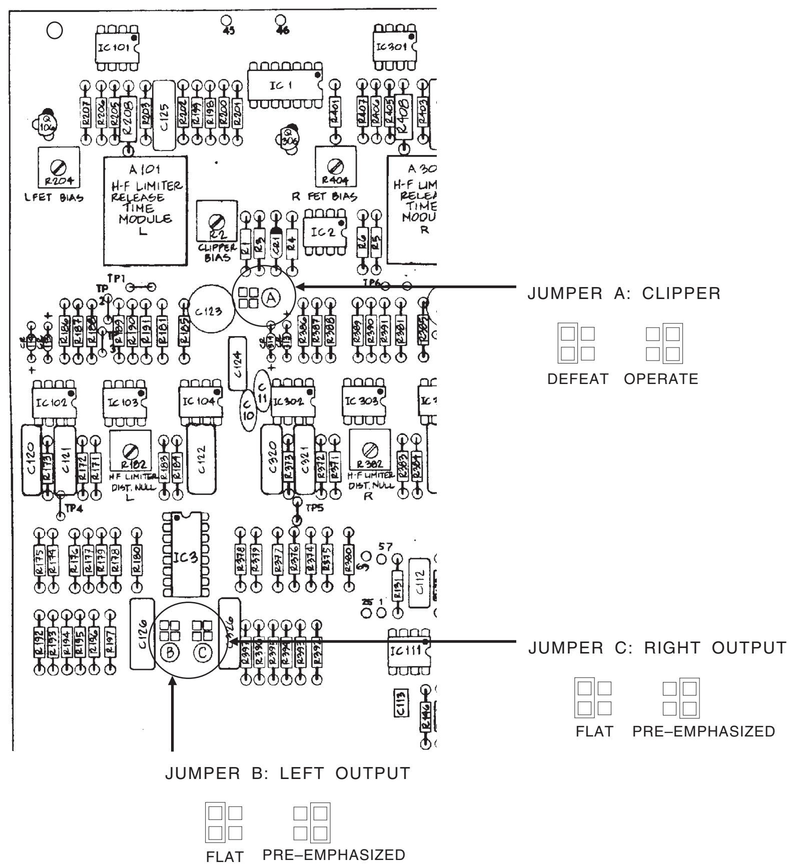


Fig. 2-5: 464A Jumper Settings

If the STL uses pre-emphasis, its input pre-emphasis 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 pre-emphasis be defeated (freeing the STL from such potential overshoot), and that the 464A be used to provide the necessary pre-emphasis.

If the STL transmitter's pre-emphasis cannot be defeated, configure the 464A for flat output. In this case average modulation levels of the STL may have to be reduced to accommodate the overshoots.

1. Configure the 464A's internal jumpers.

A) Remove all screws holding the 464A's cover in place, and then lift it off.

B) Refer to Fig. 2-5.

a) Place jumper A in the "OPERATE" position.

b) If you have defeated the STL transmitter's pre-emphasis, place jumpers B and C in the "PRE-EMPHASIZED" position.

c) If you cannot defeat the STL transmitter's pre-emphasis, place jumpers B and C in the "FLAT" position.

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

2. Install the 464A in the rack. Connect the 464A's audio input and output.

Refer to the 464A Operating Manual if you require information about installation, audio input and output connections to the 464A.

3. Calibrate the 464A's output level and Peak Output Level meters.

There is no quick way to calibrate the 464A's output level and PEAK OUTPUT LEVEL meters using a tone. If your STL has input meters that give an accurate indication of program peaks, calibration may be achieved quickly and precisely using program material following the procedures detailed in this step.

If you wish to use tone to calibrate the 464A's output level and PEAK OUTPUT LEVEL meters, follow the procedure in the 464A Operating Manual, page 2-10, steps 10 and 11. Repeat for the right channel. Note that this procedure instructs you to calibrate the 464A's meters to indicate +3 dB for 100% modulation of the STL. However, you may wish to calibrate them to indicate 0 dB for 100% modulation of the STL. Just be consistent in steps 10 and 11.

To calibrate the 464A's output level and PEAK OUTPUT LEVEL meters using program material:

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

METER CAL	0
HF LIMIT PRE-EMPHASIS	set to pre-emphasis of your STL; if no pre-emphasis, 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

B) Play program material from your studio.

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

D) Adjust the *OUTPUT ATTEN* controls to drive the STL to 100% modulation on program peaks, as shown on its modulation indicator.

4. Set 464A's controls for normal operation with program material.

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

METER CAL	Do not change
HF LIMIT PRE-EMPHASIS	Do not change
OUTPUT ATTEN	Do not change
GATE THRESH	5
RELEASE TIME	5
REL SHAPE	SOFT
LEVEL	ON
COMPR	OFF
HF LIMIT	OPERATE
SYSTEM	OPERATE
POWER	ON
MODE	DUAL

B) Feed the 464A either with tone at your system reference/line-up level, or with typical program material at normal levels.

C) Adjust the *L* and *R INPUT ATTEN* controls for the desired amount of gain reduction—we recommend 5-10 dB.

D) Switch the *MODE* switch to STEREO.

If you are using an Orban 4000 Transmission Limiter

If the STL uses pre-emphasis, its input pre-emphasis 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 pre-emphasis be defeated (freeing the STL from such potential overshoot), and that the 4000 be used to provide the necessary pre-emphasis.

If the STL transmitter's pre-emphasis cannot be defeated, configure the 4000 for flat output. In this case average modulation levels of the STL may have to be reduced to accommodate the overshoots.

1. Configure the 4000's internal jumpers

A) Remove the top and bottom covers to access the main circuit boards.

Note that the 4000 is a two-channel unit and has two boards with identical jumpers for resetting.

B) Refer to Fig. 2-6 for jumper locations.

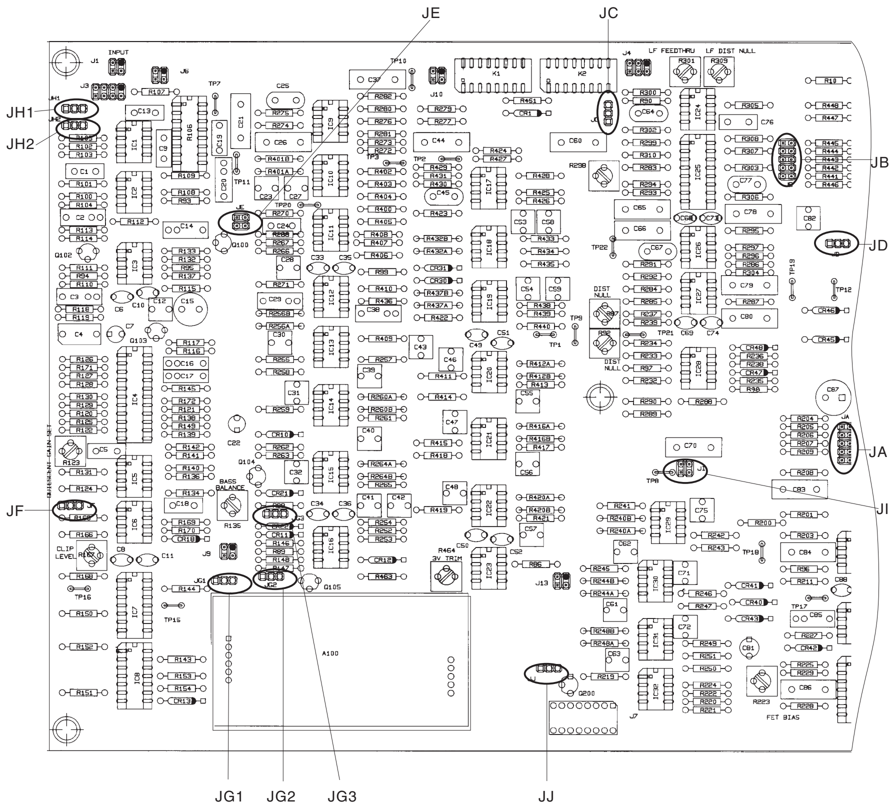


Fig. 2-6: 4000 Jumper Locations

C) Activate the high frequency limiter.

Place jumpers JI and JJ in the "HF LIMITER ACTIVE" position.

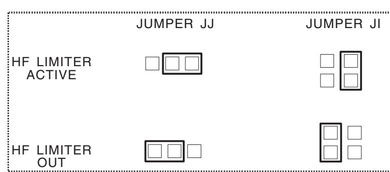


Fig. 2-7: 4000 HF Limiter Jumpers

D) Set pre-emphasis of the high frequency limiter.

Place jumpers JF, JE, JA and JB in the position for the pre-emphasis of your STL (25µs, 50µs, 75µs, 150µs, or J.17.)

E) Set the output for pre-emphasized or flat response, as appropriate.

- If you have defeated the STL transmitter's pre-emphasis, place jumper JD in the "PRE-EMPHASIZED" position.
- If you cannot defeat the STL transmitter's pre-emphasis, place jumper JD in the "FLAT" position.

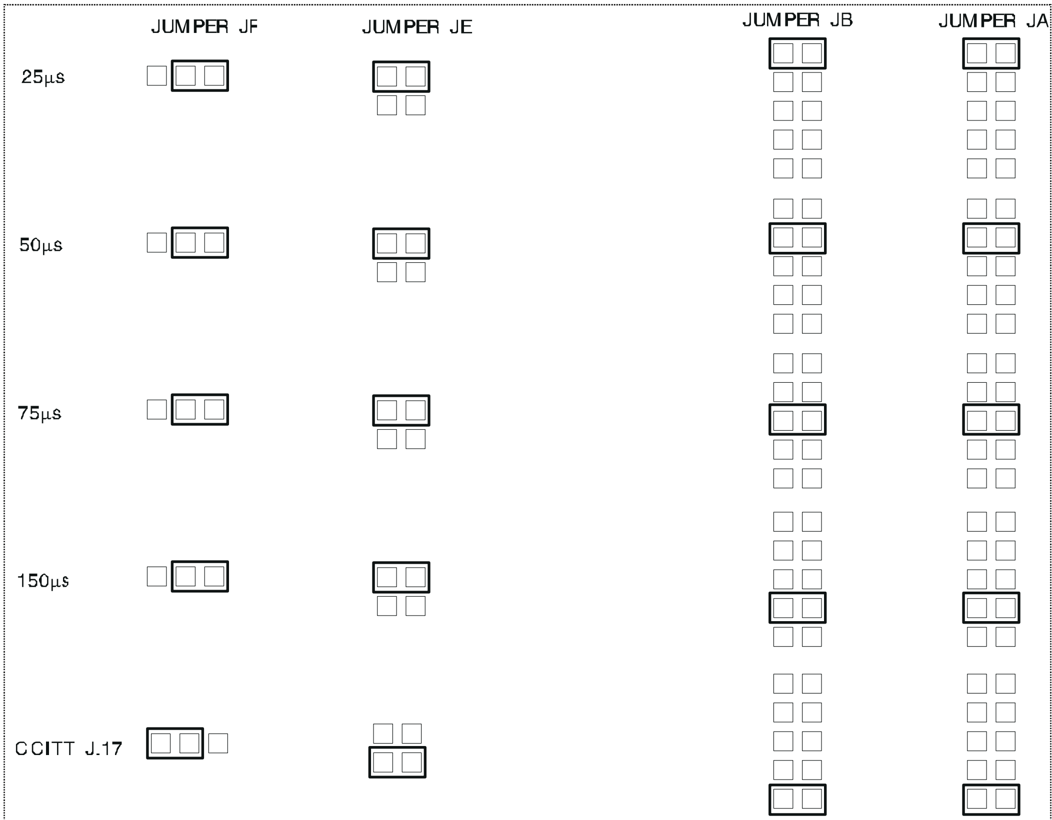


Fig. 2-8: 4000 Pre-Emphasis Jumpers

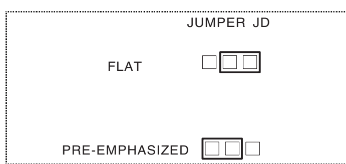


Fig. 2-9: 4000 Pre-Emphasis Jumpers

F) Set the two channels for stereo coupling.

Place jumpers JG1, JG2, and JG3 in the "COUPLED" position.

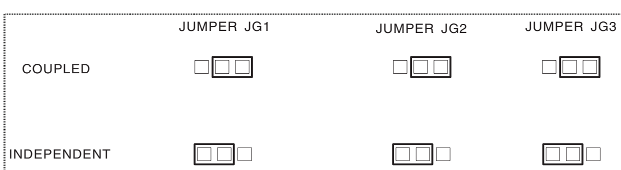


Fig. 2-10: 4000 Stereo Coupling Jumpers

G) Replace the top and bottom covers.

2. Install the 4000 in the rack. Connect the 4000's audio input and output.

Refer to the 4000 Operating Manual if you require information about installation, audio input and output connections to the 4000.

3. Calibrate the 4000's Output level to the STL.

A) Press both *TONE* buttons on the 4000's front panel.

B) Adjust the 4000's Channel A and B *OUT* (Output) levels for 100% peak modulation of the STL.

4. Calibrate the 4000's Input level for normal operation using tone.

[Skip this step if you wish to calibrate the 4000's Input level using program material. (Refer to step 5 on page 2-23.)]

Some facilities have specific standards for transmission line-up. For example, a transmission standard may state that +4 dBu at 400Hz produces 50% modulation of a microwave link. Or PPM6 might allow 8 dB of headroom so it would modulate the link to 40%.

Determine the input level to the studio-to-transmitter link that produces 100% modulation of the link.

In step 4-(D), you calibrate the gain of the 4000 below the threshold of limiting.

For facilities using VU meters, we suggest:

100% peak level (dBu) – 0VU level (dBu) – 14 dB = gain of the 4000

For example, with an STL where 100% modulation = +18 dBu, and with a studio where 0VU = +4 dBu:

+18 – (+4) – 14 = set gain of the 4000 to 0 dB

For facilities using PPM meters, we suggest:

100% peak level (dBu) – PPM reference level (dBu) – 9 dB = gain of the 4000

For example, with an STL where 100% modulation = +15 dBu, and with a studio where PPM reference level = +6 dBu:

+15 – (+6) – 9 = set gain of the 4000 to 0 dB

A) Turn the *IN* (Input) control fully counterclockwise.

B) Press the *OPERATE* button, then the *TEST* button.

C) Apply a line-up tone to the 4000 input, at your organization's standard reference line-up level.

D) Calibrate the 4000 for the pre-determined gain or loss.

Measure the output level of the 4000 with an AC level meter. Adjust the 4000's *IN* control to achieve the desired gain or loss.

E) Press the *OPERATE* button. Observe the LIMITING meter.

If no gain reduction is indicated, the standard line-up level is below threshold.

If gain reduction is indicated, the standard line-up level is above threshold (less than 7 dB below 100% modulation). System calibration will require that the *TEST* button be pressed, either on the front panel or by remote control, when system line-up calibration is performed. You may consider calibrating the 4000 for less than unity gain by reducing the input control setting.

5. Calibrate the 4000's Input level for normal operation using program material.

[Skip this step if you wish to calibrate the 4000's Input level using tone. (Refer to step 4 on page 2-22.)]

A) Turn the *IN* (Input) control fully counterclockwise.

B) Press the *OPERATE* button, then the *TEST* button.

C) Play program material from your studio at normal levels.

D) Adjust the *IN* level controls so that the 4000 goes into gain reduction only on the highest program peaks.

Quick Setup

The 8400's Quick Setup feature provides a guided, systematic procedure for setting up the 8400. It should be fully 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. In most cases, you will not need this extra information.

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 Setup: Quick Setup 1).

3. Select the language in which you want to work.

Like Henry Ford's Model T (available in any color as long it was black), version 3.0 offers any language as long as it happens to be English. We hope to add more languages in future software revisions.

4. Set Time & Date.

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

B) Set the Clock Control to Internal or Line Sync.

Line sync uses the AC line as a reference. We recommend using Internal sync for higher accuracy. (Internal sync uses the 8400's crystal reference, which is backed up by a battery so it always runs even when power is turned off.)

C) Set the Time Zone Offset to "0 hrs."

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

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

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

F) Choose the desired date format.

G) Set today's date.

H) If you want the clock to reset itself automatically to conform to Daylight Savings Time (Summer Time), use the Begins and Ends fields to specify when Daylight Savings 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.

5. Set Pre-Emphasis in Regional Settings screen.

A) *Locate* to the Regional Settings screen (System Setup: Quick Setup 3).

B) Select the pre-emphasis (either 75μS or 50μS) used in your country.

6. Set studio chassis mode.

A) *Locate* to Studio Configuration screen (System Setup: Quick Setup 4).

B) Set studio chassis mode.

Set the field to Yes if you have a studio chassis (such as an Orban 8200ST OPTIMOD-Studio, Orban 464A Co-Operator, or similar AGC) installed at your studio feeding the studio-to-transmitter link. This setting appropriately defeats the 8400's AGC for all presets.

If you do not have a studio chassis installed, set the field to No; this setting enables the 8400 AGC status to be determined by the selected preset.

Most of the processing structures in the 8400 control level with a preliminary AGC (Automatic Gain Control). If you are using a suitable Automatic Gain

Control at the studio (such as an Orban 8200ST OPTIMOD-Studio or 464A Co-Operator), the AGC in the 8400 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.

If you are using an Orban 4000 Transmission Limiter, set field to No (so that the AGC function in the 8400 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 8400.

7. Set Input Levels.

A) Adjust Input Reference Level.

a) Feed normal Program material to the 8400.

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 8400 to the normal operating level to be expected at the 8400, so the 8400'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 8400, optimum adjustment is achieved when the desired 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.

c) Set the input to Analog and adjust Analog Reference Level.

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

d) Adjust the Analog Reference Level so that the meter reads an average of 10 dB gain reduction.

[–9 dBu to +13 dBu (VU), or –1 to +21 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 8400, set the Analog Reference Level to this level, but please verify it with the steps shown directly below.

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.

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 AES/EBU digital input.

e) Set the input to Digital and adjust Digital Reference Level.

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

f) Adjust the Digital Input Reference so that the meter reads an average of 10 dB gain reduction.

[–30 to –10 dBFS (VU), or –22 to –2 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 8400, set the Digital Reference Level to this level, but do verify it with the steps shown directly below.

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.

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.

B) Select primary Input Source

a) Switch the input to the source (analog or digital) that you will use for normal programming.

8. 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 pre-emphasis) 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 pre-emphasis. However, if you cannot defeat the pre-emphasis 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 8400 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 Pre/Flat control to Pre-E (for pre-emphasis) or Flat.

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

(See the notes immediately above.)

D) Set the Digital Output Sample Rate to 32, 44.1, or 48 kHz.

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

The 8400's fundamental sample rate is always 32 kHz, but the internal sample rate converter sets the rate at the 8400's digital output. This adjustment sets the output sample rate to ensure compatibility with equipment requiring a fixed sample rate.

E) Choose the function of the Composite 2 Output.

This output can provide a second composite signal with an independently adjustable level. This usually drives a back-up FM exciter. The second output can also be set to provide a 19 kHz Pilot Reference output for RDS/RBDS generators.

9. 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).

If you are using program material, make sure that the program material is loud enough to produce peaks of frequent recurrence that hit the 8400's peak limiting system, thereby defining the maximum peak level that the 8400 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 multiplex power 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 8400'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 8400 has the ability to reduce audio modulation to compensate for subcarriers. 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 3 (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 8400 GPI input. *If you have the same subcarrier injection at all times*, a more convenient alternative is to set the desired modulation level by using the Composite Level control(s). Then turn up the pilot injection control until the injection equals 9% modulation.

10. 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*, and some of the most competitive presets (the "Loud" and "Impact" families) are purposely 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 later with the 8400's one-knob Less/More feature. Refer to Section 3.

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.

You can easily modify a preset with the 8400's one-knob Less/More feature. 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.

11. Complete Station ID.

The Station ID is an optional setting that you can provide to associate the 8400 with the station providing the program material (e.g., "WXXX"). 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 press *Enter* on the Shift key.

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

12. 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. Or, 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).

13. If you are required to meet the "multiplex power" limitations of ITU-R 412-7 in your country, activate the 8400'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 8400 (including the transmitter), 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 8400'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 8400'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 412-7, the Multiplex Power Threshold should be set to OFF.

The following material provides detailed instructions on how to set up the 8400. 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. However, for most users, this material is only for reference, because Quick Setup has enabled them to set up the 8400 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. Temporarily set the studio chassis mode to "No."

A) Navigate to the Input/Output: Utilities screen and set Studio Chassis to No.

If you are using a studio chassis like the Orban 8200ST, you will restore this setting to Yes after the setup procedure is complete.

2. Adjust Input selector.

A) Navigate back to the Input/Output: Input screen.

B) Set the Input to Analog.

The 8400 will automatically switch to analog input if signal lock is unavailable at the AES/EBU input.

3. Adjust Clip Level control.

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

This step matches the level at which the 8400's A-D (Analog-to-Digital) converter clips to the absolute maximum peak level that your installation supplies to the 8400's analog input.

This setup maximizes the 8400's signal-to-noise ratio. If the clip level is set too low, the 8400'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 misadjustment of the Clip Level control, such that the A/D converter is clipping and distorting. This will *always* be obviously indicated by the Input meters' going into the red part of their scale.

If you are adjusting the 8400 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 2-31.

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 2-31, or
- Calibrate with a Studio Level Control System that has a built-in 100% Calibration Tone, such as the Orban 8200ST-Studio Chassis or the Orban 4000 Transmission Limiter, step (C), page 2-31, or
- Calibrate with an Orban 464A Co-Operator, step (D), page 2-32,

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 8400 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 8400. 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 produce the highest peak level output that your system can produce.

b) Adjust the 8400'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 8400. 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 8200ST-Studio Chassis or the Orban 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 8400'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 8400 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 PRE-EMPHASIS	set to pre-emphasis of your STL; if no pre-emphasis, 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 8400'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.

4. Adjust Reference Level.

[–4 dBu to +13 dBu (VU), or +4 to +21 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 8400's gain reduction range to the level to which your studio operators peak their program material on the studio meters. This ensures that the 8400'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 8400 will receive, set the Reference Level to this level, but do verify it with the steps shown directly below.

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

B) Highlight the Rock-Medium preset.

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

D) Calibrate using Tone.

[Skip this step if you are using Program material to calibrate the 8400 to your standard studio level. Skip to step (E).]

a) Verify Studio Chassis is set to no.

Refer to step 1 on page 2-30 above.

b) Feed a tone at your reference level to the 8400

If you are not using a studio level controller, 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 8400's AGC meters indicate 10 dB gain reduction.

d) When finished, reset Studio Chassis to Yes, if required (e.g., if that was its setting prior to setting Reference Level).

e) Skip to step 5.

E) Calibrate using Program.

[Skip this step if you are using Tone to calibrate the 8400 to your standard studio level—see step (D) above.]

a) Verify Studio Chassis is set to no.

Refer to step 1 on page 2-30 above.

b) Feed normal Program material to the 8400

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 8400'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), re-adjust the Reference Level to a lower level.

If the AGC gain reduction meter averages more gain reduction (lower on the meter), re-adjust the Reference Level to a higher level.

d) When finished, reset Studio Chassis to yes, if required (e.g., if that was its setting prior to setting Reference Level level).

5. Adjust 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

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 8400 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 8400 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).

A) Adjust Right Balance to achieve correct left/right channel balance.

6. 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 4 and 5 above using the Reference Level and Right Balance controls for the Digital section.

7. Set Output and configuration level.

A) Navigate to the Input/Output: Output screen.

B) 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.

C) Set the analog output pre-emphasis control to Pre-E (for pre-emphasis) 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 the stereo encoder being driven by the analog output does not have to restore the pre-emphasis. However, if you cannot defeat the pre-emphasis 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 8400 through a digital link that uses a lossy codec (like MPEG, APT-X, or Dolby), set the output Flat. Lossy codecs cannot handle pre-emphasized signals.

D) Set the Digital Output Pre/Flat control to Pre-E (for pre-emphasis) or Flat.

[Skip steps (D) through (I) if you will not be using the digital output.]

(See the notes immediately above.)

E) Set the digital output Samp Rate (sample rate).

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

The 8400's fundamental sample rate is always 32 kHz, but the internal sample rate converter sets the rate at the 8400's digital output. This adjustment sets the 8400 output sample rate to ensure compatibility with equipment requiring a fixed sample rate.

F) Adjust the SR SYNC control.

[Internal / SYNC IN / INPUT]

You can lock the sample rate of the 8400's AES/EBU output to the sample rate of a reference AES/EBU signal that is applied to either the AES/EBU input or the SYNC input.

The selections are Internal, SYNC IN, and INPUT. INPUT sets the AES/EBU output sample rate and synchronization to the same sample rate present at the AES/EBU input. Likewise, SYNC IN uses the sync input's sample rate and synchronization as the source. Internal sets the AES/EBU output synchronized to the 8400's internal clock and uses the Samp Rate setting to determine its output sample rate.

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 output sample-rate selector. 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 AES/EBU signal is provided to the 8400 Input or Sync In, set SR Sync to Internal and select the desired output sample rate.

G) Set the desired output Wrd Length (word length).

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

The largest valid word length in the 8400 is 24 bits. The 8400 can also truncate its output word length to 20, 18, 16 or 14 bits. The 8400 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.)

H) Adjust Dither to In or Out, as desired.

[In] or [Out]

When set to In, the 8400 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 maximum 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.

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 8400.

I) Set 8400 User Bits to receive or block user bits.

[pass] or [block]

If you want to transparently pass incoming user bits through to your 8400 digital output, set User Bits to pass. Otherwise, set this parameter to block, in which case the 8400 will transmit user bits as all zeros.

J) 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 8400's peak limiting system, thereby defining the maximum peak level that the 8400 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 8400'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 8400'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%."

In software versions below 1.0, the feature to change modulation level to compensate for subcarriers is not yet implemented. If you are using such software, we strongly urge you to upgrade to the latest version. You can download it from www.orban.com. Click "Downloads" and navigate to the 8400 folder.

8. Configure Composite Outputs

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

A) Navigate to the Input/Output: Composite screen.

B) Choose the function of the Composite 2 Output.

This output can provide a second composite signal with independently adjustable level. This usually drives a back-up FM exciter. The second output can also be set to provide a 19 kHz Pilot Reference output for RDS/RBDS generators. If required, set the Output 2 to Pilot Ref.

C) Adjust the composite level to produce 100% modulation of the FM carrier on modulation peaks. Alternately, you can use tone.

If you are using subcarriers, this screen allows you to specify the amount by which the 8400 automatically reduces main and stereo subchannel modulation to accommodate the subcarrier within the modulation limits specified by the governing authority.

See step 9-(C) on page 2-27 for a more detailed discussion.

D) Ordinarily, you will set the Pilot to On and 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 8400 (including the transmitter), you may have to increase the setting of the Pilot Level so that the pilot is still at 9% modulation.

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

If you do not need the 8400's analog output to drive a transmitter, you can configure it to receive the output of a special version of the multiband compressor (without look-ahead). This signal is suited for driving headphones. The input/output delay is approximately 5-10 milliseconds (depending on the setting of AGC Crossover Type). Even though normal 8400 presets have a delay of about 20 ms, which most DJs, announcers, and presenters can learn to use without discomfort (although they may need some time to become accustomed to it), the low-delay output will cause less bone-conduction comb filtering. However, in most cases, the low-delay output will not be necessary to ensure adequate talent comfort.

The normal delay is 20 ms except for "LL" ("low latency") presets, which have 15 ms delay.

To configure the Analog Output for Low-Delay Monitoring:

A) Choose Input/Output from the main menu.

B) Navigate to the Output screen (Input/Output 2).

C) Choose Out Feeds: Monitor.

D) Set the analog output pre-emphasis control to Flat.

CAUTION: The low-delay output has no peak limiting and is therefore not suited for driving a transmitter. If you use the low delay output, you must drive your transmitter with the AES/EBU digital output or with the composite output.

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 8400'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 Feeds parameter located in the Input/Output 2 screen needs to be set to Monitor to make the Monitor Mute feature work.

10. End Analog and Digital I/O setup.

If you are using a studio chassis and you temporarily set the Studio Chassis to No in step 1 on page 2-30, set the Studio Chassis to Yes.

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

11. 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 8400 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.

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 8400's one-knob Less/More feature. 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.

Using Clock-Based Automation

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

A) Navigate to System Setup, Place/Date/Time.

B) Navigate to Screen 1 to set the time and date.

a) Set the Clock Control to Internal or Line Sync.

Line sync uses the AC line as a reference. We recommend using Internal sync for higher accuracy. (Internal sync uses the 8400's crystal reference, which is backed up by a battery so it always runs even when power is turned off.)

b) Set the Time Zone Offset to "0 hrs."

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

d) Set hours, miniutes, and seconds, in that order.

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

e) Choose the desired date format.

f) Set today's date.

g) If you want the clock to automatically reset itself to conform to Daylight Savings Time (Summer Time), use the Begins and Ends fields to specify when Daylight Savings 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.

C) (Optional): Navigate to Screen 2 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, and use the rotary encoder 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 8400 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. Starting with Version 2.1, you can recall any factory or user preset, and can activate Bypass mode (for scheduled network testing) or Exit Bypass.

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) 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

If version 2.1 or higher is installed in your 8400, you can use multi-level passcodes to control access to the 8400 via the front panel and via PC Remote. You can configure a given passcode to allow one of the following five levels of access:

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

4. Presets and Automation

5. 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.

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 8400 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, 2, 4, and 8 hours, or Off.

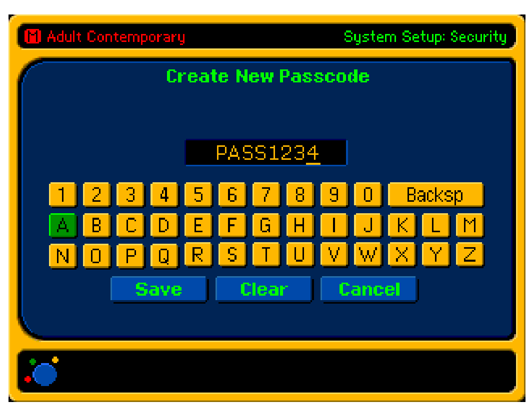
Front Panel lockout only occurs when the lockout value is not Off.

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. 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 when you are finished. The System Setup screen appears.

Choosing Done on the Security Screen automatically saves all of your Passcode settings. If you would like to cancel any new settings, Escape out of the screen via the *Escape* button on the 8400 (or via your PC's ESC key if you are connected through PC Remote).

4. Edit or delete an existing passcode (optional).

A) Navigate to System Setup and then to Security

If the 8400 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 8400 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, an "Invalid Passcode" message appears.

C) Choose Unlock.

You will be able to access 8400 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 2-41. Provided you have All Screens privileges, you can set the delay as desired by following the instructions in that step.

8400 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 to prevent your competitors from seeing them if your 8400 is installed in a shared facility.

The diagnostic and contrast 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 8400, there is one default passcode: ADMIN (all capitals), which has All Screens privileges. This passcode permits an initial connection to the 8400/PD via PC remote; you must enter ADMIN when PC Remote asks you for a passcode.

The 8400/PD has a blank front panel. Only an external PC running the 8400 PC Remote application can control it.

The front panel lockout feature's default setting is Off, so standard 8400s (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 8400 (via step 3 on page 2-41) 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 8400 from that PC. Conversely, if you connect to the 8400 with a Passcode that only allows access to the "Presets" on the 8400, 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 8400 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 8400 will allow the PC Remote application to connect via direct, modem and Ethernet connections at the level authorized by the passcode.

1. If no Passcodes are assigned to 8400 except the ADMIN default Passcode:

When you attempt a connection to the 8400 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 8400 via the PC Remote.

To ensure that your 8400 is fully protected, create a new passcode that has All Screens access. Then delete the ADMIN passcode.

See step 3 on page 2-41 for instructions on how to create a new passcode and step (4.C) on page 2-42 for instructions on how to delete a passcode.

2. Using passcodes to end PC Remote connections from the 8400 front panel:

If you try to access an 8400 from its front panel while a remote connection exists, a message will appear asking you whether you want to interrupt the remote connection. If you choose to interrupt the connection, the "Enter Passcode Screen" will appear. Only an All Screens passcode will cause the remote connection to disconnect. The 8400 ignores passcodes with lower access levels.

Doing a Software Update to an 8400 with Version 2.1 or Higher Already Installed:

The following information applies to updates done on 8400s that already have multilevel security, meaning 8400s with version 2.1 or higher software.

When you establish a connection between the PC and the 8400 to do an update, a prompt will appear during the update process asking you for a passcode. You can only do an update via a passcode with All Screens privileges. All other Passcodes will yield an "Invalid Passcode" message.

This prevents a hacker from interrupting programming by connecting to your 8400 and doing an unauthorized software update.

If you have forgotten your "All Screens" passcode…

You can access the 8400 even if you have forgotten your All Screens passcode. There are several ways to do this.

1. If your unit is an 8400 (i.e., if it has a full-featured front panel):

A) Press the *Enter* button within one second after the 8400 displays its initial splash screen upon boot-up.

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

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:

- 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. If your unit is an 8400/PD (i.e., if it has a blank front panel):

There are two ways to unlock an 8400/PD: the method described immediately below, and via a connection to a PC running a terminal program. The latter is described in **Administering the 8400 via a Terminal Program on a PC** on page 2-47.

A) Remove AC power from your 8400.

B) Connect pins 2 and 3 on Serial Port 2 of your 8400.

You can prepare a special dB9 connector with pins 2 and 3 soldered together, although it is probably easiest to make the connection by a bent paper clip inserted into pins 2 and 3 of the null modem cable shipped with your 8400. Maximum voltage on this clip with respect to ground is 12 volts DC, which is very unlikely to cause an electric shock. However, do not use the "paper clip" technique if it violates the safety regulations in your country.

C) Apply power to your 8400 with pins 2 and 3 still connected. Keep pins 2 and 3 connected for at least 70 seconds after application of power.

This will create a new passcode, ADMIN, having All Screens access. You can now use this passcode to access the security screens and administer existing passcodes as desired.

All existing passcodes will be retained.

To maintain good security, it is important create a new All Screens passcode and then delete the ADMIN passcode. You should also delete any old passcodes that may compromise security.

See step 3 on page 2-41 for instructions on how to create a new passcode and step (4.C) on page 2-42 for instructions on how to delete a passcode.

This procedure will also work with a standard 8400, which has a full-featured front panel. However, it is more convenient to use the procedure in step 1 on page 2-45 instead.

D) Remove the connection between pins 2 and 3.

Administering the 8400 through Serial Port #2

You can connect a PC to the 8400's serial port #2, using a terminal program like HyperTerminal to administer security and to recall presets. This section gives details.

- To facilitate maintaining security at sites shared with others, the 8400 monitors Serial 2 for 30 minutes after power-up or after the last valid command is received, after which all commands at Serial 2 are ignored except for recalling a preset.
- The 8400's Serial 2 interface can be used with any computer or terminal that is compatible with the RS-232 standard interface. The character code supported is ASCII.
- Users will connect their computer or terminal to the 8400 with the supplied null-modem cable. Only direct connections are supported; there is no provision for communications via modem at Serial Port #2.
- Communications configuration is 9600, N, 8, 1, no handshaking (flow control = none).
- Valid commands are in either upper or lower case, not a combination.
- Only one valid command is permitted per line.
- The 8400 will not respond to unrecognized commands.

In the following tables of commands and responses:

Text that the user enters appears in **`MONOSPACED BOLD`**.

Responses that the 8400 transmits at Serial 2 port appear in `monospaced normal`.

The symbol "↵" means CR (for received commands) and CR+LF (for transmitted responses from 8400).

Connecting to the 8400 via a Terminal Program on a PC

The procedure in step 2 on page 2-44 provides a simple means to regain access to an 8400 from which you are locked out. You can also do this via a PC running a terminal program like HyperTerminal.

To maintain security, you can only administer the 8400 through Serial Port #2 for 30 minutes after the 8400 has been powered up or after the last command is received. However, there is no timeout for the "recall preset" functionality.

A) Connect an available RS232 serial port (COM port) on your computer to Serial Port #2 on the 8400.

You do not need to remove power from either your computer or the 8400 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.

You can now type in commands described in the specification in **Administrative Operations**, immediately below.

Administrative Operations

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 either upper- or lower case.

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 8400/8400/PD:

The following command assigns an All Screens passcode. This passcode is then available from the front panel or when you connect normally via the 8400PC application (through the 8400’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 8400PC:

- Passcode length must be 1– 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** is the specified IP address;
yyy.yyy.yyy.yyy is the present (or default) IP address in use.

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** is the specified subnet mask;
yyy.yyy.yyy.yyy is the present subnet mask in use.

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 Modem Init string:

Applies only for modem connections (via 8400's Serial 1 port)

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

The 8400 appends CR+LF to the modem init string as transmitted to a modem (physically connected to Serial 1). The 8400 will not perform any case conversion to the argument (i.e., lower case arguments will be transmitted to the modem as lower case).

6. 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↵

7. To recall a preset:

Command	Response
RP XXXXXXX[PASSCODE]↵	(valid passcode and preset name) ON AIR: XXXXXXX (invalid passcode) ERROR: [PASSCODE] DOES NOT EXIST (invalid preset name) ERROR: XXXXXXX DOES NOT EXIST

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

- If a non-existent preset name and/or an invalid passcode is entered, the 8400 will ignore the command.
- You can apply this command anytime after the 8400 boots up. The 30-minute timeout does not apply.
- This command is useful in interfacing automation systems to the 8400.

Diagnostic Operations

These provide a status report indicating essential information:

Command	Response
ST↵	8400 Status: Ver. 2.1.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 PC Card 1: not installed PC Card 2: not installed↵

Remote Control Interface Programming

[Skip this step if you do not wish to program the opto-isolated GPI remote control interface at this time.]

Note: To facilitate doing EAS tests, units running pre-V1.0 software are hard-coded 8400 with remote 1 set for Bypass and remote 2 set for Exit Test. With V1.0 and higher, 8400 remote control supports user-programmable selection of up to eight inputs for various 8400 functions. We recommend that you always upgrade to the latest available 8400 software.

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 by the control interface.

Input: Analog: selects the analog inputs.

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 the Tone Test Mode preset on the air.

Exit Test: If a test preset is switched on the air, exit test reverts to the previous processing preset.

Stereo: switches the 8400's stereo encoder on.

Mono From Left, Mono From Right, or Mono From Sum: switches the 8400's stereo encoder off, using the Left, Right, or Sum (L+R) respectively, as the program source.

SCA Comp 1, or SCA Comp 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 Minute: resets the internal clock to the nearest minute. For example, 3:03:10 would be reset to 3:03:00, while 3:53:40 would be reset to 3:54:00. Use this function to re-sync the 8400's internal clock periodically to your station's master clock.

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.

Networking

[Skip this step if you do not wish to network to 8400, either for downloading software upgrades or for PC Remote Control.]

As of version 1.0 and higher, Orban supports two different Ethernet PC Cards for use in the 8400's rear PC Card slot: The Linksys EC2T Combo PCMCIA Ethernet Card, and the Linksys NP10T PCMCIA Ethernet 10BaseT Card. Use *only* one of these cards if you wish to implement Ethernet networking with your 8400.

Do not install or remove a PC card in the 8400 when the 8400 is powered.

1. Installing the card:

A) Remove power from the 8400.

B) Insert the network card in the rear PC card slot.

The card cannot be "hot-swapped"; be sure to power down the 8400 before installing it.

C) Restore power to the 8400.

2. Set up network parameters.

The 8400 uses the TCP/IP network protocol. See your network administrator to get the data required in the following procedure.

You can only perform the following procedure from the 8400 front panel—you cannot do it with the 8400 PC Remote application. You can also change the IP address and subnet mask via serial port #2. See **Connecting to the 8400 via a Terminal Program on a PC** on page 2-47.

3. Prepare the 8400 for a network connection:

In software versions earlier than version 2.1, you could select Direct, Modem, or Network connections. However, even in this older software, the network connection was always active even in the connection type was set to Direct or Modem. Therefore, in version 2.1, we removed Network as a connection type. This doesn't change any functionality; the network connection is still always active.

A) Set the IP address:

a) From the main menu, navigate to System Setup / Network Remote.

b) Toggle the Locate button to select IP Address and the Set Address button. The IP Address should read 192.168.155.101. Press the *Enter* button to access the Set IP Address screen.

192.168.155.101 is the 8400 default setting.

c) Using the Locate button, toggle to Clear, then press the *Enter* button.

This will allow you to enter your IP address.

d) Use the Locate button to toggle 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, toggle to Save and press the Enter button.

B) If necessary, set the Subnet Mask assigned by your network administrator:

a) Toggle the Locate button to select Subnet Mask and the Set Subnet button.

The Subnet Mask should indicate 255.255.255.0.

b) Press the *Enter* button to access the Set Subnet Mask screen.

c) Use the Locate button to toggle 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.

C) Connect your network's Ethernet cable to the card.

This completes setup of network parameters.

4. Edit "8400.ini" as necessary.

When connecting the PC Remote application to the 8400 via an Ethernet network, you must enter the IP address assigned by your network administrator into the `8400.ini` file. The `8400.ini` file is normally installed on your PC in the following locations:

- A shortcut to `8400.ini` in your Start Menu: Start / Programs / Orban / Optimod 8400; then select `8400.ini`

  OR

- The actual file location of `8400.ini` on your local PC drive: `c:\Program Files\Orban\Optimod 8400\`

When connecting the PC Remote application to the 8400 via an Ethernet network, type the following into the `8400.ini` file, using a text editor like Notepad:

- The IP address that you entered in step (3.A) above, followed by a carriage return.

  For example:
  `###.###.###.###`
  [carriage return]

Do not enter the words "carriage return"; instead, add a carriage return by pressing the Enter key on your computer's keyboard.

If you revert to connecting from PC Remote via your computer's serial port (either via direct connect, RAS null modem, or through a modem) then you must enter the following IP address into `8400.ini`:

`192.168.168.101`
[carriage return]

Do not enter the words "carriage return"; instead, add a carriage return by pressing the Enter key on your computer's keyboard.

`192.168.168.101 is` the data in the default `8400.ini` file that is installed automatically when you install the PC Remote application. *If the IP address in this file is wrong, the PC Remote application will be unable to connect to your 8400 if you attempt the connection by RAS null modem or through a modem.*

5. Edit "update.ini" as necessary.

When updating your 8400 system software via an Ethernet network, you must enter the IP address assigned by your network administrator into the `update.ini` file. The `update.ini` file is normally installed on your PC in the following location on your local drive:

- `c:\Program Files\Orban\Optimod 8400\`

When updating your 8400 system software via an Ethernet network, type the following into the `update.ini` file, using a text editor like Notepad:

- LINE 1: The IP address that you entered in step (3.A) above, followed by a carriage return.
- LINE 2: The following characters: 8400_new.zip

For example:
`###.###.###.###`
[carriage return]
`8400_new.zip`

Do not enter the words "carriage return"; instead, add a carriage return by pressing the Enter key on your computer's keyboard.

Setting Up an 8400 Memory Card

As of version 1.0 and higher, Orban supports SanDisk Flashdisk 8MB and 16MB Memory Cards (SanDisk part numbers SDP3B-8-101-B4 and SDP3B-16-101-B4 respectively). Use *only* these cards if you wish to use a memory card with your 8400.

CAUTION!
Do not install or remove a Memory card in the 8400 when the 8400 is powered. This could cause the on-air audio to mute for 50 seconds.

1. Installing the card:

A) Remove power from the 8400.

B) Insert the Memory card in the front PC card slot.

The card cannot be "hot-swapped"; be sure to power down the 8400 before installing it.

C) Restore power to the 8400.

2. Backing up and restoring user presets to the memory card:

A) From the 8400 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) From the System Setup screen, toggle the *Locate* button to select the Memory icon, and then press the *Enter* button.

C) Toggle the *Locate* button to highlight either user presets saved on the 8400 or user presets saved on the memory card.

D) Toggle the *Locate* button to highlight Copy, Copy All, or Delete, and then press the *Enter* button to use that key functionality.

E) Toggle the *Locate* button to highlight Done and press the *Enter* button when you have completed backing up or restoring user presets on the Memory Card.

Installing 8400 PC Remote Control Software

This section summarizes the procedure for installing 8400 PC Remote software on existing 8400s. You will find more detailed instructions in the .pdf file automatically installed on your computer by Orban's installer program, `Setup8400_x-y_remote.exe`, where "x" and "y" represent the software version you are installing. (For example, for version 1.0 software, x = 1 and y = 0.)

The PC Remote software is supplied on a CD shipped with your 8400. You can also download it from `www.orban.com`. Click "Downloads" and navigate to the 8400 folder.

Section 3 of this manual contains instructions for using the PC Remote software.

Installing the Necessary Windows Services

The Optimod PC Remote application uses Windows services to communicate between your Windows PC and your 8400. If you wish to remote control your 8400 from your computer, you will have to install these services. This section summarizes the procedure for installing these services. You will find more detailed instructions below.

- If you want to communicate through a local PC, you will need to connect a serial (COM) port of the PC to the SERIAL 1 port of your 8400 through a null modem cable (supplied with your 8400), and then use Dial-Up Networking with a RAS null modem (Windows 98) or Direct Cable Connect (Windows 2000 and XP) to make the basic connection. (The Orban update installer application installs the necessary .inf file for the RAS null modem.)

  The RAS null modem is a file that installs a "virtual modem" in Windows. It looks like a modem to Windows, but instead allows your computer and the 8400 to communicate directly between their serial ports through a null modem cable. To use it, you establish a new Dial-Up Networking Connection and specify the RAS null modem when asked which modem you would like to use. You then treat this Dial-Up Networking connection as you would any other Dial-Up Networking connection that uses a real, physical modem.

- If you want to communicate through a pair of modems, you will use the Windows Dial-Up networking service to make the connection.

- If you want to communicate through an Ethernet network, your PC must be set up for TCP/IP Ethernet networking, your 8400 must be running version 1.0 software or higher, and the optional Ethernet PC Card must be installed in your 8400's rear-panel PC card socket. (See **Networking** on page 2-52.)

Check Hardware Requirements

To use the Optimod PC Remote application, you will need the following:

- Orban 8400 OPTIMOD-FM.
- **If connecting by cable**: a null modem cable (also called a "reverse" cable), supplied by Orban with your 8400 when it was shipped. This cable has dB9 female connectors at both ends for connecting the 8400 to the serial port on your computer. If your computer has a dB25 connector, you will need an adapter.
- **If connecting by modem**: a 3Com/U.S. Robotics® 56kbps fax modem EXT for the 8400 side of the connection. Note that Orban has not certified any other type of modem, although most modern modems are likely to work.
- **If connecting by network**: An installed Ethernet PC Card in the rear slot of your 8400 (see **Networking** on page 2-52). If you intend to connect your 8400 directly to your computer's Ethernet port, you will also need a "crossover" Ethernet cable and a female/female RJ45 in-line adapter to connect the crossover Ethernet cable to the cable emerging from your 8400's PC Card. Version 1.0 software (or higher) must already be installed in your 8400.
- Pentium, 80486 or compatible computer, running Windows 98, 2000, or XP.

Your computer must meet the following minimum requirements to run the Optimod PC Remote application. This application will work best with at least the recommended requirements. (See table below.)

Recommended Requirements:	
Computer	Pentium (any speed)
Disk Space	25MB
RAM	32MB
Display	16-bit or 24-bit SVGA
Microsoft Windows	98, 98SE, 2000, or XP
COM Port	16550 (or compatible) UART
Minimum Requirements:	
Computer	486DX 66MHz or compatible
Disk Space	25MB
RAM	16MB
Display	VGA
Microsoft Windows	98
COM Port	16550 (or compatible) UART

WARNING!

When connecting your 8400, it is essential to use shielded cable to protect the pins in the RS232 connector from electrostatic discharge.

WARNING!

See **Networking** on page 2-52 before attempting to install any network interface cards.

Your computer must meet the following minimum requirements to run the Optimod PC Remote application. This program will work best with at least the recommended requirements. (See table above.)

Running the Orban Installer Application

You have should have received `Setup8400_x-y_PC_Remote.exe` (where `x-y` represents the version number of the software) on a CD packed with your 8400. If you are planning to use PC Remote, you must install the version that matches the software installed in the 8400 it is controlling. You cannot "mix and match" different versions of 8400 firmware and PC Remote software.

> `Setup8400_x-y_PC_Remote.exe` is also available via Internet download from www.orban.com (click on `Downloads` and navigate to the 8400 folder). If you obtain it via the Internet, just download `Setup8400_x-y_PC_Remote.exe` and run it to install the software automatically.

A) Insert the CD into your computer's disk drive. Using Explorer, navigate to your CD drive and double-click `Setup8400_x-y_PC_Remote.exe.` Follow the instructions on your screen.

This program adds an Orban/Optimod 8400 folder to your computer's Start Menu. This folder contains shortcuts to the PC Remote application.

B) You have now installed all files necessary to use the PC Remote software.

C) The next step is to install and configure the Windows communications services that allow your computer to communicate with your 8400. Because these procedures and instructions are subject to change, the instructions are located in a file called 8400_Vxxx_installion.pdf (where xxx represents the version number of the software), which you can access from the Orban/Optimod 8400 folder in your computer's Start Menu. You can use Adobe's .pdf reader application to open and read this file. The .pdf reader, it is available for free download from www.adobe.com.

Conclusion

By carefully following the instructions in `8400_Vxxx_installion.pdf`, you should have successfully installed the necessary Windows services and connected to your

8400. However, if you experience any problems with this process, or have any other 8400 questions, please contact Orban Customer Service:

phone: +1 510 351-3500

email: custserv@orban.com

For details on your new 8400 software, from new features to operational suggestions, refer to our FTP site (`www.orban.com/8400`; click "Downloads").

About 8400/PD

The 8400/PD is identical to the 8400 except that the 8400/PD has a blank front panel and must therefore be controlled and administered remotely through its serial or Ethernet ports.

Use the included 8400 PC software to communicate to the 8400/PD via its Ethernet network card (supplied with 8400/PD), an external modem, or null modem cable connected to its serial port #1. See **Using the 8400 PC Remote Control Software** in Section 3 of this manual.

Windows 98, Windows 2000, and Windows XP are the only supported operating systems. Orban software for 8400 is not compatible with Macintosh, UNIX or UNIX-variant systems.

For Ethernet connections, the 8400's factory default IP address is 192.168.155.101. Its default subnet mask is 255.255.255.0. These can be changed by the end user with one of the methods described below.

For connections through Serial 1 (direct cable connect, RAS null modem, and modem), the 8400's factory default IP address is 192.168.168.101, which cannot be changed by the user. This address is separate from the address used for Ethernet connections.

Whether you choose to connect to 8400 with an Ethernet connection or through Serial 1, the PC application must know which IP address to use. A special initialization file contains this address. For the 8400PC control application, the initialization file is `8400.ini`; for connections to make software updates, the initialization file is `update.ini`. Both of these files are located in the same folder where each application was installed, and can be edited with any plain ASCII text editor, such as Windows Notepad.

If you want to connect to different 8400s using different IP addresses from the same PC, you must edit the initialization file each time you change addresses.

You can change the IP address and subnet mask for an Ethernet connection in two ways:

1. Using the PC Remote application:

A) Edit your `8400.ini` file to contain the 8400's default IP address (192.168.155.101). See step 4 on page 2-54.

B) Launch the Orban PC Remote application. It should connect to your 8400/PD after requesting and receiving your passcode. If you wish to change the IP address and/or subnet mask, you must enter a passcode with All Screens privileges.

The default All Screens passcode for version 2.1 is "ADMIN." If this passcode does not allow you to access your 8400/PD through PC Remote, someone may have deleted it and replaced it with another All Screens passcode. If ADMIN does not allow you access to your 8400/PD (and you have not previously deleted this passcode and replaced it with another), refer to **If you have forgotten your "All Screens" passcode** on page 2-45.

C) From the main menu, navigate to System Setup / Network Remote.

D) Enter the desired IP address and subnet mask in the appropriately named fields.

E) Quit the PC Remote application.

The IP address and subnet mask that you entered will now be accepted by the 8400/PD.

The next time you start the PC Remote application, use the new IP address and subnet mask to communicate with your 8400/PD.

The `8400.ini` file is automatically updated to reflect your changes.

2. Using a terminal program connected to the 8400's serial port #2:

A) Connect to your 8400 by following the instructions in **Administering the 8400 through Serial Port #2** on page 2-46.

B) Follow instructions in steps 3 and 4 starting on page 2-49 to set the IP address and subnet mask.

About 8400HD

The 8400 HD ("High Definition Digital Radio") option's output is designed to feed streaming, netcasting, and digital radio channels, with particular emphasis on the iBiquity® High Definition Radio system (formerly known as "IBOC"—"In-Band On-Channel") approved for use in the United States.

The 8400 HD option consists of two circuit boards that are installed in your 8400 above the existing DSP and input/output boards. The added DSP board implements the digital-channel look-ahead limiter and other miscellaneous signal processing functions. The digital-channel input/output board adds an AES/EBU output, an AES/EBU sync input, and left and right analog outputs. The analog outputs can be configured to emit the 8400's low-latency monitor signal or to emit the signal that is processed for the HD digital channel.

An advanced-design look-ahead limiter that operates in parallel with the FM peak limiting processes the signal at the added AES/EBU output. The look-ahead limiter (which receives the 8400's "monitor" signal) is optimized to make the most of the limited bitrate codec (96 kbps) used in the High Definition Radio system's digital channel. By eschewing any clipping, the digital-channel 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 look-ahead limiter includes a parametric high frequency shelving equalizer that can be placed either pre-gain reduction or post-gain reduction. You can use it to equalize texture disparities between the analog and digital channels and to reduce codec artifacts at high frequencies.

15 kHz band limiting of the digital output also reduces codec artifacts. While the codec used in the High Definition Radio system can provide a satisfying experience for the consumer, at 96 kbps it is not audibly transparent—transparency would require a higher bitrate. By not wasting bits encoding the 15-20 kHz frequency range (which few radio listeners can hear), the codec instead provides higher quality encoding of the crucial 20-15,000 Hz band.

The HD output is designed to feed digital channels without pre-emphasis, which include almost all such channels. The only high-quality digital channels using pre-emphasis of which we are aware are NICAM channels (which use J.17 pre-emphasis) and some older CDs (which use EIAJ—50µs/15µs shelving pre-emphasis). If you use the HD output to feed a digital channel with pre-emphasis, you must allow extra headroom to compensate for the unpredictable peak level changes that the pre-emphasis induces.

If the HD output is driving a channel without pre-emphasis, it will control peak levels with an uncertainty of less than 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 for this task.

If not factory-installed, the HD kit comes with stand-alone instructions on how to field-install it.

Delay Difference between Digital-Channel and FM Outputs

In order to make the receiver analog/digital crossfade free from comb filtering, the time delays in the HD Radio's analog and digital channels must have a fixed and predictable offset, correctly implementing the HD Radio receiver's "time diversity" processing. The 8400's HD output's delay is automatically adjusted so that it always exactly 10.500ms longer than the FM output's delay, regardless of the FM output's delay (which can vary depending on processing settings). Therefore, the HD Radio exciter should be preset to compensate for this 10.5 ms offset between the FM output and HD output. Once this has been done, the time diversity delay will always be correct even if you choose a different 8400 preset.

HD I/O Setup Controls

Located in Input/Output Screen 5

Section Label	Control Name	Values
Monitoring	Phone Src	HD/FM Mon
	Meter Sel	FMOutLevel/HD GR
	HD HF EQ	Pre/Post
Analog Output	Out Source	HDpre/HDpost/FM Mon
	Out Level	–6… +24 dBU
Digital Output	St./Mono	Stereo/MonoL/MonoR/MonoSum
	Out Level	0… –20 dBFS; 0.1 dB steps
	Samp Rate	32kHz/44.1kHz/48 kHz
	Word Leng	14/16/18/20/24 bits
	Dither	In/Out
	Sync	Internal/Sync In
	Format	AES/SPDIF

Monitoring

Phone Src (“Headphone Source”) determines the source feeding the 8400’s front-panel headphone jack. You can select the digital-channel feed or the signal that feeds the FM analog outputs. The latter will be either the fully processed FM feed or the FM low-delay monitor feed, depending on how the FM analog output is configured.

Meter Sel determines if the Main Meter screen will display the left and right output levels of the FM processing or the left and right gain reductions of the digital-channel look-ahead limiter.

> 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 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.

Analog Output

Out Source (“HD Analog Output Source Feed”) determines if the HD analog output receives the signal before the digital-channel look-ahead limiter, the final post-limiter digital-channel signal, or the FM low-delay monitor feed. The latter can be useful if you wish to use the low-delay monitor feed but you need the FM analog outputs to carry the main processed FM feed.

> When you choose the digital-channel pre-limiter feed, this will also include the effects of the digital-channel HF shelving filter if this has been set to Pre.

Out Level ("HD Analog Output Level") determines the peak output level (in dBu) of the HD I/O card's analog outputs.

Digital Output

St./Mono ("HD Output Stereo/Mono Mode") determines if the digital-channel output will be fed by the normal stereo output of the multiband compressor/limiter, or by a mono feed from the multiband compressor/limiter'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 8400 does not set the AES/EBU stereo/mono status bits to reflect the setting of this control. The AES/EBU status bits appearing at the HD output are always set "stereo" even when the two audio channels carry identical mono signals.

Out Level ("HD AES/EBU Output Level") sets the digital-channel output level with respect to digital full scale.

Samp Rate ("HD Output Sample Rate") sets the output sample rate of the digital-channel output.

> The 8400's fundamental sample rate is always 32 kHz, but the internal sample rate converter sets the rate at the 8400's digital output. This adjustment allows you to ensure compatibility with downstream equipment requiring a fixed sample rate.

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

> The largest valid word length in the 8400 is 24 bits. The 8400 can also truncate its output word length to 20, 18, 16 or 14 bits. The 8400 can also add dither, and you should set it to do so if the input material is insufficiently dithered for these lower word lengths.

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

> The amount of dither automatically tracks the setting of the Word Length control. This first-order noise shaped dither considerably reduces added noise in the midrange by comparison to white PDF dither. However, unlike extreme noise shaping, it adds a maximum 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.
>
> 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 8400.

Sync determines if the sample rate appearing at the digital-channel output is synced to the 8400's internal clock or to a signal appearing at the HD sync input.

This function is independent of the sync function for the FM digital output. The sample rates of the digital-channel and FM digital outputs will be identical when both are set to internal sync and the same sample rate.

If there is no sync signal detected at the HD sync input, the digital-channel output will always sync to the 8400's internal clock.

If you want the digital-channel and FM digital outputs to have identical sample rates when externally synchronized, you must apply identical reference sample rates to the HD sync input and to the FM sync input or the digital audio input as desired. (Set the FM digital output so that it syncs to the FM sync or digital audio input respectively.)

There is no means for internally synchronizing the digital-channel output's sample rate to the 8400's digital audio input or FM sync input. If you wish to sync the digital-channel output sample rate to the sample rate present at the digital audio input, you must apply the same sample rate to the digital audio input and the HD sync input.

Format determines if the digital-channel output follows the professional AES/EBU or consumer SPDIF standard.

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

HD Audio Controls

Included with each preset in Advanced Control.

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 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 EQ control in the I/O Setup section).

When placed before the look-ahead limiter, this equalizer is useful for reducing the audible disparity between the FM and digital-channel outputs. 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 8400's tuning for FM.

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.

This control is located in the Clippers screen (Advanced screen 7) in Advanced Control.

The factory default is +4.

Because its loudness must match the FM channel, there is no need to overprocess the digital channel. 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.

Section 3 Operation

8400 Front Panel

Headphone Jack allows you to monitor the output of the processing through headphones. When the HD option is installed, you can choose the output of the FM or HD processing. Headphone impedance should be 75Ω or higher.

The headphone jack is connected to chassis ground, not circuit ground. Therefore, the headphones will not work if the 8400 GROUND switch is in the Lift position and there is no other connection between the 8400's circuit and chassis grounds.

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 8400's color LCD displays the following meters and indicators:

In meters show the peak input level applied to the 8400'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.

> If the Five-Band structure is operational, all the meters display gain reduction (G/R) activity. If the Two-Band structure is operational, only the two leftmost meters display G/R activity.

Multiplex Power meter indicates the action of the ITU Multiplex Power controller. It shows how much the MPX Power Controller has reduced the clipper drive, thereby reducing the average power in the processed audio.

> This meter, labeled "P," is displayed on the 8400's screen. It always appears when the Two-Band Structure is active. If the Five-Band Structure is active, it only appears if the MPX Power Controller is turned on.

HD (High Definition Digital Radio) G/R meter shows the gain reductions in the left and right HD look-ahead limiters.

> This meter only appears when the HD option is installed and the Meter Sel. Switch (on page 5 of the Input/Output menu) is set to HD GR.

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 important to be aware of the many negative subjective side effects of excessive density when setting controls that affect the density of the processed sound.

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

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

> The 8400 uses look-ahead techniques in several parts of the processing to minimize overshoot for a given level of processing artifacts, among other things.

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 any others. 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!

At this writing, there has been a very disturbing trend in CD mastering to apply levels of audio processing to CDs formerly only used by "aggressively-processed" radio stations. These CDs are audibly distorted (sometimes blatantly so) before any further OPTIMOD processing. The result of 8400 processing can be to exaggerate this distortion and make these recordings noticeably unpleasant to listen to over the air.

There is very little that a radio station can do with these CDs other than to use conservative 8400 presets, which will cause loudness loss that may be undesired in competitive markets. There is a myth in the record industry that applying "radio-style" processing to CDs in mastering will cause them to be louder or will reduce the audible effects of on-air processing. In fact, the opposite is true: these CDs will not be louder on air, but they will be audibly distorted and unpleasant to listen to, lacking punch and clarity.

Another unfortunate trend is the tendency to put so much high frequency energy on the CDs that this energy cannot possibly survive the FM pre-emphasis/de-emphasis process. Although the 8400 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.

About the 8400's Signal Processing Features

By comparison to its predecessor (Orban's 8200), the 8400 contains new signal processing innovations that advance the state-of-the-art in FM on-air processing.

Dual-Mono Architecture

The 8400 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 8400 would always process stereo program material.

Signal Flow

The signal flows through the 8400 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

Each of these blocks is significantly improved by comparison to its predecessor (if any) in Orban's 8200.

Input Conditioning: The 8400 operates at 32 kHz sample rate and power-of-two multiples thereof (up to 512 kHz in the stereo encoder). No commercial A/D converters or sample rate converter chips convert to 32 kHz while maintaining the standards we demanded for this product. Therefore, to ensure high quality A/D and sample rate conversion, we operate both the SRC and A/D chips at 64 kHz-output sample rate and then downsample to 32 kHz in DSP. By designing and implementing our own downsampler, we can ensure full frequency response to 15 kHz with very low spurious images.

> Despite myths circulating in the marketplace regarding the supposed superiority of higher sample rates in FM stereo processors, be assured that 32 kHz is far preferable to 48 kHz as a basic sample rate for these devices. 32 kHz allows us to use DSP horsepower at least 1.5x more efficiently than we could at 48 kHz, adding features that *really* improve the sound. It also makes it easier to protect the stereo pilot tone and RDS subcarriers spectrally by strictly limiting our output bandwidth to 16 kHz. Although a 16 kHz bandwidth limitation is more than is strictly needed to protect the pilot tone, the RDS requires protection over a substantially wider bandwidth, and 16 kHz provides this protection.
>
> Like the 8200, the 8400's output spectral control is immaculate, ensuring maximum stereo and RDS coverage. Moreover, the 8400's digital output will pass through any uncompressed digital STL without added overshoot and without the need for distortion-producing overshoot compensation schemes.

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

Stereo Enhancement: The 8400 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 popular “Max” technique. This passes the L–R signal through a delay line and adds this “decorrelated” signal to the un-enhanced 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.

Two-Band Gated AGC: One of the 8200’s secrets was that its AGC is a two-band device, using Orban’s patented “master/bass” band coupling. The 8200’s LCD showed only the AGC “master” band gain reduction.

In the 8400, we bring out all of the two-band AGC controls so the user can adjust them, including thresholds, ratio, attack times, release times, and master/bass coupling. We also add an 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.

Another user-requested feature is the ability to operate the AGC in left/right or sum-and-difference modes. The user can preset the maximum amount of gain difference permitted between the sum and difference channels. Sum-and-difference can provide a different style of stereo enhancement than the 8400’s purpose-built stereo enhancers, which we nevertheless prefer for this application because of their more sophisticated gating and decreased tendency to add multipath distortion.

In sum-and-difference mode, you can operate the sum and difference channels with different thresholds. If you set the sum threshold higher than the difference threshold, you can force the AGC to reduce separation automatically on program material with excessive separation, like old Beatles records.

For the first time, 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.

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

Equalization: The 8400 improves on the 8200’s steep-slope bass shelving equalizer and adds three bands of fully parametric bell-shaped EQ.

In the 8400, you can set the slope of the bass shelving EQ to 6, 12, or 18 dB/octave and adjust the shelving frequency. This significantly adds to its versatility.

The 8400'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 8400 HF Enhancer is a program-controlled HF shelving equalizer that was originally introduced on Orban's 2200 OPTIMOD-FM. 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 a sound that is bright and present without being excessively shrill.

Multiband Compression: The basic sound of the five-band compressor is similar to the 8200's five-band compressor. However, in response to user requests we have increased the number of release time settings from four to seven, doubling the resolution of this audibly important parameter.

The multiband compressor now uses a "look-ahead" topology, which means that the program audio is delayed until the gain control signal has time to attack fully. This significantly reduces compressor overshoot and eases the burden on the following peak limiting stages.

Often, transient response sounds punchier if transients overshoot in the multiband compressor and hit the clippers instead. Consequently, we have made look-ahead operation defeatable in the multiband compressor. To optimize performance for all program material, we also offer an automatic option that turns the look-ahead on for speech material (to achieve lowest distortion) and off for music (to achieve best transient punch).

Like the 8200, we control high frequencies with distortion-canceled clipping. However, the clipper in the 8400 operates at 256 kHz-sample rate and is full anti-aliased.

> We are well aware of the controversy regarding the audible benefits of anti-aliasing a digital clipper and we continue to believe that it provides no audible benefit with program material in a well-designed system at sufficient sample rate. Nevertheless, we chose to put the issue to rest by adding anti-aliasing to all primary 8400 clippers.

The 8200 determined the gain reduction in band 5 from the gain reduction in band 4; these bands are only independent from the viewpoint of the downward expander and multiband clippers. In the 8400, we have added a high frequency limiter, which 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 8400's "back end" clipping system to do this.

We also improved on the embedded bass clipper that protects bands 1 and 2 from transient overshoot. In the 8400, you can adjust this clipper for "hard," "medium," or "soft" operation. Each step gives a further reduction in audible distortion by means of increasingly sophisticated signal processing. Each step from "soft" to "hard" adds a touch more bass harmonic distortion that can improve the apparent bass response of small receivers like clock radios. The hard clipper also has a shape control, allowing you to control the "knee" of its input/output transfer curve.

"Intelligent" Clipping: There have been major changes and improvements in the 8400's back-end clipper by comparison to the 8200. The clipping system is "what separates the men from the boys" in on-air processing. A good clipping system is the key to being simultaneously loud, clean, and bright.

In general, the improvements come in two areas: (1) more intelligence in preventing audible clipping distortion with difficult program material, and (2) better overshoot compensation. (1) means that the 8400 is much less likely to encounter program material that unexpectedly causes gross clipping distortion—a particular problem with systems that rely on simple composite clipping for peak control. (2) means that the 8400 controls overshoots more tightly than does the 8200, and does so with less audible distortion and HF loss.

We prevent excess clipping distortion by reducing the drive level to the clippers as required. Although this is done in a sophisticated, frequency-dependent way using look-ahead techniques to minimize audible side effects, you may hear some audible compression and intermodulation artifacts when you use this mechanism to excess. If you hear such artifacts and find them objectionable, you should reduce the setting of the MB Clipping (Multiband Clipping) control.

In most cases, reducing the setting of the Less/More control does this automatically.

One preset, LOUD-COMPRESSED, exploits these compression artifacts to achieve a dense, "highly compressed" sound, which some people like to use for certain rock and pop music formats.

DSP-derived Stereo Encoder: The 8400's stereo encoder is derived from algorithms first developed for the high-performance Orban 8218 stand-alone encoder. The 8400's stereo encoder operates at 512 kHz-sample rate to ease the performance requirements of the D/A converter's reconstruction filter, making it possible to achieve excellent stereo separation that is stable over time and temperature.

The 8400 has two independent composite outputs, whose levels are both software-settable. The second output can be configured to provide a 19 kHz-reference output for subcarrier generators that need it. For convenience, two SCA inputs sum into the 8400's analog composite output amplifier.

The 8400 does not digitize SCAs.

Composite Level Control Processor: 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 8400 offers a composite processor that provides excellent spectral protection of the pilot tone and SCAs (including RDS), while still providing approximately 60 dB separation when a single-channel composite waveform is clipped to 3 dB depth. To ensure accurate peak control, it operates at 512 kHz sample rate.

> Like conventional composite clipping, the new algorithm 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 end up at new frequencies not harmonically related to the original frequency that generated them.
>
> The 8400's composite processor has a defeatable 19 kHz notch filter to protect the pilot tone. 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. The notch filter will cause overshoot with some program material because it removes spectral energy that would otherwise help control peak levels. We therefore allow the user to trade off overshoot against pilot tone protection by choosing whether the 19 kHz notch filter is in or out.

We still prefer to use the 8400's main clipping system to do the vast majority of the work because of its sophisticated distortion-controlling mechanisms. This means that the 8400, unlike some of its competitors, does *not* rely on composite processing to get loud. Therefore, people using its left/right-domain AES3 digital output can enjoy the loudness benefits of the 8400's processing—the 8400 gets *very* loud without composite clipping.

ITU-R 412 Compliance

ITU-R 412 requires the "average multiplex power" to be limited to a standard value. The 8400 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 8400 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 8400, preventing excessive reduction of the multiplex power.

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

Patents are pending on this system.

Two-Band Purist Processing

In addition to five-band processing, suitable for pop music and talk formats, the 8400 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 8400’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

The HD (“High Definition Digital Radio”) output is the de-emphasized output of the 8400’s two-band or five-band compressor/limiter (depending on the preset in use), as processed through an advanced-design look-ahead limiter. This limiter minimizes IM distortion in addition to harmonic distortion. The resulting peak limiting is almost always undetectable when used with reasonable amounts of gain reduction (i.e., frequently reoccurring 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 alarm 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. 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.

The 8400 offers this adjustment in the form of a parametric high frequency shelving filter, which can supply a high frequency rolloff to tame 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.

See **About HD** in Section 2 of this manual for a complete description of all HD setup and subjective adjustment controls.

Input/Output Delay

The algorithmic improvements in the 8400 have one significant cost—the input/output time delay is a minimum of approximately 15 ms and can be as high as 40 ms, depending on the setting of the BassClpMode control. To make intelligent decisions about how to process, the 8400 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.

20 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.

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 ms, where the talent starts to hear slap echo in addition to frequency response colorations.

Summary

The 8400 provides numerous important advances over the digital signal processing in Orban's popular 8200. By devoting over five times the DSP processing power to the problem, the 8400 reduces bass distortion, opens up high frequency response, and prevents audible clipping distortion on difficult program material. Added stereo enhancement, high frequency enhancement, and composite peak control complement the basic processing improvements. Together, they provide the most powerful and effective on-air FM processing system that Orban has yet created.

Customizing the 8400's Sound

The subjective setup controls on the 8400 give you the flexibility to customize your station's sound. Nevertheless, as with any audio processing system, proper adjustment of these controls consists of balancing the trade-offs between loudness, density, and audible distortion. The following pages provide the information you need to adjust the 8400 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 8400 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 8400'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 8400 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 hundred 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 8400, Less/More affects only the dynamics processing (compression, limiting, and clipping). Unlike the 8200, the 8400 has equalization and stereo enhancement that are decoupled from Less/More. You can therefore change EQ or stereo enhancement and not lose the ability to use Less/More. When you create a user preset, the 8400 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.

For the first time in an Orban FM processor, compressor attack times and thresholds are available, along with settings affecting the automatic clipping distortion control (new to the 8400). These controls can be exceedingly dangerous in inexperienced hands, leading you to create presets that sound great on some program material and 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 if you 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 8400's behavior when you switch pre-emphasis. All factory presets actually have two variations, one for 50µs and one for 75µs. The 8400 uses the appropriate variation is automatically. However, once you have created a user preset, it will no longer automatically switch its parameters when you change pre-emphasis. 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 pre-emphasis curve, Band 5 of the Five-Band Structure is capable of 30 dB of gain reduction.

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

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 8400.

In the 8400, a processing structure is a program that operates as a complete audio processing system. Only one processing structure can be on-air at a time. Just as there are many possible ways of configuring a processing system using analog components (like equalizers, compressors, limiters, and clippers), there are several possible processing structures that the 8400's DSP hardware could realize. Unlike an analog system, where creating a complete processing system involves physically wiring its various components together, the 8400 realizes its processing structures as a series of high-speed mathematical computations made by Digital Signal Processing (DSP) integrated circuit chips. In the 8400, both structures operate simultaneously so there is no delay in switching between them, which is done with a smooth cross-fade.

There are two basic structures: **Two-Band** and **Five-Band**.

Two-Band is a purist, phase-linear structure. When correctly configured it can be used for protection limiting, and we provide two presets that use it for this. It is also the basis 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.

Readers who are familiar with the 8200 should be aware that we have omitted the Two Band Normal structure from the 8400 because the Five-Band Structure is overwhelmingly preferred for formats that might otherwise use the Two Band Normal structure. The Two-Band Purist structure in the 8400 is substantially improved over its 8200 counterpart, with more effective HF limiting and clipper distortion control.

In fact, the 8200 *always* used its Two Band Purist DSP code to implement the Protection Limiter structure. The only difference between Two Band Purist and Protection was that Protection hid some of the controls ordinarily available in the Two-Band Purist structure. This was intended to simplify operation and reduce the opportunity to misadjust the Protection presets. Therefore, our approach to Protection in the 8400 is actually the same as it was in the 8200. However, in the 8400 we don't hide the extra controls.

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 8400'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 8400'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.

Factory Programming Presets

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! Orban presets are only suggestions. 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 LL in their names use the Hard LL bass clipper mode to achieve 15 ms input-output delay. Other presets have approximately 20 ms delay (5-band) and 23 ms delay (2-band).

FACTORY PROGRAMMING PRESETS		
Preset Names	**Source Preset**	**Normal Less-More**
PROTECTION	PROTECTION	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
COUNTRY-MEDIUM	ROCK-SMOOTH	7.0
COUNTRY-LIGHT	ROCK-LIGHT	7.0
FOLK-TRADITIONAL	ROCK-SOFT	7.0
GOLD	GOLD	9.5
GREGG	GREGG	9.5
GREGG LL	GREGG LL	9.5
IMPACT	IMPACT	9.5
IMPACT LL	IMPACT LL	9.5
INSTRUMENTAL	JAZZ	7.0
JAZZ	JAZZ	7.0
LOUD-BIG	LOUD-BIG	9.0
LOUD-FAT	LOUD-FAT	7.0
LOUD-COMPRESSED	LOUD-COMPRESSED	9.5
LOUD-HOT	LOUD-HOT	8.5
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
MINIMUM-DELAY	MINIMUM-DELAY	8.5
NEWS-TALK	NEWS-TALK	7.0
OLDIES-CLASSIC	OLDIES-CLASSIC	7.0
OLDIES-MODERN	OLDIES-MODERN	7.0

FACTORY PROGRAMMING PRESETS		
OLDIES-PROCESSED	OLDIES-PROCESSED	7.0
ROCK-DENSE	ROCK-DENSE	7.0
ROCK-LIGHT	ROCK-LIGHT	7.0
ROCK-MEDIUM	ROCK-MEDIUM	7.0
ROCK-MEDIUM+Mid-bass	ROCK-MEDIUM+Mid-bass	7.0
ROCK-MEDIUM+LOW BASS	ROCK-MEDIUM+LOW BASS	7.0
ROCK-OPEN	ROCK-OPEN	7.0
ROCK-SMOOTH	ROCK-SMOOTH	7.0
ROCK-SOFT	ROCK-SOFT	8.5
SPORTS	SPORTS	7.0
URBAN-LIGHT	URBAN-LIGHT	7.0
URBAN-HEAVY	URBAN-HEAVY	7.0

Table 3-1: Factory Programming Presets

PROTECTION: PROTECTION is a two-band preset with a high amount of band coupling. It is intended for use below threshold most of the time, 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: As their names imply, the CLASSICAL-Five-Band and CLASSICAL-Two-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-Two-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-Two-Band preset provides, use the CLASSICAL-Five-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.

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 OLDIES-MODERN preset. This preset tends to re-equalize them automatically. Other potential presets for this format include OLDIES-CLASSIC, ROCK-SMOOTH, and ROCK-LIGHT.

IMPACT: There are two IMPACT presets.

IMPACT is intended for CHR and similar formats where attracting a large audience (maximizing cume) is more important than ensuring long time-spent-listening. This is a loud, bright, "major-market" preset that is competitive with other processors that are not as scrupulous as the 8400 about controlling audible distortion with certain program material. It also has a great deal of presence energy to cut 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" without being turned down with Less-More.

IMPACT LL is the low-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 specifically tailored toward stations that play mostly instrumental music. It is a quiet preset with a very clean, mellow high end to prevent stridency on the saxophone solos so often heard in "light jazz" programming. The preset produces very low listening fatigue, so it is a good choice for stations that want listeners to stay all day.

LOUD: There are twelve LOUD presets.

LOUD-HOT was designed to be competitive with the sound of a popular processor from another manufacturer, but without the competitor's dependence on composite clipping. This preset 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 (as can the competing product!). In all cases, speech will be cleaner than with the competing product and the bass is not permitted to "shut down" the main clipper on bass waveform peaks.

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 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, but which has not previously been obtainable in an FM OPTIMOD product.

> The new DSP code first introduced in version 1.0 software greatly reduced the tendency of this preset to produce audible IM distortion and excessive pumping by comparison to the DSP in earlier software versions.

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. It was first introduced in version 2.0 software.

LOUD-FAT has dramatic punch on percussive material and a very fat-sounding low end, plus outstandingly effective distortion control. It exploits the DSP improvements introduced in version 2.0 to avoid overt bass distortion despite the full bass sound. It is slightly quieter than the loudest of the "loud" preset family.

GREGG and GREGG LL 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, these presets are similar to the LOUD-HOT+BASS presets.

> The GREGG presets were introduced with version 3.0 software.

MINIMUM DELAY: This is a general-purpose preset using low latency mode to achieve a delay of 15 ms (even though it does not have an "LL" in its name). We have retained this preset for compatibility earlier versions. We have also attempted to retain some of its open sound—a sort of "8200 on steroids"—while improving its high frequency distortion control and peak control properties by comparison to the MINIMUM DELAY preset in version 1.0 software.

OLDIES: There are four OLDIES presets.

GOLD is optimized for music recorded from the 1950s through the 1970s by exploiting 8400 DSP features that were unavailable when the original series of "Oldies" presets was created. (GOLD was first introduced with version 2.1 software.)

Although the original "Oldies" presets were tuned for a nostalgic, "AM-radio" type of quality, we expect that most oldies or "solid gold" formatted stations will prefer the new Gold preset because it is substantially louder and more "hi-fi"-sounding, while still respecting the limitations and basic flavor of the recordings from this era.

> 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 they are over-equalized in an attempt to emulate the high frequency balance of recently recorded material.

OLDIES-MODERN is designed for records from approximately 1965 to the present. It is a bit more conservative than the ROCK presets and is designed to effectively re-equalize recordings from different eras.

OLDIES-CLASSIC is specially tuned to complement "Top-40" records from the early 1950s through the mid-'60s. Recordings from this era often had extremely present vocals with audible soft clipping from the vacuum tube consoles and mic preamps used at the time. Soft clipping (either on the master tape or when the songs were cut to disc) was often used to make the records cut through when played on jukeboxes and AM radio (which used very simple audio processing back then). OLDIES-CLASSIC preserves the unique sound quality of these classic recordings without significantly changing the texture.

OLDIES-PROCESSED is tuned for the same records as OLDIES-CLASSIC, but is designed for a louder sound that will significantly change the texture of the source material. It is purposely tuned for a "retro" spectral balance with a hint of "AM-radio" quality—highs are smooth, mellow, and rolled-off, while the midrange is prominent.

NEWS-TALK: This preset is quite different from the others above. 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-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.

For Contemporary Hit Radio (CHR) we recommend the ROCK-DENSE or ROCK-MEDIUM versions. In competitive markets, you may need to use LOUD-HOT (you can use Less/More to get it even louder) or even LOUD-HOT+BASS or IMPACT. However, the "rock" presets are noticeably cleaner, and are therefore more likely to encourage longer times spent listening than are the "loud" presets.

For Album-Oriented Rock (AOR) we recommend the ROCK-MEDIUM or ROCK-OPEN versions, although you might prefer the more conservative ROCK-LIGHT or ROCK-SMOOTH 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.

URBAN: There are two URBAN (Rap) presets: HEAVY and LIGHT. These 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. Highly competitive Urban stations might also use LOUD-HOT+BASS or LOUD+SLAM, modified versions of LOUD-HOT that maximize bass punch.

Equalizer Controls

The table summarizes the equalization controls available for the Five-Band structure.[1]

Equalizer Controls			
Group	**Basic/Intermediate Name**	**Advanced Name**	**Range**
Bass Shelf	Bass Shelf Hinge Frequency	Bass Freq	80 … 500 Hz
	Bass Shelf Gain	Bass Gain	0 … 12 dB
	Bass Shelf Slope	Bass Slope	6,12,18 dB/Oct
Low	Low Frequency	Low Freq	20 ... 500 Hz
	Low Gain	Low Gain	–10.0 … +10.0 dB
	Low Width	Low Width	0.8 ... 4 octaves
Mid	Mid Frequency	Mid Freq	250 ... 6000 Hz
	Mid Gain	Mid Gain	–10.0 … +10.0 dB
	Mid Width	Mid Width	0.8 ... 4 octaves
High	High Frequency	High Freq	1.0 … 15.0 kHz
	High Gain	High Gain	–10.0 … +10.0 dB
	High Width	High Width	0.8 ... 4 octaves
---	Brilliance	Brilliance	0.0 … +6.0 dB
---	DJ Bass	DJ Bass	off, 0 … 10
---	HF Enhancement	HF Enhance	0 … 15

Table 3-2: Five-Band Equalization Controls

Most 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 Fac-

[1] 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.

tory 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 Five-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 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.

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 8400'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 8400'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 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. 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 control determines the amount of bass boost produced on some male voices. In its default Off position, it causes the gain reduction of the lowest frequency band to move quickly to the same gain reduction as its nearest neighbor when gated. This fights any tendency of the lowest frequency band to develop significantly more gain than its neighbor when processing voice because voice will activate the gate frequently. Each time it does so, it will reset the gain of the lowest frequency band so that the gains of the two bottom bands are equal and the response in this frequency range is flat. The result is natural-sounding bass on male voice.

If you like a larger-than-life, “chesty” sound on male voice, set this control away from Off. When so set, gating causes the gain reduction of the lowest frequency band to move to the same gain reduction (minus a gain offset equal to the numerical setting of the control) as its nearest neighbor when gated. You can therefore set the maximum gain difference between the two low frequency bands, producing considerable dynamic bass boost on voice.

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

The amount of bass boost will be highly dependent on the fundamental frequency of a given voice. If the fundamental frequency is far above 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.

Stereo Enhancer Controls

You can operate the stereo enhancer 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 3-7 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 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:

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
Rumble Filter 30 Hz HPF	30 Hz HPF	Off/On
Phase Rotator	Phase Rotat	Out/In

Table 3-3: Stereo Enhancer 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 prevent multipath distortion. 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.

Rumble Filter 30 Hz HPF determines if a 30 Hz 18 dB/octave highpass filter will be placed in-circuit before other processing. Although not a stereo enhancer control, it is found on the stereo enhancer page (for convenience) because, like the stereo enhancer, it can be adjusted without eliminating Less/More functionality.

Phase Rotator is not a stereo enhancer control. It 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 8400's back end processing.

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 8400 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 8400's new distortion control will work to help prevent audible speech distortion even when the phase rotator is switched out.

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.

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	0-100 %
---	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 Controls

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.

It 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 a studio level controller (like Orban's 8200ST). 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 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 than 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 8400. 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 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 Multiband Gate Thresh (Gate Threshold) control is turned Off, the DJ Bass control (in the Equalization screen) is disabled.

AGC Bass Coupling control sets the balance provided in the AGC between bass and the rest of the frequency spectrum.

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. The AGC 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 AGC Bass Coupling control at 100% will sometimes cause bass loss, the most accurate frequency balance will often be obtained with this control between 70% and 90%. The optimal setting depends on the amount of gain reduction applied and on the AGC release time. Usually, you will adjust the AGC Bass Coupling control until the Master AGC and Bass AGC Gain Reduction meters track as closely as possible unless you want the AGC to provide some gentle automatic re-equalization of the input material.

With the Master AGC Release control set to 2 dB/second, setting the AGC 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 AGC Bass Coupling toward 100% (wideband) do not sound good. Instead, set the AGC 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 Five-Band structure.

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 soundfield.

Window Size determines the size of the "target zone" window in the AGC. If the input level falls within this target zone, the AGC release time is set to the number specified by the Window Release control. This is usually much slower than the normal AGC release, and essentially freezes the AGC gain. This prevents the AGC from building up density in material whose level is already well controlled. If the level goes outside the window, then the AGC switches to the release rate specified by Master AGC Release, so the AGC can still correct large gain variations quickly.

> 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 8400'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 8400'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 8400 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.

> This control reduces the available range of AGC gain reduction because it acts by attenuating the gain control signal produced by the AGC sidechain.

The range is 25 dB at ∞:1 and 12 dB at 2:1. However, the range of input levels that the AGC can handle is unaffected, remaining at 25 dB.

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 "100%," 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.

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.

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.

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 8400 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 available in Two-Band presets only. It 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.

Linear was used in all Two-Band and Five-Band presets in v. 1.0 and below, and was the only available characteristic. It is still available for Two-Band presets only. While it 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.

Clipper Controls

Clipper Controls		
Intermediate Name	**Advanced Name**	**Range**
---	OSComp Dr	–2.0 … +2.0
Bass Clip Thresh	BassClpThre	–6.0 … 0.00
Bass Clip Mode	BassClpMode	Soft, Medium, Hard
---	Hard Clip Shape	0 … 10
HF Clipping	HF Clipping	0 ... 6.0 dB

Final Clip Drive	FinalClpDr	–3.0 … +5.0
(Composite) Clip Drive	Comp Drive	Off, 0 ... 3 dB
Pilot Protect	PilotProtect	Off, On

Table 3-5: Clipper Controls

The clipper controls are common to the Two-Band and Five-Band structures, except as noted in the control descriptions on the following pages.

All of the clipper controls are common to the Intermediate Modify and Advanced Modify screens, except OSComp Dr, which is only available in the Advanced Modify screen.

Bass Clip Thresh sets Orban's patented embedded bass clipper. It 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 don't smash against the back-end clipping system, causing overt intermodulation distortion between the bass and higher frequency program material.

Some 8400 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 distortion and pumping. Be careful when setting this control; do not adjust it casually. Listen to program material with heavy bass combined with spectrally sparse midrange material (like a singer accompanied by a bass guitar) and listen for IM distortion induced by the bass' pushing the midrange into the clipping system. In general, unless you have a very good reason to set the control elsewhere, we recommend leaving it at the factory settings, which were determined following extensive listening tests with many types of critical program material.

> In the Five-Band structure, the clipper is located after bands 1 and 2 are summed. In the Two-Band structure, the clipper is located after the Bass band.

Bass Clip Mode sets the operation of the bass clipper to Hard, LL Hard, Medium, or Soft.

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. So 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.

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 15 ms.

To minimize speech distortion, the speech/music detector will automatically switch the bass clipper to Medium when speech is detected, provided that the Five-Band structure is active, Latency is High, and the Bass Clip Mode is set to Hard. (See "Lookahead" on page 3-57 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 distortion that might otherwise occur when speech is applied to this clipper.

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.

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 20 ms of delay to the processing (so that the total is approximately 40 ms).

MED 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.

HF Clipping determines the amount of protection provided by the 8400's high frequency multiband clipper. This control was first introduced in the 8200 and allowed users to trade off distortion against brightness. Because of the improvements in the 8400's 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 loudness/distortion trade-off is primarily determined by the Final Clip Drive control.

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 8400'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.

Composite Clip Drive sets the drive level, in dB, into the 8400's composite limiter (which is not a clipper, despite this control's name).

> This control has no effect on the 8400's left and right analog or digital outputs.
>
> The Composite Clip 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 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. Mathematically, these harmonics will be located at the same frequencies as harmonics caused by clipping in a simple digital-domain clipper (with no anti-aliasing) operating at 38 kHz sample rate.
>
> 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 Clip Drive control setting proportionately.

Pilot Protection Filter turns the 19 kHz notch filter on or off. It affects the composite output only.

The 8400'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. (Fig. 3-1).

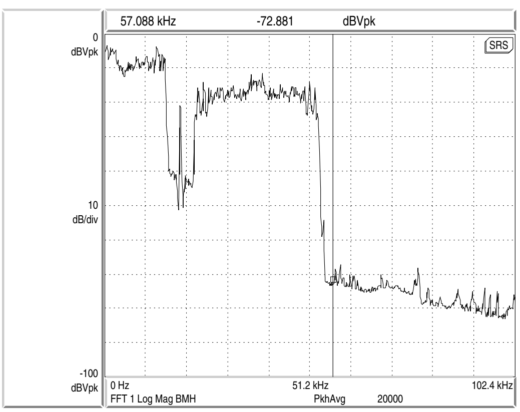


Fig. 3-1: 0-100 kHz Baseband Spectrum (Loud-Hot preset)

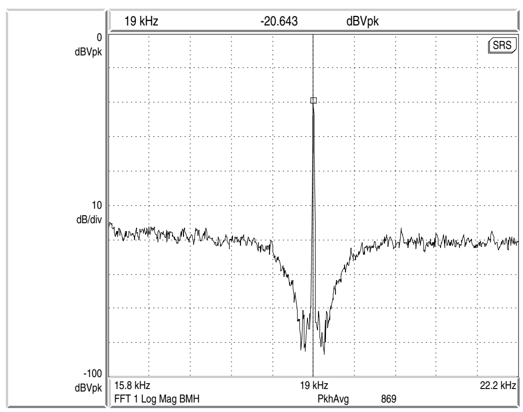


Fig. 3-2: 19 kHz Pilot Notch Filter Spectrum (Loud-Hot preset; detail)

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.

Advanced Clipper Controls

The following Clipper control is found only in the Advanced Modify screen.

OSComp Dr (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 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, and corresponds to the characteristic in version 0.94 and earlier software, where the transition between linear operation and flat-topping happens 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."

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 new color to the 8400's sound palette—one that is not duplicated by the existing 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 3-36.)

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. Like the "Two-Band Purist" structure in Orban's OPTIMOD-FM 8200, the 8400'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.

The Protection Presets

There are two Protection Factory Presets. Both use the same DSP code as the Two-Band Purist structure, but with the AGC defeated.

PROTECTION 0 dB sets the limiting threshold so that limiting almost never occurs, while PROTECTION 5 dB sets the limiting threshold so that program material at the maximum normal input level (as determined by a PPM or VU meter monitoring the input program line) produces an average limiting of 5 dB.

To set up the Protection Limiter, recall preset PROTECTION 0 dB if you want limiting to occur only when the program level *exceeds* the maximum normal input level as determined by a PPM or VU meter monitoring the input program line. Recall preset PROTECTION 5 dB if you want about 5 dB of limiting to occur at the maximum normal input level.

The Less/More control affects only the input drive, and you can use it to set a nominal limiting level different from 0 dB or 5 dB.

The Protection presets have the same Intermediate and Advanced Modify controls available as the Two-Band structure.

Setting Up the Two-Band Structure for Classical Music

To set up the Two-Band structure, recall preset CLASSICAL-2 BAND or CLASSICAL-2B+AGC.

These are the only two-band presets (other than the PROTECTION presets).

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 preset is a two-band preset with the AGC turned off. It uses considerable bass coupling to preserve the spectral balance of the input as well as possible. Its Less/More control primarily affects the amount of compression, rather than maximum loudness. It sounds essentially identical to the Protection structure.

CLASSICAL-2B+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 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 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 CLASSICAL-2 BAND preset) 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 two five-band classical presets. The CLASSICAL-5 BAND preset uses the five-band structure with AGC defeated. It uses substantial inter-band 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, with additional Two-Band controls available in the Advanced Modify screen.

Two-Band Controls		
Intermediate Name	**Advanced Name**	**Range**
Drive	2B Drive	0 … 25 dB
Release	2B Release	0.5 … 20 dB/S
Gate Thresh	Gate Thresh	Off, –44 … –15 dB
Bass Coupling	Bass Coupl	0 … 100 %
---	Lookahead	0 … 5 milliseconds
---	Master Compression Thresh	–15 … 0
---	Bass Thresh	–12.0 … 2.5 dB
---	Master Attack	4 … 50, Off
---	Bass Attack	4 … 50, Off
---	2B Clipping	–4 … +5
---	HF Limiting	–4.0 … +2.0
---	HF Clip Thresh	–16.0 … –0.2, Off
---	2B Crossover	LiNoDly, Linear

Table 3-6: Two-Band Controls

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 0-25 dB.

Regardless of the release time setting, we feel that the optimal 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 or the window gate.

It 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. In general, you should use faster release times for mass-appeal pop or rock formats oriented toward younger audiences, and slower release times for more conservative, adult-oriented formats (particularly if women are an important part of your target audience).

The action of the 2B Release control has been optimized for resolution and adjustability. But 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 amount of gain reduction desired. Decide based on listening tests how much gain reduction gives you the density that you want without 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 8400. 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 the compressor gate is 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 quieter and 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 optimal setting depends on the amount of gain reduction 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 almost always 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.)

2B Crossover sets the structure of the 2-Band crossover to Linear or Linear with No Delay. See page 3-35 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 equally affects the bass and master bands. 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.

MB Limit Threshold 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.

2B HF Limiting 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).

HF Clip Threshold 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.

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.

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 the overall flavor and quality of the sound.

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

Each of these presets can be edited with the Less/More control to optimize the trade-off between loudness and distortion according to the needs of the format. They can be further edited with Basic, Intermediate or Advanced Modify to fine-tune them.

The controls in the Five-Band structure give you the flexibility to customize your station sound. Nevertheless, as with any audio processing system, proper adjustment of these controls requires proper balancing of 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 summarize 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 in Section 3.

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
Gate Threshold	Gate Thres	Off, –44 ... –15 dB
Clipping	MB Clipping	–4.0 ... +5.0 dB
Down Expander	Down Expand	Off, –6.0 … 12.0 dB
Multiband Clipping	Multiband Clipping	–4.0 ... +5.0
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
---	MB Limit Thr	–3.0 +6.0, Off
---	Max Dist Ctrl	0 … 18 dB
---	B5 Clip Thre	–16.00 … –1.3, Off
---	HF Limiter	Off, –23.8 ... 0.0 dB
---	DownExpStCpl	Off, On
---	B1/B2 Xover	100. 200 Hz
---	Speech Thr	–4.0 ... +5.0 dB

Table 3-7: Multiband Controls

MB Attack/Release		
Intermediate Name	**Advanced Name**	**Range**
Release	MB Release	Slow, Slow2, Med, Med2, MFast, MFast2, Fast

---	B1 Attack	4 … 39 k, Off
---	B2 Attack	4 … 39 k, Off
---	B3 Attack	4 … 39 k, Off
---	B4 Attack	4 … 39 k, 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
---	Delta Release 1	–6 … 6
---	Delta Release 2	–6 … 6
---	Delta Release 3	–6 … 6
---	Delta Release 4/5	–6 … 6

Table 3-8: MB Attack/Release Controls

Band Mix		
Intermediate Name	Advanced Name	Range
Band Mix 1	B1 Out Mix	–3.0 … +3.0
Band Mix 2	B2 Out Mix	–3.0 … +3.0
Band Mix 3	B3 Out Mix	–3.0 … +3.0
Band Mix 4	B4 Out Mix	–3.0 … +3.0
Band Mix 5	B5 Out Mix	–3.0 … +3.0
Solo 1	Solo 1	Solo On, Solo Off
Solo 2	Solo 2	Solo On, Solo Off
Solo 3	Solo 3	Solo On, Solo Off
Solo 4	Solo 4	Solo On, Solo Off
Solo 5	Solo 5	Solo On, Solo Off
---	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 3-9: MB Band Mix 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 control scarcely affects density. Instead, the primary danger is that the excessive drive will cause noise to be excessively increased 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 release time of the multiband compressor is set to its faster settings, the setting of the MB Drive control becomes much more critical to sound quality because density increases as the control is turned up. Listen carefully as you adjust it. With these fast release times, there is a point beyond which increasing multiband compressor drive will no longer yield more loudness, and will simply degrade the punch and definition of the sound.

We recommend no more than 10 dB gain reduction as shown on the meters for band 3. More than 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. This is what we did in the factory Less/More presets for the Fast multiband release time.

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). And for background music formats, these settings ensure that your sound doesn’t lose its highs and lows. Because it creates a more consistent frequency balance between different pieces of source material than does the Two-Band structure, Slow is almost always preferable to the Two-Band structure for any popular music format.

The **Medium Slow** settings (Med and Med2) are appropriate for more adult-oriented formats that need a glossy show-business sound, yet whose ratings depend on maintaining a longer time spent listening than do conventional Contemporary Hit Radio (CHR) formats. With the single-ended noise reduction activated, it is also appropriate for Talk and News formats. This is the sound texture for the station that values a clean, easy-to-listen-to sound with a tasteful amount of punch, presence, and brightness added when appropriate. This is an unprocessed sound that sounds just right on music and voice when listened to on small table radios, car radios, portables, or home hi-fi systems.

The **Medium Fast** settings (MFast and MFast2) are ideal for a highly competitive Contemporary Hit Radio (CHR) format whose ratings depend on attracting a large number of listeners (high "cume") but which does not assume that a listener will listen to the station for hours at a time. This is the major market competitive sound, emphasizing loudness as well as clean audio. The sound from cut to cut and announcer to announcer is remarkably consistent as the texture of music is noticeably altered to a standard. Bass has an ever-present punch, there is always a sense of presence, and highs are in perfect balance to the mids, no matter what was on the original recording.

The **Fast** setting is used 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 than for 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.

There is nothing written in stone saying that you can't experiment with this sound for music-oriented programming as well. However, even with these settings, your sound is getting farther away from the balance and texture of the input. We think that this is as far as processing can go without causing unacceptable listener fatigue. However, this sound may be quite useful for stations that are ordinarily heard very softly in the background because it improves intelligibility under these quiet listening conditions. Stations that are ordinarily played louder will probably prefer one of the slower release times, where the multiband compressor takes more gain reduction and where the AGC is operated slowly for gentle gain riding only. These slower sounds are less consistent than those produced by the Fast setting. Using Slow preserves more of the source's frequency balance, making the sound less dense and fatiguing when the radio is played loudly.

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 8400. 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 quickly move 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. Activate the single-ended dynamic noise reduction by setting the MB Down Expander control to a setting other than Off.

The single-ended noise reduction system combines a broadband downward expander with a program-dependent low-pass filter. These functions are achieved by 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 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 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.

> This control was first introduced with Version 2.0 software.

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 ±3 dB range.

> Their range has been purposely limited because the only gain control element after these controls is the back-end clipping system (including the mul-

tiband 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 8400 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.

Solo 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 8400 presets.

Please note that a single band will interact with the back-end clipping system quite differently than will that band when combined with all of the other bands. Therefore, don't 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 controls set the compression threshold in each band, 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 strongly affect the amount of distortion produced by the back-end clipping system.

B1-B4 Attack (Time) controls set the speed with which the gain reduction in each band responds to level changes at the input to a given band's compressor. These controls, which have never previously been available in an Orban processor, 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.

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

The look-ahead delay time in bands 3, 4, and 5 automatically tracks the setting of the Attack time controls to minimize overshoot for any attack time setting.

MB Limit Thr sets the threshold of the clipping distortion controller with reference to the threshold of the final clipper, in dB.

The most effective setting for this control is "0 dB" for almost all program material. However, the loudest and most intense-sounding 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 artifact has been greatly reduced in version 1.0 software and above by comparison to v. 0.9x software.)

Max Dist Ctrl limits the maximum amount of final clipper drive reduction (in dB) that the 8400's clipping distortion controller can apply, 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 was first introduced with Version 2.0 software.

B5 Clip Threshold 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.

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.

<u>MB DownExpStCpl (Multiband Downward Expander Stereo Coupling)</u> determines whether the multiband downward expander stereo coupler is on or off.

<u>B1/B2 Xover</u> (Band 1 to Band 2 Crossover Frequency) sets the crossover frequency between bands 1 and 2 to either 100 Hz (as it was in all previous versions of the 8400 software) 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.

<u>Band 1-5 MaxDeltGR</u> controls set the maximum permitted gain difference between the left and right channels for each band in the multiband limiter. The 8400 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.

<u>Limiter Attack</u> 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.

<u>Delta Release</u> 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 GR 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 Gr control to 0. Thus, your settings automatically track any changes you make in the Multiband Release control. In our example, the release time in band 3 will always be two "click stops" slower than the setting of the 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.

<u>Lookahead</u> 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 8400'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 side effects.

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.

MB Speech Threshold ("Speech Thr," located in the Multiband 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. 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.

ITU-R Multiplex Power Controller

The ITU-R recommends that the power in the composite baseband signal (including the pilot tone), integrated over any 60-second interval, not exceed the power in a sinewave that modulates the FM carrier to ±19 kHz (25.3% modulation). Many European countries are now enforcing this recommendation. (See **ITU-R 412 Compliance** on page 3-11 for more information.)

Multiplex Power Threshold: The 8400 provides a means to limit the integrated multiplex power to the ITU standard by a closed-loop technique (patent pending) that allows you to use any preset and to create customized presets freely. The multiplex power controller is adjusted in the Input/Output: Utilities screen by the Multiplex Power Threshold control. Set it Off if your country does not enforce the standard.

> The control is located in the Input/Output: Utilities screen because the regulation applies to 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 8400. Setting the control at "0" will correctly control the multiplex power when there is no overshoot after the 8400. This will typically be true when you are using the 8400's built-in multiplex encoder to drive the transmitter directly.

Section 1 of this manual has an extensive discussion of overshoot in transmission paths. See page 1-13 and following pages.

However, many paths have overshoot, and this forces you to reduce the average modulation to avoid over-deviating 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 to the peak overshoot of the transmission system following the 8400. For example, if RF peak deviation exceeds the peak deviation produced by the 8400's sinewave oscillator (set for 100% modulation) by 3 dB, set the Multiplex Power Threshold to "+3."

8400 Processing and the Multiplex Power Threshold Control: The multiplex power controller reduces multiplex power by increasing the gain reduction in the multiband compressor, thereby reducing the drive into the clippers. Typically, the attack time of the controller is about 2 seconds, and its release time is about 30 seconds.

With no power control, some of the louder 8400 presets can exceed the ITU standard by approximately 6 dB. This means that the multiplex power controller must increase the gain reduction in the multiband compressor by up to 6 dB, and this will vary according to the dynamics and spectral content of the input program material.

Unfortunately, the multiplex power specification does not specify psychoacoustic weighting to reflect the sensitivity of the ear as a function of frequency. It therefore does little to predict the psychoacoustic loudness of co-channel interference (its stated goal), and heavy bass in the program material can force perceptually unnecessary reduction of the average modulation. To minimize loudness loss with the power controller active, we therefore recommend avoiding excessive bass boost in the 8400 processing. You may also wish to set the threshold of the Band 1 compressor lower, minimizing any dynamic bass boost caused by the processing.

Further, the power controller sees a pre-emphasized signal. Therefore, boosting frequencies above 6 or 7 kHz can also produce unnecessary loudness reduction because these frequencies are significantly rolled off by the receiver's de-emphasis. This reduces their ability to contribute to the loudness heard by the listener.

Because of the lack of psychoacoustic weighting in the ITU specification, excessive action of the power controller is likely to be audible as loudness changes that are inexplicable and annoying to the listener, who is expecting a smooth, consistent presentation. We therefore recommend turning down the Less/More setting on a given preset until the multiplex power controller (as indicated by the "P" [power] meter on the Meters screen) is doing very little work. If the meter indicates more than one-quarter scale, you are likely to notice long-term loudness inconsistency. Similarly, if you create a custom preset for use in connection with the multiplex power controller, we recommend backing off the MB Clipping and MB Drive controls until the "P" meter reads one-quarter scale or less.

You must allow some time to make this observation because of the power controller's slow two-second-attack time and even slower 30-second release time. Allow at least a minute.

The multiplex power controller is operational with both the Two-Band and Five-Band processing structures. It is not operational in Test mode and will not prevent the 8400'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%.)

If you apply sinewaves below 100 Hz to the Five-Band structure and use a slow attack time and fast release time on the Band 1 compressor, the multiplex power controller may not be able to produce enough extra gain reduction in Band 1 to control power correctly. This is because this combination of settings causes Band 1 to lower its compression ratio, and if the output level of the Band 1 compressor is too large, then the peak level is controlled by the bass clipper instead of the Band 1 compressor. However, even if you can find some combination of test tones and custom processing settings that are insufficiently controlled, program material will always be controlled correctly.

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.

Table 3-10: Test Modes (below) 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 … +15	1
Bypass 30 Hz HPF	---	ON	Off, On	---
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; MONO	---
Pilot	---	ON	ON, OFF	---

Table 3-10: Test Modes

Getting the Bass Sound You Want

Probably the most frequently asked question we get regarding 8400 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 8400 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 lots of bass, it's going to 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, the distortion occurring when the bass waveforms push the midrange and high frequency material into the 8400's final clipper.

There is one psychoacoustic trick you can do to create more apparent bass while efficiently using modulation headroom. 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 8400, 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 flat-tops 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 also 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 like 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 8400 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 fixed tone controls that apply coloration equally to all program material going into the main dynamics processing section of the 8400. (They do not affect the AGC section, being located after it in the signal flow.) Accordingly, the five-band compressor in the 8400 will attempt to undo any coloration added in the equalizer setting and 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 8400, 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 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 is *not* reproduced by small clock and portable radios), while band 2 affects the mid-bass and lower midrange. Setting the crossover at 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.

To wrap it up: 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 8400 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 and truly understand what you are doing. This manual is there to help, and it's worthwhile to reserve some time with if you want to become an 8400 bass expert.

Using the 8400 PC Remote Control Software

Optimod-FM 8400 must be upgraded to version 1.0 or greater prior to using 8400PC.

CAUTION: We recommend that you do not change your 8400's pre-emphasis by using the PC Remote program. Only change pre-emphasis locally from the 8400's front panel. (Usually, you will only change pre-emphasis once: when you first install your 8400.)

Your 8400 PC Remote control software is easy to use because it looks like the 8400's front panel. The menus and other features are organized the same way as they are in the 8400. The principal extra feature is the ability to archive user presets on your computer's storage devices (hard drives, floppy drives, etc.) and to restore them as user presets in your 8400.

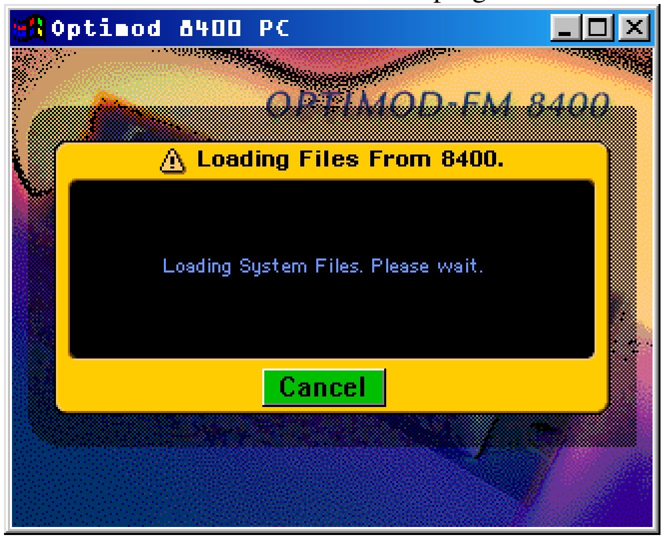
The 8400 PC Remote software can be connected to your 8400 via modem, direct serial cable connection, or Ethernet network. It communicates with your 8400 via the TCP/IP protocol, regardless of how it is connected to your 8400. (Networking requires a network card to be installed in the 8400's rear-panel PC Card slot. See "Networking" in Section 2 of this manual for instructions on how to set up 8400 networking.)

To start using 8400 PC Remote software, **first establish a connection from the client computer to the 8400** through a Windows Direct Cable Connection (Windows 2000 and XP), a RAS null modem connection (Windows 98), a Dial-Up Modem connection, or a TCP/IP Ethernet connection. If you are running version 2.1 software or higher, you must enter an "All Screens" passcode to initiate the network connection. When a network connection has been established, launch `8400_PC.exe` to start the software.

Before running 8400 PC Remote, you must have installed the appropriate Windows communications services on your computer. Because procedures and instructions for connecting to a PC are subject to development and change, we have placed these instructions in a file called `8400_Vxxx_installion.pdf` (where `xxx` represents the version number of the software). You can access this file from the Orban/Optimod 8400 folder in your computer's Start Menu after you have run Orban's PC Remote installer software or version 1.0 or greater of Orban's 8400 software update software. You can use Adobe's .pdf reader application to open and read this file. If you do not have the .pdf reader, it is available for free download from `www.adobe.com`.

By default, the installer installs a shortcut (labeled "Optimod 8400 PC Remote") to 8400_PC.exe on your desktop and in your Start Menu under Orban\Optimod 8400.

If you forget to establish a connection before launching `8400_PC.exe`, the program will time out after 45 seconds. Do not establish a connection until the program has finished timing out and has quit. Then, after establishing the connection, re-launch the program.



If you are running version 2.1 software or higher, you must enter an "All Screens" passcode to initiate the connection from the application to your 8400. A window will appear saying, "Connecting to the 8400, please wait." A few moments later, a new message should appear: "Loading System Files, Please wait."

When run, the Orban PC Remote software installer makes copies of all 8400 factory preset files on your local hard drive and the PC Remote software reads these files to speed up its initialization. If any of these files have been accidentally deleted or damaged, the PC Remote software will refresh them by downloading them from the 8400. 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.

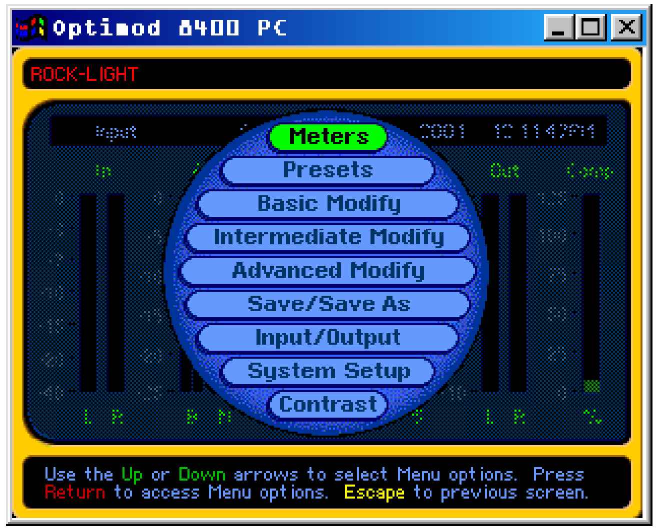
When this download is finished, the main meter screen will appear.

The 8400 PC Remote software is organized the same as your 8400's front panel. You can navigate through the 8400 PC Remote software via mouse or keyboard. 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.

The window in which the PC Remote software runs has a fixed size and resolution (measured in pixels) and cannot be resized (because it is based on bitmaps). The higher your screen resolution, the smaller this window will be.

To call up the main menu: Click the meters or help area (bottom of the screen).



To choose an item on the main menu: Double-click it.

To navigate to another screen at the same menu level (if any): Click the yellow left or right arrows at the top of the current screen.

To modify a control setting: Choose either Basic, Intermediate, or Advanced Modify from the main menu. To set a control, click it (it will become highlighted) and then use the wheel on the mouse to adjust it.

The Basic Modify screen is accessible after a short delay to recall the Less-More files from your local hard drive. If these are missing, the PC Remote software will refresh your local hard drive by downloading these files from the 8400. An informational screen appears while PC Remote downloads the Less-More tables from the 8400.

If you do not have a wheel mouse, you can use the + and – keys on the numerical keypad to adjust any control.

To recall a preset: Double-click the Preset screen from the main menu. Click the desired preset once. It will jump to the top of the screen. Click it again (when it is at the top of the screen) to put it on-air.

Continuously clicking on the preset showing at the top of the screen will toggle between the current and previous on-air presets.

To save a user preset you have created: Double-click the Save/Save As screen from the main menu. The current preset name will appear in a one-line window above the "keyboard" on the screen. Click on the line, and edit it with your computer keyboard as desired, using standard Windows techniques. Click on Save to save the preset to the 8400 as a User Preset.

To archive a user preset onto your computer's hard drive:

a) Double-click the Save/Save As screen from the main menu.

b) Click on Memory.

The Memory screen will appear.

c) Click on a preset in the 8400 column. Use the wheel on your mouse to bring the desired preset to the top of the column, where it will be highlighted in green.

d) Click Copy to copy the preset to your PCs' hard drive. Click Copy All to do a batch copy of all user presets in the 8400 list.

User presets are stored in the directory ~\user, where ~ is the path where you initially installed the PC Remote software. The default for ~ is: `c:\Program Files\Orban\Optimod 8400`.

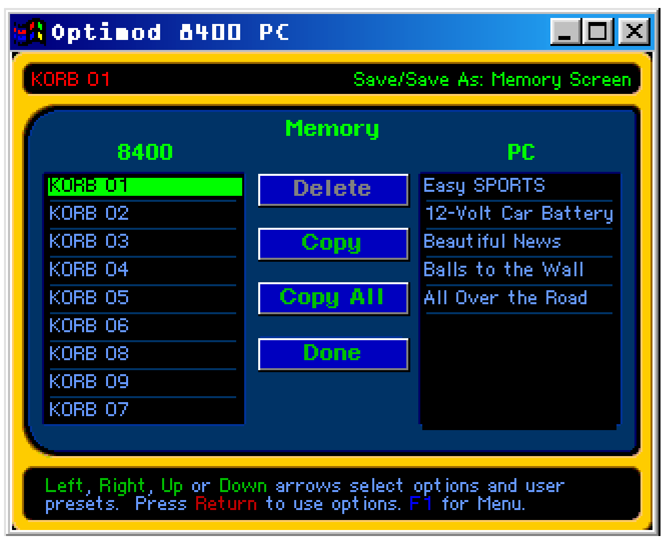
To restore an archived preset to your 8400:

Note that if you want to restore a user preset to your 8400, it must be in the ~ \ user directory (where ~ is the path to which you initially installed the PC Remote software) or the 8400 PC Remote software will be unable to find it. The default for ~ is: `c:\Program Files\Orban\Optimod 8400`.

a) Double-click the Save/Save As screen from the main menu.

b) Click on Memory.

The Memory screen will appear.



c) Click on a preset in the PC column. Use wheel on your mouse to bring the desired preset to the top of the column, where it will be highlighted in green.

d) Click Copy to copy the preset to your 8400's nonvolatile memory as a User Preset. Click Copy All to do a batch copy of all user presets in the PC list.

The Input/Output and System screens work as they do in the 8400.

The Contrast screen is not functional.

TEST modes: To prevent on-air accidents, in order to activate the Bypass or Tone test modes you must highlight the appropriate button and then hit ENTER on your computer's keyboard. You cannot enter a Test mode by using only the mouse.

You can access the test modes from the System Setup screen.

Navigation Using the Keyboard

Use the *ESC* key to bring up the meter screen, as in the 8400.

If you are in the meter screen, *ESC* brings up the main menu. So pressing *ESC* twice will bring up the main menu from almost anywhere.

Navigate around the screens using the left, right, up, and down arrow keys. To move to the next screen in the same menu, hold the left or right arrow key down for about a half second.

Use the + and – keys on the numeric keypad to adjust control settings.

Use the *ENTER* key to put highlighted presets on-air, and to “click” buttons that you have highlighted by navigating to them with the arrow keys.

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.

Section 4 Maintenance

Routine Maintenance

The 8400 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 8400's output.

A good ear will pick up many faults. Familiarize yourself with the "sound" of the 8400 as you have set it up, and be sensitive to changes or deterioration. But if problems arise, please don't jump to the conclusion that the 8400 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 8400 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 gets soiled.

Wash the front panel with a mild household detergent and a damp cloth. Stronger solvents should not be used because they may damage plastic parts, paint, or the silkscreened lettering (99% isopropyl alcohol can be safely used).

Removing and Replacing Parts and Assemblies

[See Section 6 of this manual for PC board and parts locator diagrams.]

1. Removing the Top Cover.

To access the main board, power supply board or display assembly, you must remove the top cover.

A) Disconnect the 8400 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 all eighteen screws holding the top cover in place and lift the top cover off.

Use a #1 Phillips screwdriver.

2. Removing the Power Supply assembly and cleaning the fan.

The entire power supply is a plug-in assembly. To remove the power supply it is unnecessary to remove the 8400 from the rack or to remove the top cover.

A) Be sure that the AC line cord is disconnected from the power supply.

B) Remove the seven #1 Phillips screws holding the power supply assembly to the rear of the chassis.

C) Using the handle on the back of the supply, pull it straight back. It will unplug from its socket.

The fan is part of the power supply assembly. Cleaning or replacing the fan only requires that the power supply be removed. As stated above, this can be done without removing the 8400 from the rack.

3. Removing the Input/Output Assembly

The input/output assembly is a plug-in assembly. It can be removed with the 8400 still mounted in a rack and with the 8400's top cover in place.

A) Make sure that AC power is disconnected from the 8400.

B) Remove the six Phillips screws holding the input/output assembly to the rear panel.

C) Pull the assembly directly back to unplug it from the motherboard.

You can use the BNC connectors and the tabs on the female XLR connectors to grasp the assembly.

4. Removing the DSP Board.

A) If you have not done so yet, remove the top cover (step 1 on page 4-2).

B) Be sure that power is off.

C) Disconnect the ribbon cable from J504.

D) Remove the five Phillips screws holding the DSP board to the bottom of the chassis.

E) Pull the board toward the front of the chassis to unplug it from the motherboard.

5. Removing the front and rear PC Card boards.

A) If you have not done so yet, remove the top cover (step 1 on page 4-2).

B) If you have not done so yet, remove the power supply (step 2 on page 4-3).

C) Remove the rear board.

a) Remove the two screws that hold the rear board to the vertical shield plate.

b) Lift the board up to remove it from its socket.

Gently rocking the board from side to side can aid in disengaging the multi-pin connector.

D) Remove the front board.

a) Remove the screw and nut holding the front board to the angle bracket that is part of the display shield assembly.

b) Lift the board up to remove it from its socket.

Gently rocking the board from side to side can aid in disengaging the multi-pin connector.

6. 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) 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.

B) Imagine a hinge on the bottom of the front panel, and tilt it toward you.

Do not stress the cables connecting the front panel to the main chassis.

There is an RFI-prevention contact spring between the front panel and the main chassis. Set this aside for later reinstallation.

C) Unplug all four cables connecting the front panel to the main chassis, noting where they are connected so you can reconnect them later.

To protect the assembly from cosmetic damage, set it down on a soft surface like foam rubber, a quilt, or a blanket.

7. Removing the Headphone Amplifier Board

Because it is socketed, you can remove and replace the headphone amplifier driver chip without further disassembly.

If you need to remove the headphone amplifier circuit board (to access components other than the headphone amplifier driver chip), pull the friction-fit knob off the headphone volume control. Remove one screw and one threaded standoff. This will free the board.

8. Preparing to Remove the Rotary Encoder Board, Color LCD Display, or Power LED

The color LCD display the power LED, and the circuit board containing the pushbuttons, joystick, and rotary encoder are all mounted on a metal shield plate. To remove it, remove six screws and lift the plate off at a 45-degree angle, following the axis of the rotary encoder.

9. Removing the Color LCD Display

Remove the four screws that hold the display to the standoffs on which it is mounted. Then lift the assembly off the standoffs.

Two of the screw heads are located under the foam tape that surrounds the display. You can find these by feeling them through the foam. Carefully peel the tape up next to the screw heads to access them.

10. Removing the Power LED Board

Remove the two screws that hold the display to the standoffs on which it is mounted. Then lift the assembly off the standoffs.

11. 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.

12. Removing the Display Board.

You must remove the front panel according to the instructions in step 6 on page 4-4 before you can remove the display board.

A) Unplug from the display board all of the cables leading from the various assemblies in the front panel.

B) Remove the two screws holding the display board to the chassis.

C) Carefully unplug the display board from its socket on the main board.

13. Removing the Main Board.

A) If you have not done so yet, remove the top cover (step 1 on page 4-2).

B) If you have not done so yet, remove the power supply (step 2 on page 4-3).

C) If you have not done so yet, remove the front and rear PC-card boards (step 5 on page 4-4).

D) If you have not done so yet, remove the display board (step 12 on page 4-5).

E) Unscrew the six threaded standoffs associated with the Serial 1, Serial 2, and Remote Interface connectors.

F) Unplug the three plugs connecting wiring harnesses and ribbon cables to the main board.

G) Remove the eight screws fastening the main board to the standoffs on the chassis.

H) Lift the main board out of the chassis.

14. Replacing the Main Board

A) Carefully insert the main board in the chassis, lining up the eight screw holes with the standoffs on the chassis.

Be sure that the rear-panel connectors fully protrude through their holes.

B) Replace the eight screws removed in step (13.G) on page 4-6.

C) Replace the six threaded stand-offs to the D-SUB connectors that were removed in step (13.E) on page 4-6.

D) Replace the display board.

a) Carefully plug the display board into its socket in the main board.

Be sure that the pins are lined up horizontally and that no pins have been bent.

b) Replace the two screws that hold the display board to the chassis standoffs.

E) If you have not done so, reassemble the front panel according to step 15 on page 4-6.

Be sure that you have plugged into the display board all of the cables that you unplugged when you disassembled the front panel.

F) Replace the PC Card boards.

a) Carefully plug the rear PC Card board into its socket on the main board.

b) Replace the two screws that hold the rear board to the vertical shield plate.

c) Carefully plug the front PC Card board into its socket on the main board.

d) Replace the screw and nut holding the front board to the angle bracket that is part of the display shield assembly.

G) Replace the plug connecting a wiring harness and the two plugs connecting ribbon cables to the main board.

H) Replace the power supply by plugging it in from the rear of the chassis and fastening it with six screws.

15. Reassembling the Front Panel

A) Replace all knobs on the rotary encoder board.

B) Remount the rotary encoder board on its four standoffs with the screws removed in the previous step.

C) If necessary, remount the LED board on its two standoffs.

The notch in the board mates with one of the standoffs supporting the LCD display.

D) If necessary, remount the LCD display on its four standoffs.

The shield plate should now have three assemblies mounted on it: the rotary encoder board, the small LED board, and the LCD display.

E) Mount the shield plate on the main front panel assembly.

a) Move the plate along the axis of the rotary encoder (at a 45 degree angle to the main front panel assembly) so that the rotary encoder goes straight through its clearance hole.

b) Replace and tighten the six screws that fasten the shield plate to the front panel.

Before tightening the screws, be sure that the joystick and both pushbutton switches protrude through the main panel normally, and that there is no binding when you operate any control.

F) If necessary, replace the headphone amplifier board and fasten it to the front panel using one screw (in the hole closest to the display) and one threaded standoff (in the remaining hole). Position over this standoff the RFI contact spring you removed in step (6.B) on page 4-4.

G) Fit the front panel assembly into the front of the chassis, bottom first. Leave the panel tilted at an angle of approximately 45 degrees and reconnect the four front-panel cables.

H) Mate the front panel assembly to the chassis, and fasten it using the six screws you removed earlier.

16. Replacing the Input/Output Assembly

A) Carefully insert the input/output assembly through the back panel so that it plugs cleanly into the motherboard.

B) Replace the six Phillips screws holding the input/output assembly to the rear panel.

17. Replacing the Top Cover.

Place top on the unit and reattach the 18 Phillips screws.

Field Audit of Performance

Required Equipment:

- Ultra-low distortion sine-wave oscillator/THD analyzer/audio voltmeter

  (With verified residual distortion below 0.01%. Sound Technology 1710B; Audio Precision System One, 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 5M Hz 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 8400'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 8400 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 8400 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 8400'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 1 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 2 screen, Set controls as in the table below:

Out Level	+10.0 dBu
Output Feeds	Xmitter
Pre-Emph	Flat
Out Level	–2.8 dBFS
Samp Rate	32 kHz

Pre-Emph	Flat
Word Leng	20
Dither	Out

d) Navigate to the Input/Output 4 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 pre-emphasis (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	+12V unregulated
2	+15V or –15V
3	+5V or –5V
4	+5V Digital
5	+3.3V #1
6	+3.3V #2
7	+3.3V #3
8	+3.3V #4
9	+3.3V #5

Table 4-1: Decoder Chart for Power Supervisor

Measuring the power supply at the test points on the power supply regulator board is not practical with the power supply installed because the heat sink makes the test points inaccessible. Therefore, it is best to take the measurements on the backplane at Molex connector J7. When one faces the connector, the voltages can be found on the pins in the following pattern:

+5V	–5V	+15V	–15V
+5V digital	+12V unreg.	Digital ground	Analog ground

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 8400 software controls are set to their default settings. (Refer to page 4-9.)

B) Feed the 8400 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 bypass preset: Navigate to the Bypass button and press Enter.

Bypass defeats all compression, limiting, and program equalization but retains pre-emphasis.

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.

4. Check frequency response of Analog I/O.

A) Verify 8400 software controls are set to their default settings. (Refer to page 4-9.)

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 pre-emphasis. 75µs pre-emphasis will cause 17 dB of boost at 15 kHz.

E) Connect the audio analyzer to the 8400'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 8400 software controls are set to their default settings. (Refer to page 4-9.)

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 8400 read 100.

You will have to reduce the output level of the oscillator at higher frequencies to compensate for the pre-emphasis boost in the 8400.

G) Measure the THD+N at the frequency levels 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 8400.

6. Test Digital Sample Rate Converter (Receiver).

A) Verify 8400 software controls are set to their default settings. (Refer to page 4-9.)

B) Be sure you are still in BYPASS mode [see step (3.F)].

C) Navigate to Input/Output. On screen Input/Output 1, Set Input To: Digital.

D) Connect the digital source generator to the AES/EBU Digital Input XLR connector of the 8400.

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 and 48 kHz. Use 24-bit words.

G) Listen to the analog outputs of the 8400 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 8400.

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 2.

C) Press IO CALIB soft key to access I/O CALIB menu.

D) Connect an AES/EBU analyzer (like the Audio Precision System 2) to the 8400's AES/EBU digital output.

E) Change the Samp Rate to 32 kHz, 44.1 kHz and 48 kHz, and verify that the frequencies measured at the 8400's AES/EBU output follow the chart below within given tolerances:

Sample Rate	Tolerance (PPM)	Tolerance (Hz)
32 kHz	50 PPM	±1.60 Hz
44.1 kHz	100 PPM	±4.41 Hz
48 kHz	50 PPM	±2.40 Hz

F) Disconnect the digital source generator from the 8400.

8. Test the 8400's stereo encoder.

A) Connect an accurate stereo monitor like the Belar FMMS-1 ("Wizard") stereo demodulator to the 8400's COMPOSITE OUTPUT 1.

This is labeled OUTPUT and appears on a BNC connector on the 8400's rear panel.

NOTE: The recommended Belar monitor is the only instrument we have encountered that can accurately measure the performance of the 8400's stereo encoder. With most older-technology monitors, you will be measuring the performance of the monitor, not the 8400's encoder. (Of course, we have not evaluated every monitor on the market.)

B) Navigate to the 8400'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 3. Set the Output 1 Level to make the stereo monitor read 100% total modulation.

E) Navigate to the 8400'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 –70 dB, 50-15,000 Hz.

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. This is right into left stereo separation. It should not exceed –70 dB, 50-15,000 Hz.

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 8400'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 3. Adjust Pilot Level as necessary.

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 8400.

O) Measure pilot tone frequency: With the Mod. Level still set to 0.0%, connect a frequency counter to either of the 8400'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 8400 PC Remote Control Software* on page 2-56.

10. Return OPTIMOD-FM to service.

A) Remove the 600Ω 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 4-9.

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 8400, or in other parts of the system.

Headphones Don't Work

The headphone jack is connected to chassis ground, and won't work if the 8400's chassis ground is not connected to its circuit ground. If the headphones don't work, switch the rear-panel Ground Lift switch to Ground.

RFI, Hum, Clicks, or Buzzes

A grounding problem is likely. Review the information on grounding on page 2-11. The 8400 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 8400 and the exciter, a ground loop may inject noise into the exciter's composite input—especially if the exciter's input is unbalanced. The Orban CIT25 Composite Isolation Transformer can almost always cure this problem.

The AES/EBU 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.

Poor Peak Modulation Control

The 8400 ordinarily controls peak modulation to an accuracy of ±2%. This accuracy will be destroyed if the signal path following the 8400 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.

Digital STLs using lossy compression algorithms (including MPEG1 Layer 2, MPEG1 Layer 3, and 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.

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 8400's audio inputs is clean. Heavy processing can exaggerate even slightly distorted material, pushing it over the edge into unacceptability.

The subjective adjustments available to the user have enough range to cause audible distortion at their extreme settings. There are many controls that can cause distortion, including Multiband Clipping, Final Clip Drive, and Composite Clip Drive Setting the LESS-MORE control beyond "9" will cause audible distortion of some program material with all but the Classical and Protect presets. Further, the "Loud" family of presets can sometimes cause audible distortion with certain atypical program material.

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 system (using System Setup). If your peak program level exceeds the peak level you have specified on setup, the 8400's A/D converter will clip and distort. (See page 2-30).

If you are using the 8400'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 8400's AGC. In this case, reduce the setting of the AGC's MaxDeltaGR controls.

If you are using an external processor ahead of the 8400, be sure it is not clipping or otherwise causing problems.

Audible Noise on Air

(See also "RFI, Hums, Clicks, or Buzzes" on page 5-2.)

Excessive compression will always exaggerate noise in the source material.

The 8400 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 8400. The first affects the AGC and the second affects the Multiband Compressor. Each has its own threshold control.

In the Multiband structure, *dynamic single-ended noise reduction* can be used to reduce the level of the noise below the level at which it appears at the input. Both the compressor gate and dynamic single-ended noise reduction features are explained in Section 3 of this manual.

If you are using the 8400'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 8400 to digitize the input. (This ratio is better than 105 dB.) It is important to correctly specify the AI CLIP level in the System Setup Analog I/O screen to optimize the noise performance available from the analog-to-digital converter. You should specify the level as the highest peak level that will be presented to the 8400 under normal operation. If, in an attempt to build in a “safety factor” or increase headroom, you specify a higher level than this, every 1 dB of extra headroom that you gain will be accompanied by a 1 dB increase in the 8400's noise floor.

The 8400's AES/EBU input is capable of receiving words of up to 20 bits. A 20-bit word has a dynamic range of approximately 120 dB. The 8400'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 8400, the STL's noise level can severely limit the overall noise performance of the system because compression in the 8400 can exaggerate the STL noise. For example, the overload-to-noise ratio of a typical analog microwave STL may only be 70-75 dB. In this case, it is wise to use the Orban 8200ST Studio Chassis 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. Section 1 of this manual has a more detailed discussion.

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.

The 8400 uses independent composite DACs for the Composite 1 and Composite 2 outputs. Therefore, a problem of this nature would likely show up at only one composite output.

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

Interference From Stereo Into SCA

A properly operating 8400 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 8400 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 8400'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 8400's composite output and the transmitter has sufficient headroom.

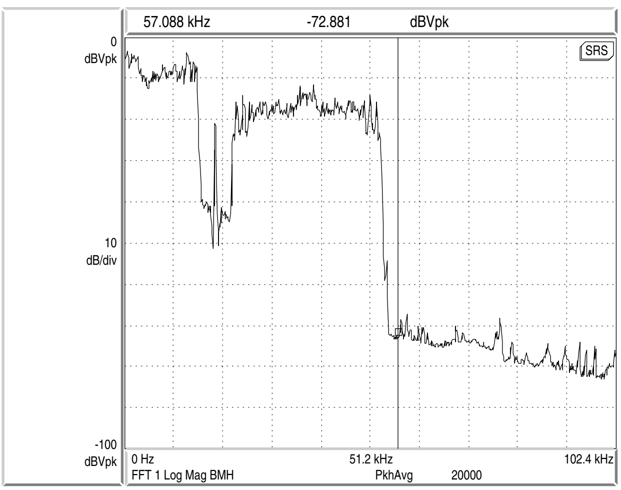


Fig. 5-1: Typical 8400 baseband spectrum with heavy processing, 0-100 kHz.

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 the Multi-Band structure, this problem can be caused by excessive HF boost in the HF Equalizer and HF Enhancer. It could also be caused by an excessively high setting of the BAND 4 THRESH control, or by excessively high settings of the BAND 4 MIX and BAND 5 MIX controls (located in Intermediate and Advanced Modify).

If you are driving an external stereo encoder with built-in pre-emphasis, you must set the 8400's output to FLAT in the System Setup/Output screen to prevent double pre-emphasis, which will cause very shrill sound (and very poor peak modulation control).

> You will *always* achieve better peak control by defeating the pre-emphasis and input filters of an external stereo encoder, permitting the 8400 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 8400's output is set to FLAT in System Setup/Output, there will be no pre-emphasis unless it is supplied somewhere else in the system. This will cause very dull sound.

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

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

The 8400 can generate test tones itself. The 8400 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 page 3-??.

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 8400, or the 8400 should be temporarily switched to bypass to pass the tones.

System Receiving 8400's Digital Output Will Not Lock

Be sure that the 8400's output sample rate is set match the sample rate that the driven system expects. Be sure that the 8400's output mode (AES/EBU 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 your monitoring service. If the problem is real, contact Orban Customer Service for a crystal replacement; there is no frequency trim available.

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

This problem is often caused by relative phase shifts between the left and right channels prior to the 8400'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.

General Dissatisfaction with Subjective Sound Quality

The 8400 is a complex processor that can be adjusted for many different tastes. For most users, the factory presets, as augmented by the gamut offered by the Less-More control for each preset, are sufficient to find a satisfactory "sound." However, some users will not be satisfied until they have accessed other Modify Processing controls and have adjusted the subjective setup controls in detail to their satisfaction. Such users *must* fully understand the material in Section 3 of this manual to achieve the best results from this exercise.

By comparison to competitive processors, the 8400 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 8400, 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.

Security Passcode Lost (When Unit is Locked Out)

Please see *If you have forgotten your "All Screens" passcode…* on page 2-45.

Connection Issues between the 8400 and a PC, Modem, or Network

- If you **cannot install a RAS null modem** because it does not appear on the list of available modems, the information file for the RAS null modem, `mdmorban.inf`, might be missing from the INF folder under your Windows or Winnt directory. Orban's installer automatically installs this file, but the file could have been accidentally deleted or damaged. Run Orban's installer, `Setup8400_3-0_update.exe`, again.

- To **uninstall the RAS null modem**, remove the file `mdmorban.inf`, from the `inf` folder under your `Windows` or `Winnt` directory.

- **Presets**: The more user presets you make, the more slowly the 8400 will respond to front-panel commands. Delete any user presets you don't need.

- **Quick Setup**: On the Station ID screen (Quick Setup 9): Use Escape in place of Cancel. The Cancel button will not work.

- **Software Updates**: Close any running Windows programs before attempting to update.

- **Foreign-language Updates**: We have heard reports (unconfirmed) that Windows' language setting may be a factor in determining the reliability of the update process. We advise setting regional language to English (Control Panel/Regional) if you have problems.

- **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 go into the Network screen and change the connection type while the modem is connected or attempting to connect.

- **Security Passcode**: If version 2.0 or below is installed on the 8400 being upgraded, the Security passcode also works for modem connections, protecting the 8400 from unauthorized connections. If you enter a passcode in the Security screen, the user will need to enter the same passcode in the Windows' Dial-up Connection procedure used when connecting the 8400 to a modem. If version 2.1 or above is installed on the 8400 being upgraded, an All Screen (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.

- **SERIAL 2 Port**: The SERIAL 2 port has two functions when version 2.1 software or higher is running in your 8400: it lets you use a simple hardware technique to create a new passcode (ADMIN) with all screens privileges, and it lets you connect your PC to your 8400 via a terminal program (like HyperTerminal) to administer the 8400. Use only the SERIAL 1 port for upgrading or for the PC Remote application.

- **Energy Saver**: The Energy Saver is hard-coded ON. (This protects the 8400's costly color LCD from premature failure.)

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 8400 in the "System Setup: Network" screen.

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 Serial 1 of the 8400.

If your Modem Connect does not work:

A) Ensure that the modem cables and phone lines are connected properly.

B) Ensure that the modem cable is connected to Serial 1 on the 8400.

C) Check that you have entered the correct phone number for connection.

D) Check that you have entered the passcode correctly on the 8400, and the passcode has also been entered correctly on your PC (they are case sensitive).

E) Ensure that you enabled the correct PC modem port settings.

F) Ensure that the external modem attached to your 8400 is set to Auto Answer.

G) Make sure that the only "Allowed Network Protocol" is TCP/IP. "NetBUI" and "IPX/SPX Compatible" must *not* be checked.

OS-Specific Troubleshooting Advice

Troubleshooting Windows 2000 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 and Dialup Connections” to bring up the Network Connections screen.

B) In the “Network Connections” window, right-click “Optimod 8400 - Direct” and choose “Properties.”

C) The “Properties” window opens for “Optimod 8400 - 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 “Components checked are used by this connection,” uncheck all except for “Internet Protocol (TCP/IP).”

H) 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) In the “Advanced TCP/IP Settings” select the “DNS” Tab.

M) In the “Advanced TCP/IP Settings” select the “WINS” Tab.

N) Click “OK” to dismiss the “Advanced TCP/IP Settings” window.

O) Click “OK” to dismiss the “Internet Protocol (TCP/IP) Properties” window.

P) Click “OK” to dismiss the window whose name is your new connection.

Q) Click “Cancel” to dismiss the “Connect [nnnn]” dialog box

R) Restart your computer. (This resets the serial port and reduces the likelihood that you will encounter problems connecting to the 8400.)

S) If you see: **“Error 777: The connection failed because the modem (or other connecting device) on the remote computer is out of order”:**

The “remote computer” is actually the 8400 and it is not out of order; you just need to set the Maximum Speed (Bits per second) to 115200. If you already set this speed when you configured your PC ports, you shouldn’t have this problem.

> **Note**: The 8400 communicates at 115200 bps. COM ports on some older PCs are incapable of communications at this rate and may not work reliably. Most newer PCs use 16550-compatible UARTS, which support the 115200 bps rate.

If you do see this warning message, you can reset the Maximum BPS Speed by accessing Properties for the connection:

a) Click Start / Programs / Accessories / Communications / Network and Dial-up Connections.

b) Right click the name of your connection and access "Properties."

c) Go to the "Generals" tab and select the "Configure" button.

d) Set the Maximum Speed (bps) to 115200.

e) Select OK and try your connection again.

T) If you see: **"Error 619: The specified port is not connected."**

Make sure the Interface Type on the 8400 is correct:

a) On the 8400, go to Menu and select System Setup.

b) In System Setup go to Network/Remote.

c) Set interface type to Direct.

d) Try your connection again.

Troubleshooting Windows 2000 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 and Dialup Connections" to bring up the Network Connections screen.

B) In the "Network Connections" window, right-click "Optimod 8400 - Modem" and choose "Properties."

C) The "Properties" window opens for "Optimod 8400 – Modem".

D) Click the "Properties" button.

E) Select the "General" tab and make sure that "Connect Using" displays the correct modem and port.

F) Click the "Configure…" button.

G) Set the "Maximum Speed (bps) to 115200.

H) Check the "Enable hardware flow control," make sure all other hardware features are unchecked. Then click "OK."

I) Click the "Networking" tab on the "Properties" window.

J) Set "Type of dial-up server I am calling" to "PPP: Windows 95/98/NT4/2000, Internet."

K) Select the "Settings" button and make sure all PPP settings are unchecked. Then click "OK."

L) In "Components checked are used by this connection," uncheck all except for "Internet Protocol (TCP/IP)."

M) Select "Internet Protocol (TCP/IP)" and then click the "Properties" button. The "Internet Protocol (TCP/IP) Properties" window opens.

N) Choose "Obtain an IP address automatically" and "Obtain DNS server address automatically"

O) Click the "Advanced…" button on the "Internet Protocol (TCP/IP)" Window.

P) In the "Advanced TCP/IP Settings" select the "General" Tab; make sure that no check boxes are checked.

Q) Click "OK" to dismiss the "Advanced TCP/IP Settings" window.

R) Click "OK" to dismiss the "Internet Protocol (TCP/IP) Properties" window.

S) Click "OK" to dismiss the window whose name is your new connection.

T) Click "Cancel" to dismiss the "Connect [nnnn]" dialog box

U) Restart your computer.

Although not strictly necessary, this resets the serial port and reduces the likelihood that you will encounter problems connecting to the 8400.

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 8400 - Direct" and choose "Properties."

C) The "Properties" window opens for "Optimod 8400 - 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 8400 – Modem" click the "Advanced" tab.

N) Make sure the "Internet Connection Firewall" setting "Protect my computer and network by limiting or preventing access to this computer from the Internet" is *unchecked*.

IMPORTANT: If you don't turn this feature OFF, then data from the 8400 does not get sent back to your computer. Thus, no meters appear, and the input type and station ID are missing.

O) Click "OK" to dismiss the window whose name is your new connection.

P) Click "Cancel" to dismiss the "Connect [nnnn]" dialog box

Q) Restart your computer.

This resets the serial port and reduces the likelihood that you will encounter problems connecting to the 8400.

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 8400 - Modem" and choose "Properties."

The "Properties" window opens for "Optimod 8400 - 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.

The "Internet Protocol (TCP/IP) Properties" window opens.

H) Choose "Obtain an IP address automatically" and "Obtain DNS server address automatically."

I) Click the “Advanced…” button on the “Internet Protocol (TCP/IP)” Window.

J) In the “Advanced TCP/IP Settings,” select the “General” Tab; make sure that no check boxes are checked.

K) Click “OK” to dismiss the “Advanced TCP/IP Settings” window.

L) On the “Properties” window for “Optimod 8400 – Modem,” click the “Advanced” tab. Make sure the “Internet Connection Firewall” setting “Protect my computer and network by limiting or preventing access to this computer from the Internet” is *unchecked*.

If you don’t turn this feature OFF, then data on the 8400 does not get sent back to your computer. Thus, no meters appear, and the input type and station ID are missing.

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 8400.)

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 8400'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. Be prepared to accurately describe the problem. Know the serial number of your 8400 — this is printed on the rear panel of the unit.

Telephone:	(1) 510/351-3500
Write:	Customer Service Orban 1525 Alvarado Street San Leandro, CA 94577 USA
Fax:	(1) 510/351-0500
E-Mail	custserv@orban.com

Please check Orban's website, www.orban.com, for Frequently Asked Questions and other technical tips about 8400 that we may post from time to time. Manuals (in .pdf form) and 8400 software upgrades will be posted there too—click on "Downloads" from the home page.

Factory Service

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

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

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

Shipping Instructions

Use the original packing material if it is available. If it is not, use a sturdy, double-walled carton no smaller than 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

Specifications apply for measurements from analog left/right input to stereo composite output and to FM analog left/right output.

Frequency Response (Bypass Mode): Follows standard 50µs or 75µs pre-emphasis curve ±0.10 dB, 2.0 Hz–15 kHz. Analog left/right output and Digital output can be user configured for flat or pre-emphasized output.

Noise: Output noise floor will depend upon how much gain the processor is set for (Limit Drive, AGC Drive, Two-Band Drive, and/or Multi-Band Drive), gating level, equalization, noise reduction, etc. 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: >65 dB, 20 Hz–15 kHz; 70 dB typical.

Polarity (Two-Band and Bypass Modes): Absolute polarity maintained. Positive-going signal on input will result in positive-going signal on output.

Installation

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.

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

Filtering: RFI filtered, with high-pass filter at 0.15 Hz.

Analog Audio Output

Configuration: Stereo. Flat or pre-emphasized (at 50µs or 75µs), software-selectable.

Source Impedance: 50Ω, electronically balanced and floating.

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

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

Signal-to-Noise: >= 90 dB unweighted (Bypass mode, de-emphasized, 20 Hz–15 kHz bandwidth, referenced to 100% modulation).

Crosstalk: <= –70 dB, 20 Hz–15 kHz.

Distortion: <= 0.01% THD (Bypass mode, de-emphasized) 20 Hz–15 kHz bandwidth.

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

D/A Conversion: 24 bit 128x oversampled.

Filtering: RFI filtered.

Digital Audio Input

Configuration: Stereo per AES/EBU standard, 24 bit resolution, software selection of stereo, mono from left, mono from right or mono from sum.

Sampling Rate: 32, 44.1 or 48 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 Output

Configuration: Stereo per AES/EBU standard. Output configured in software as flat, pre-emphasized to the chosen processing pre-emphasis (50µs or 75µs), with– or without J.17 pre-emphasis.

Sample Rate: Internal free running at 32, 44.1 or 48 kHz, selected in software. Can also be synced to the AES/EBU SYNC input or the AES/EBU digital input at 32, 44.1, or 48 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.

Digital Sync Input

Configuration: Used for synchronization of the Digital Output signal to an external reference provided at this input.

Sampling Rate: 32, 44.1 and 48 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.

Composite Baseband Output

Configuration: Two outputs, each with an independent software-controlled output level control, output amplifier and connector. Output 2 may be configured in software as a pilot frequency reference.

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).

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: <= 0.02% THD (Bypass mode, de-emphasized, 20 Hz – 15 kHz bandwidth, referenced to 100% modulation, unweighted).

Stereo Separation: At 100% modulation=3.5Vp-p, > 70 dB, 30 Hz – 15 kHz. At 100% modulation=1.0 – 8.0 Vp-p, > 60 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).

57 kHz (RDS/RBDS) Protection: –50 dB relative to 4% subcarrier injection, ±2.0 kHz (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).

Connectors: Two BNC, floating over chassis ground, EMI suppressed.

Maximum Load Capacitance: 0.047 microfarad (0Ω source impedance). Maximum cable length of 100 feet/30 meters RG–58A/U.

Filtering: RFI filtered.

Subcarrier (SCA) Inputs

Configuration: Subcarrier inputs sum into composite baseband outputs.

Impedance: 600Ω

Sensitivity: The gain from the subcarrier input to the composite output is fixed at –20 dB. Therefore, the gain is scaled so that 1.5V peak at the subcarrier input produces 10% subcarrier injection with reference to 3.0Vp-p=100% modulation of the FM carrier.

Connectors: Two BNC, unbalanced and floating over chassis ground, EMI suppressed.

HD (High Definition Digital Radio) Option

Configuration: Two field-installable cards: DSP and Input/Output. The DSP 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 outputs are always simultaneously active.

Frequency Response: Frequency response from input to HD output is ±0.10 dB, 2.0 Hz–15 kHz; there is no pre-emphasis.

Sync Input: Used for synchronization of the HD Digital Output signal to an external reference provided at this input. Other specifications as in **Digital Sync Input** on page 6-3.

Digital Output: Stereo per AES/EBU standard. Designed to feed digital transmission paths without pre-emphasis; output is flat. Other specifications as in **Digital Audio Output** on page 6-3.

Analog Output: L and R. Can supply output of HD look-ahead limiter, input to HD look-ahead limiter, or low-delay monitor signal. Output is flat. Other specifications as in **Analog Audio Output** on page 6-3.

Relative Time Delay between FM and HD Outputs: HD output is delayed by 10.500 ms with respect to the FM AES/EBU digital output under all operating conditions (excluding BYPASS and TEST modes)..

Remote Computer Interface

Configuration: TCP/IP protocol via direct cable connect, modem, or Ethernet interface. Suitable null modem cable for direct connect is supplied. Modem, Ethernet PC Card, and other external equipment is not supplied.

Connectors: RS–232 port (2) dB–9 male, EMI-suppressed. Ethernet or Modem card supported with rear-panel PC Card slot.

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.

Power

Voltage: 100–132 VAC or 200–264 VAC, switch-selected on the rear panel, 50–60 Hz, 50 VA.

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.

Environmental

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

Humidity: 0–95% RH, non-condensing.

Dimensions (W x H x D): 19” x 5.25” x 16.5” / 48.3 cm x 8.9 cm x 41.9 cm. Depth shown indicates rack penetration; overall front-to-back depth 18.75” / 47.6 cm. Three rack units high.

Humidity: 0–95% RH, non-condensing.

RFI / EMI: Tested according to Cenelec procedures.

Shipping Weight: 40 lbs. / 18.1 kg

Warranty

One Year, Parts and Labor: Subject to the limitations set forth in Orban's Standard Warranty Agreement.

Specifications are subject to change without notice.

Circuit Description

This section provides a detailed description of user-serviceable circuits used in the 8400. 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 8400 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.

Figure 6-1 on page 6-22 shows circuit board locations.

Overview

The Control Circuits control the DSP, display, and input/output sections of the 8400 system.

The Input Circuits include the connectors and RF filtering for the analog and digital audio inputs, the digital sync input, and the circuitry to interface these inputs to the digital processing. The optional HD inputs use circuitry identical to the main Digital and Sync inputs. The PC board layout for the HD I/O module is also identical to that of the main FM I/O module. The only difference is that some parts are not stuffed on the HD board.

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 optional HD outputs use identical circuitry to the main FM output circuitry except that there is no composite output.

The DSP Circuits implement the bypass, test tone, and audio processing using digital signal processing.

The Power Supply provides power for all 8400 circuit sections.

A block diagram of the DSP signal processing appears on page 6-44.

Control Circuits

The control circuit is based on an Elan SC400 microprocessor, which is a 486-class processor running an Orban executable program over a third-party real-time operating system. A disk-on-chip memory emulates a hard drive. The memory is non-volatile and does not rely on battery back-up to retain information when mains power is off.

The disk-on-chip holds the operating system, the Orban executable program, and all preset files, both factory and user. There is also a boot ROM that allows the microprocessor to start up before control is handed to the operating system.

The control circuits process and execute user-initiated requests to the system. The source of these requests is the front panel buttons, joystick, and rotary encoder, the rear panel RS-232 ports, and the remote contact closures. These changes affect hardware function and/or DSP processing. The control circuits also send information to the quarter-VGA passive-matrix color LCD display.

The control circuit contains a real-time crystal-controlled clock to keep time for the 8400's automation functions. It is backed up by a DL2032 battery so that it keeps accurate time even when the 8400 is powered down.

The control circuit communicates with the DSP and display circuitry through the Elan's ISA bus.

The Elan 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 Elan. The DSP chips will continue to process audio until the time comes to reload DSP program code into them. At this point, the audio will mute for about 30 seconds until the DSP code download has been completed. If you hear a 30-second audio mute on air, you can assume that the 8400 has rebooted for some reason. Be prepared to convey this fact to Orban customer service if you call for technical assistance.

User Control Interface and LCD Display Circuits

The user control interface enables the user to control the functionality of the 8400 unit. A rear panel GPI connector implements optically isolated remote control of certain functions, such as recalling presets. Two RS232 serial ports allow you to connect a modem or computer to the 8400. Front panel pushbutton switches and a joystick select between various operational modes and functions.

1. Remote Interface and RS-232 Interfaces

Located on control board

A remote interface connector and circuitry implements remote control of certain operating modes; Model 8400 OPTIMOD-FM 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 8400 will respond to the most recent control operation 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, U28, 30, and then to the control circuitry.

The RS232 ports are buffered by octal drivers U14, 17.

U28, 30 and U14, 17 are socketed for easy field replacement in the event of overload. All other circuitry is surface-mount and is not field-repairable.

2. Color LCD Display

The color LCD is a passive-matrix quarter-VGA panel. The CPU addresses it by memory mapping.

The backlight on the display has a finite lifetime (ordinarily a few years of continuous operation). Therefore, the 8400 always implements a screen-saver timeout. You will maximize the life of this costly component by setting the screen-saver timeout as short as practical.

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 AES/EBU-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 that has an adjustable (digitally controlled) gain. Analog switches set the gain. The outputs of a latch set the state of the switches. By writing data to the latch, the control circuits set the gain according to what the user specifies from the front panel controls. The gain 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, R100-109, C100-103, CR100, and CR103. (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.) The state of the switches is set by the outputs of digital latch IC108.

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. 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, IC107.

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. (An Orban-designed synchronous sample rate converter in the 8400's DSP performs the final decimation to 32 kHz. This ensures the flattest frequency response to 15 kHz without aliasing.)

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 AES/EBU 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 rates in addition to any digital audio sample rate within the range of 25 kHz and 55 kHz. The SRC converts the input sample rate to 64 kHz, with the final, high-quality decimation to the 8400 system sample rate being done in the system DSP, as was done for the analog input.

A second receiver/converter, IC502, accepts a second digital audio signal to enable the 8400's output sample rate to be locked to a reference sample rate presented to J501.

These chips are surface-mounted and are 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 AES/EBU-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:

The left channel signal emerging from IC211 is filtered by IC201 and associated components. 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 possibly 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 AES/EBU line driver chip, IC502, converts the 32 kHz 8400 system sample rate to any of the standard 32 kHz, 44.1 kHz or 48 kHz rates, and also contains a digital audio interface transmitter to encode digital audio signals using the AES/EBU interface format (AES3-1992). This chip is surface-mounted and is not field-replaceable.

4. Composite Output Circuit

Located on the input/output board

Both composite output circuits are entirely independent, including D/A converters. This permits the composite level to be set independently for each output in DSP.

Component-Level Description:

We will describe composite output #2. 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, R328. (This filter is equalized and phase-corrected in DSP to obtain excellent flatness and phase-linearity. This achieves very high stereo separation.)

IC302 buffers the output of the anti-imaging filter and sends its output to the small I/O Composite Board on which the two composite output BNC connectors are mounted, along with line driver amplifiers. These are composite amplifiers consisting of opamps IC3A and IC3B, followed by high-current buffers IC1 and IC2, which are inside the feedback loops of IC3A and IC3B.

The two SCA inputs are summed into IC3B and IC3A. These therefore are not indicated on the Composite Level meter displayed by the 8400, because this meter indicates only the composite signal generated by the DSP.

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 eight Motorola DSP56362 24-bit fixed-point DSP chips that execute DSP software code to implement digital signal processing algorithms.

The algorithms filter, compress, and limit the audio signal. The eight DSP chips, each operating at approximately 100 million instructions per second (MIPS), for a total of 800MIPS, provide the necessary signal processing. A sampling rate of 32 kHz and power-of-two multiples thereof, up to 512 kHz, is used.

System initialization normally occurs when power is first applied to the 8400 and can occur abnormally if the 8400's watchdog timer forces the Elan to reboot. Upon initialization, the Elan CPU downloads the DSP executable code stored in the disk-on-chip memory. This typically takes about 40 seconds. Once a DSP chip begins executing its program, execution is continuous. The Elan 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 Elan 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, which is identifiable by the small toroidal inductor, L700, which is mounted on it—see *Figure 6-1* on page 6-22. This board contains a local switching regulator that reduces the "raw" 10 volts from the power supply module to 3.3V for the DSP chips. Because of the danger of damaging the DSP chips, we do not recommend that this regulator be repaired in the field.

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 8400. 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 powers the fan and also feeds local regulators.

The power supply circuits are straightforward and no explanation is required beyond the schematic itself. Be aware that C19, C50, C52, and C53 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 8400. See step (2.B) on page 4-10.

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
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 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
HF	high-frequency
HP	high-pass
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/P-DIF	Sony/Philips digital interface
TRS	tip-ring-sleeve (2-circuit phone jack)
THD	total harmonic distortion
TX	transmitter
μs	Microseconds. For FM pre-emphasis, the +3 dB frequency is $1/(2\pi\tau)$, where τ is the pre-emphasis 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 8400 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 many of the parts used in the 8400 (including surface-mount devices), but inclusion of a part in this list does not imply that the part is field-replaceable. We have endeavored to include all field-replaceable parts in this list. However, some non-field-replaceable parts have been omitted.

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 may be contacted through their web sites.

Power Supply

PART #	DESCRIPTION	COMPONENT IDENTIFIER
21123.410.01	CAPACITOR, RADIAL LEADS, 0.1UF, 50V, 20%	C10 C11 C12 C4 C5 C6 C7 C8 C9
21209.822.01	CAPACITOR, RADIAL LEADS, 2200uF, 50/63V	C1 C2 C3
21227.710.01	CAPACITOR, RADIAL LEADS, 100uF 16V HFS	C50
21227.747.01	CAPACITOR, RADIAL LEADS, 470uF 16V HFS	C52 C53
21229.610.01	CAPACITOR, RADIAL LEADS, 10uF 35V HFS	C13 C14 C15 C16
21230.710.01	CAPACITOR, RADIAL LEADS, 100uF 50V HFS	C19
21256.000.01	CAPACITOR, RADIAL LEADS, 1000uF, 35V, 20%	C17
21258.910.01	CAPACITOR, ALUMINUM ELECTROLYTIC 10,000uF, 25V	C18
22083.022.01	DIODE, SURGE SUPPRESSOR, 22 VOLT	CR10 CR11 CR12 CR13 CR14
22083.033.01	DIODE, SURGE SUPPRESSOR, 33 VOLT	CR6 CR7
22083.068.01	DIODE, SURGE SUPPRESSOR, 6.8 VOLT	CR15 CR16 D1
22208.040.01	DIODE, SCHOTTKY, 31DQ04, 3.3	D2
22303.000.01	DIODE, RECTIFIER, 100V, 4 AMP	CR8 CR9
22305.000.01	DIODE, BRIDGE GBPC2506W	CR1
22500.271.01	ZENER, TRANSSORB, VARISTOR	V1 V2
24323.000.01	IC, SIMPLE SWITCH, 0 TO 220	U38
26143.000.01	SW, SLIDE, VOLT, 115/230	S2
26146.000.01	SW, SLIDE, SPDT, VERT MOUNT	
28112.005.01	BODY, FUSEHOLDER, PC MNT	FH1
29262.000.01	LINE FILTER, PC MNT, 1A	J1
29504.150.01	INDUCTOR, 2A, PE53113	L45

29527.000.01	INDUCTOR, FIT44-4	L46
15025.000.01	TRANSTOR, MTG, KIT TO 220	
15028.003.01	INSULATOR, 8400 P/S	
24308.901.01	IC, LINEAR, DC REG, 5V NEG	U4
24319.000.01	IC, DC NEGATIVE VOLTAGE REGULATOR	U2
24325.005.01	IC, VOLT REGULATOR, "T", POS, 5 VOLT	U3
24325.015.01	IC, VOLT REGULATOR, "T", POS, 15 VOLT	U1
27496.000.01	CONNECTOR, RCPT, 3 WIRE, CLOSED END	
28004.150.01	FUSE, 3AG, SLOBLO, 1/2 AMP	
28112.003.01	KNOB, FUSE, DOM, GRY, FOR 281	
29265.000.01	FAN, 12V	
29267.000.01	FILTER FAN GUARD 80MM	
51001.000.01	HEAT SINK INSULATOR, 8400	
57129.000.01	BRACKET, POWER SUPPLY FAN	
57130.000.01	BRACKET, POWER SUPPLY HEATSINK	
57131.000.01	HEATSINK	
57128.000.02	PANEL, FAN	
55041.000.02	TRANSFORMER TOROID 8400 115/230	T1

Input/Output Circuit Board

PART #	DESCRIPTION	COMPONENT IDENTIFIER
27053.001.01	CONNECTOR, XLR, HOUSING, MALE	J201 J202 J502
27054.001.01	CONNECTOR, XLR, FEMALE, HSNG	J100 J103 J500 J501
20041.100.01	RESISTOR, MF, 1/8W, 1%, 1.00 K ohm	R100 R107 R115 R120
20058.187.01	RESISTOR, MF, 1/8W, 0.1%, 1.87K	R301 R316
20058.205.01	RESISTOR, 1/8W.1%, 2.05K	R302 R319
20121.100.01	RESISTOR, RF, 1/8W, 1%, 10 ohm	R154 R200 R232 R522
20122.110.01	RESISTOR, TF, 1/8W, 1%, 110 ohm	R500 R514 R517 R238 R330
20123.100.01	RESISTOR, TF, 1/8W, 1%, 1k	R304 R317 R521 R600 R601
20123.150.01	RESISTOR, TF, 1.8W, 1%, SURFACE MOUNT 1	R131 R134 R140 R141 R144 R146
20123.499.01	RESISTOR, TF, 1/8W, 1%, 4.99K	R101 R103 R105 R108 R116 R118 R121 R124
20124.100.01	RESISTOR, TF, 1/8W, 1%, SURFACE MOUNT 1	R110 R125 R237 R519 R243 R244
20126.100.01	RESISTOR, TF, 1/8W, 1%, 1.00M	R142 R152 R225 R231 R306 R322
20129.150.01	RESISTOR, 1/8W, 1%, 150 ohm	R138 R151 R235 R236
20129.249.01	RESISTOR, 1/8W, 1%, 249 ohm	R137 R139 R149 R150 R155
20129.768.01	RESISTOR, 1/8W, 1%, 768 ohm	R111 R126
20130.162.01	RESISTOR, 1/8W, 1%, 1.62K	R132 R153 R156 R157
20130.178.01	RESISTOR, 1/8W, 1%, 1.78K	R502 R510 R515
20130.210.01	RESISTOR, 1/8W, 1%, 2.10K	R112 R127
20130.221.01	RESISTOR, 2.21K	R305 R318
20130.249.01	RESISTOR, 1/8W, 1%, 2.49K	R300 R315
20130.348.01	RESISTOR, 1/8W, 1%, 3.48K	R204 R210 R217 R220
20130.562.01	RESISTOR, 1/8W, 1%, 5.62K	R113 R128
20130.845.01	RESISTOR, 1/8W, 1%, 8.45K	R201 R202 R205 R207 R208 R211 R212 R214 R215 R218
20131.113.01	RESISTOR, 1/8W, 1%, 11.3K	R206 R219 R233 R234
20131.143.01	RESISTOR, 1/8W, 1%, 14.3K	R221 R224 R227 R230
20131.147.01	RESISTOR, 1/8W, 1%, 14.7K	R114 R129

20131.249.01	RESISTOR , 1/8W, 1% 24.9K	R203 R209 R213 R216
20131.499.01	RESISTOR, 1/8W, 1%, 49.9K	R222 R223 R228 R229 R501 R504 R507 R513 R520 R524 R525 R526 R239 R240 R241 R242
20131.825.01	RESISTOR, 1/8W, 1%, 82.5K	R104 R123 R303 R320
20132.154.01	RESISTOR, 1/8W, 1%, 154K	R328 R329
20511.310.01	TRIMPOTS, 10K, 20%, TOP ADJ	VR200 VR201
21112.210.01	CAPACITOR, CER, 0.001UF, 1KV, 10%	C100 C102 C104 C106
21123.510.01	CAPACITOR, RADIAL LEADS, 1.0uF, 50V, 20%	C224 C230
21137.282.01	CAPACITOR, 8200PF, +-15%, 50V	C507 C512 C518
21137.447.01	CAPACITOR .47UF 25V 10% 1206	C113 C117 C340 C350 C503 C506 C511
21138.247.01	CAPACITOR, SMD1206, 4700PF, 50V, 5%	C109 C110 C115 C116
21140.000.01	CAPACITOR, NPO, 470PF, 1%	C217 C218 C219 C220 C336 C339 C344 C349
21141.000.01	CAPACITOR, NPO, 1000PF, 1%	C226 C228 C337 C347 C517 C519 C521
21142.000.01	CAPACITOR, NPO, 100PF, 1%	C338 C348
21143.000.01	CAPACITOR, NPO, 1500PF, 1%	C221 C222 C225 C227
21144.000.01	CAPACITOR, 5%, 100V, 47PF	C101 C103 C105 C107 C108 C114 C333 C343
21145.000.01	CAPACITOR, NPO, 5%, 100V, 33PF, 1206	C231 C335 C346 C351
21156.020.01	CAPACITOR, 12pf	C223 C229 C334 C345
21263.710.01	CAPACITOR, RADIAL LEADS, 100uF, 25V, 10%	C304 C305 C308 C325 C326 C329
21318.510.01	CAPACITOR, TANTALUM, 1.0uF, 35V, B-CASE	C200 C201 C515 C516 C232
22101.001.01	DIODE, 1N4148WT/R	CR101 CR102 CR106 CR107
22102.001.01	DIODE 1N5711TR	CR500
22106.000.01	DIODE, SMCJ26C, TRAN20RB	CR100 CR103 CR104 CR105 CR202 CR203 CR204 CR205
23415.000.01	TRANSISTOR, JFET SST113 SURFACE MOUNT	Q200 Q201 Q202 Q203
24024.000.01	IC, OPA2134PA	IC100 IC102
24634.000.01	IC, OCTAL 3 STATE NONINVR	IC504
24728.302.01	IC, QUAD, SPST SW, DIP/16	IC101 IC103
24748.000.01	IC, LM339M S014	IC210
24857.000.01	IC 74HC374 DLATCH SOL20	IC108 IC209
24858.000.01	IC, SO/14, SURFACE MOUNT	IC604
24896.000.01	IC, SO/16, SURFACE MOUNT	IC507
24900.000.01	IC, HEX INVERTER, SURFACE MOUNT	IC603
24924.000.01	IC CSS3310KS	IC203
24938.000.01	IC, SINGLE 2 INPUT, SURFACE MOUNT	IC508
24957.000.01	IC, PCM1704U	IC300 IC304
24958.000.01	IC, DRV134PA, DIP	IC207 IC208
24960.000.01	IC, OPA2134UA	IC104 IC105 IC106 IC201 IC202 IC204 IC206 IC302 IC305
24961.000.01	IC, OPA627AP	IC308 IC309
24962.000.01	IC CS8420CS REV D	IC500 IC501 IC502
24963.000.01	IC, 5383 VS	IC107
24992.000.01	IC, 74AHCT244 SOIC	IC600 IC601
24997.000.01	IC, DAC AK4393 SSOP28	IC211
27106.000.01	CONNECTOR, BNC, RIGHT ANGLE, PC MOUNT	J301 J302
27147.008.01	IC, SOCKET, DIP, 8 PINS, DUAL	IC208 IC100 IC102 IC207
27147.016.01	IC, SOCKET, DIP, 16 PIN, DUAL	IC101 IC103
27147.020.01	IC, SOCKET, DIP, 20 PIN, DUAL	IC602
27174.044.01	IC, SOCKET, 44 PIN, LOW PROFILE	IC503

29015.000.01	TRANSFORMER, AES/EBU	T500 T501 T502
29508.210.01	FILTER, EMI SUPPRESSION, 50V-	L100 L102 L104 L106 L305 L306 L307 L500 L501 L502 L503 L504 L505 L200 L201 L202 L203
29521.000.01	INDUCTOR, 3.9UH, JM391K	L204 L205 L206 L207 L308 L309 L310
29522.000.01	INDUCTOR, 1200UH, 5%, 1-M-10-22	L101 L103 L105 L107
29707.002.01	INDUCTOR, FERRITE POT CORE, 3.501mh	L300 L312
29707.003.01	INDUCTOR, FERRITE POT CORE, 3.39mh	L301 L313
44082.100.01	FIRMWARE, FM I/O, IC503, 8400	IC503
44083.100.01	FIRMWARE, FM I/O, IC602, 8400	IC602
20151.365.01	RESISTOR, 0.1% 3.65K	R130 R133 R135 R136 R143 R145 R147 R148
20151.536.01	RESISTOR, 0.1%, 5.36K	R102 R109 R117 R122

Front Panel Subassembly

PART #	DESCRIPTION	COMPONENT IDENTIFIER
19555.000.01	KEY, CAPACITOR, ENTER,	
19556.000.01	KEY, CAPACITOR, ESCAPE	
19557.000.01	KNOB, J KNOB,	
19558.000.01	KNOB, ROTARY,	
19559.000.03	KNOB, VOLUME CONTROL	
43037.000.01	CABLE, 6P MOLX IF-HP	J2 TO J1
43038.000.01	CABLE, 16P, IDC-IDC 7.5	
50531.000.01	PANEL, FRONT, MOLDED FACE	
50532.000.01	PANEL, FRONT, MOLDED LEFT BEZEL	
50533.000.01	PANEL, FRONT, MOLDED RIGHT BEZEL	
50534.000.01	LENS, FRONT PANEL, 8400	
52000.300.01	COLOR LCD DISPLAY	

Display Circuit Board

PART #	DESCRIPTION	COMPONENT IDENTIFIER
240-003	INDUCTOR, 2A, 2.2UH	L1
20121.100.01	RESISTOR, RF, 1/8W, 1%, 10 ohm	R1
20129.100.01	RESISTOR, 100 ohm	R27
20130.100.01	RESISTOR, 1.00K 1%	R14 R29
20130.221.01	RESISTOR, 2.21K	R4 R6 R11 R12
20130.562.01	RESISTOR, 1/8W, 1%, 5.62K	R28
20131.100.01	RESISTOR, 10K	R15 R16 R17 R18 R19 R20 R25 R30 R31 R32 R33 R34
20131.140.01	RESISTOR, 14.0K	R3 R5 R8 R10
20131.200.01	RESISTOR, 20.0K 1%	R2 R7 R9 R13
20132.100.01	RESISTOR, 100K	R21 R24 R26
20135.121.01	RESISTOR, 1.21M	R22 R23
20896.000.01	DIGITALLY-CONTROLLED POT, MAX5160LEUA	U7
21139.000.01	CAPACITOR, X7R, 0.1UF, 10%	C3 C4 C5 C8 C11 C12 C13 C15 C18 C19 C20
21142.000.01	CAPACITOR, NPO, 100PF, 1%	C17
21147.022.01	CAPACITOR, 22pf, 1%	C9 C10
21157.218.01	CAPACITOR 1800PF 50V 5%	C1 C2
21318.510.01	CAPACITOR, TANTALUM, 1.0uF, 35V, B-CASE	C6 C7

21323.615.01	CAPACITOR, 15uf, TANTALUM, C-CASE	C14
21324.610.01	CAPACITOR, 10uF, 35V, D-CASE	C16
22209.000.01	DIODE, SHOT 1A, 60V, SMD	D1
23214.000.01	TRANSISTOR NPN MMBT3904	Q1
24747.000.01	IC, MAX6501, TEMP 65c	U5
24959.000.01	IC, CS4390KS	U1
24960.000.01	IC, OPA2134UA	U2
24988.000.01	IC, SED1353FOA	U4
24989.000.01	IC, LT1613C35	U6
24990.000.01	IC, CY62128, 32P, TSOP	U9 U10
24994.000.01	IC, 74ACT04, SOIC 14P	U11
28086.000.01	CRYSTSL, 4.0 MHz, HC49US	Y1
29523.000.01	INDUCTOR, 636CY-150M	L2
29525.000.01	INDUCTOR, 1812-564J	L3
44084.100.01	FIRMWARE, DISPLAY INTERFACE, U3, 8400	U3

Display Interface Circuit Board

PART #	DESCRIPTION	COMPONENT IDENTIFIER
20122.274.01	RESISTOR, TF, 1/8W, 1%, 274 ohm	R604 R605
20131.100.01	RESISTOR, 10K	R301 R302 R303 R304 R305 R306 R505 R801 R802
20132.100.01	RESISTOR, 100K	R101 R102 R103 R104 R502 R503 R504 R601 R602 R603
20221.101.01	RESISTOR, NET, SINGLE-INLINE-PACKAGE, 2%, 100K, 10-PIN	R501
21137.282.01	CAPACITOR, 8200PF, ±15%, 50V	C101, C103, C105, C107, C109, C111, C113, C115
21137.447.01	CAPACITOR 0.47UF 25V 10% 1206	C102 C104 C106 C108 C110 C112 C114 C116
21227.747.01	CAPACITOR, RADIAL LEADS, 470uF 16V HFS	C764 C765
21230.710.01	CAPACITOR, RADIAL LEADS, 100uF 50V HFS	C763
21305.610.01	CAPACITOR, RADIAL LEADS, 10uF, 20V, 10%	C766
21309.622.01	CAPACITOR, TANTALUM, SURFACE MOUNT, 22	C735 C736
22083.033.01	DIODE, SURGE SUPPRESSOR, 33 VOLT	CR700
22083.068.01	DIODE, SURGE SUPPRESSOR, 6.8 VOLT	CR702
22208.040.01	DIODE, SCHOTTKY, 31DQ04, 3.3	CR701
24943.000.01	IC, TQFP, SURFACE MOUNT, 8 MHZ, 74AHC08	IC801
24944.000.01	IC, EPM 7064AETC44-10, SURFACE MOUNT	IC503
24945.000.01	IC 74AHC541 OCTLBUF SOL20	IC501
24946.000.01	IC, 8 BIT-DUAL TRANSVR W/3	IC502
24948.000.01	IC, 74LVC2244 OCTLBUFSOL20	IC601, IC602, IC605
24949.000.01	IC, INVERTER, SURFACE MOUNT	IC604
24964.000.01	IC, LM2576T, 3.3 FLOW LB03	IC700
24991.000.01	IC, DSP, MOTOROLA 56362PV100	IC101 IC102 IC103 IC104 IC105 IC106 IC107 IC108
24993.000.01	IC, EPM7256AETC100-10	IC603
29512.000.01	INDUCTOR, SHIELDED, 1670-1; 25	L702
29526.000.01	INDUCTOR, PE92108K	L700

Composite Input/Output Circuit Board

PART #	DESCRIPTION	COMPONENT IDENTIFIER
20039.750.01	RESISTOR, MF, 1/8W, 1%, 75.0 ohm	R13 R14
20040.604.01	RESISTOR, MF, 1/8W, 1%, 604 ohm	R5 R6 R7 R8
20130.162.01	RESISTOR, 1/8W, 1%, 1.62K	R9 R10
20130.200.01	RESISTOR, 2.00K	R11 R12
20131.200.01	RESISTOR, 20.0K 1%	R1 R2 R3 R4
21139.000.01	CAPACITOR, X7R, 0.1UF, 10%	C11 C12 C3 C4 C5 C6
21143.000.01	CAPACITOR, NPO, 1500PF, 1%	C7 C8 C9 C10
21145.000.01	CAPACITOR, NPO, 5%, 100V, 33PF, 1206	C1 C2
24024.000.01	IC, OPA2134PA	IC3
24025.000.01	IC, BUF634P, DIP8	IC1 IC2
29508.210.01	FILTER, EMI SUPPRESSION, 50V-	L1 L2 L3 L4
29521.000.01	INDUCTOR, 3.9UH, JM391K	L5 L6 L7 L8

Control Circuit Board

PART #	DESCRIPTION	COMPONENT IDENTIFIER
20080.301.01	RESISTOR, MF, 1/2W, 1%, 301 ohm	R68
20121.100.01	RESISTOR, RF, 1/8W, 1%, 10 ohm	R20 R59 R96 R97
20121.750.01	RESISTOR, TF, 1/8W, 1%, 75 ohm	R30 R47 R78 R79 R80 R81 R83 R84 R85 R86 R94 R82
20123.499.01	RESISTOR, TF, 1/8W, 1%, 4.99K	R89 R91
20129.301.01	RESISTOR, 301 ohm	R130
20130.100.01	RESISTOR, 1.00K 1%	R29 R36 R46 R49 R52 R55 R58 R61 R64 R117 R120 R92 R93
20130.162.01	RESISTOR, 1/8W, 1%, 1.62K	R69 R70
20130.200.01	RESISTOR, 2.00K	R124
20130.221.01	RESISTOR, 2.21K	R19
20130.332.01	RESISTOR, 1% 3.32K	R76 R140
20130.365.01	RESISTOR, 1/8W, 1%, 3.65K	R90
20130.475.01	RESISTOR, 4.75K	R31 R32 R33 R34
20130.562.01	RESISTOR, 1/8W, 1%, 5.62K	R75
20131.140.01	RESISTOR, 14.0K	R126 R87
20131.200.01	RESISTOR, 20.0K 1%	R88
20131.301.01	RESISTOR, 30.1K	R141
20131.499.01	RESISTOR, 1/8W, 1%, 49.9K	R23
20132.100.01	RESISTOR, 100K	R24 R28 R45 R48 R51 R54 R57 R60 R63 R66 R67 R116 R95 R105 R106 R107 R108 R109 R110 R111 R112
20132.332.01	RESISTOR, 332K	R21
21140.000.01	CAPACITOR, NPO, 470PF, 1%	C19 C36
21141.000.01	CAPACITOR, NPO, 1000PF, 1%	C18 C23 C25 C28 C31 C34 C47 C58 C63 C70
21145.000.01	CAPACITOR, NPO, 5%, 100V, 33PF, 1206	C37
21146.310.01	CAPACITOR, 0.01uf, 10%	C17 C22 C24 C27 C30 C33 C45 C57 C62 C68 C94
21147.022.01	CAPACITOR, 22pf, 1%	C20 C89 C92 C93
21149.015.01	CAPACITOR, 15pf, 1%	C40
21150.133.01	CAPACITOR, 330pf, 1%	C39
21154.433.01	CAPACITOR, 0.33uf, 20%	C133, C134, C135
21209.710.01	CAPACITOR, RADIAL LEADS, 100uF, 63V, 20%	C100

21318.510.01	CAPACITOR, TANTALUM, 1.0uF, 35V, B-CASE	C1 C14 C15 C35 C48 C60 C78 C88
21321.633.01	CAPACITOR, 33uf, TANTALUM, 6032 D	C26 C59
21322.547.01	CAPACITOR, 4.7uf, TANTALUM, 6032B	C43 C51 C53
22016.000.01	DIODE, MMSZ5231B, SOD-123	D11
22083.015.01	DIODE, SURGE SUPPRESSOR, 15 VOLT	D10
22101.001.01	DIODE, 1N4148WT/R	D1 D3 D6 D7 D8 D9
22209.000.01	DIODE, SHOT 1A, 60V, SMD	D15 D16
23214.000.01	TRANSISTOR NPN MMBT3904	Q1 Q2
24326.000.01	IC, REGULATOR, 1086, 3.3 VOLT	U8 U22
24327.000.01	IC, REGULATOR, 1086, 3.6 VOLT	U19

Schematics, Assembly Drawings

The following drawings are included in this manual. These are the sections of the 8400 likely to contain parts serviceable outside the Orban factory.

Page	Function	Description	Drawing
6-22	Chassis	Circuit Board Locator	Top view
6-23	Backplane Interface Board	Interconnects main circuit boards	Schematic 1 of 1
6-24	Power supply		Assembly Drawing
6-25			Schematic 1 of 1
6-26	CPU/Remote/RS232	Serial port 1 & 2 support	Assembly Drawing
6-27			Schematic 1 of 1
6-28	Display Interface board	*containing*:	Assembly Drawing
6-29		Connectors and Headers	Schematic 1 of 3
6-30		Color LCD support	Schematic 2 of 3
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6-32	Rotary encoder board		Assembly Drawing
6-33			Schematic 1 of 1
6-34	Headphone board		Assembly Drawing
6-35			Schematic 1 of 1
6-36	Composite I/O	Daughterboard with BNCs	Assembly Drawing
6-37			Schematic 1 of 1
6-38	FM Input/Output Board	*containing*:	Assembly Drawing
6-39		Analog Input	1 of 5
6-40		Analog Output	2 of 5
6-41		Composite and SCA	3 of 5
6-42		Control and Digital I/O	4 of 5
6-43		Interface and Power Distribution	5 of 5
6-44	Block Diagram	Shows signal processing	

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 6-14. Please note that because surface-mount parts are used extensively in the 8400, 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.

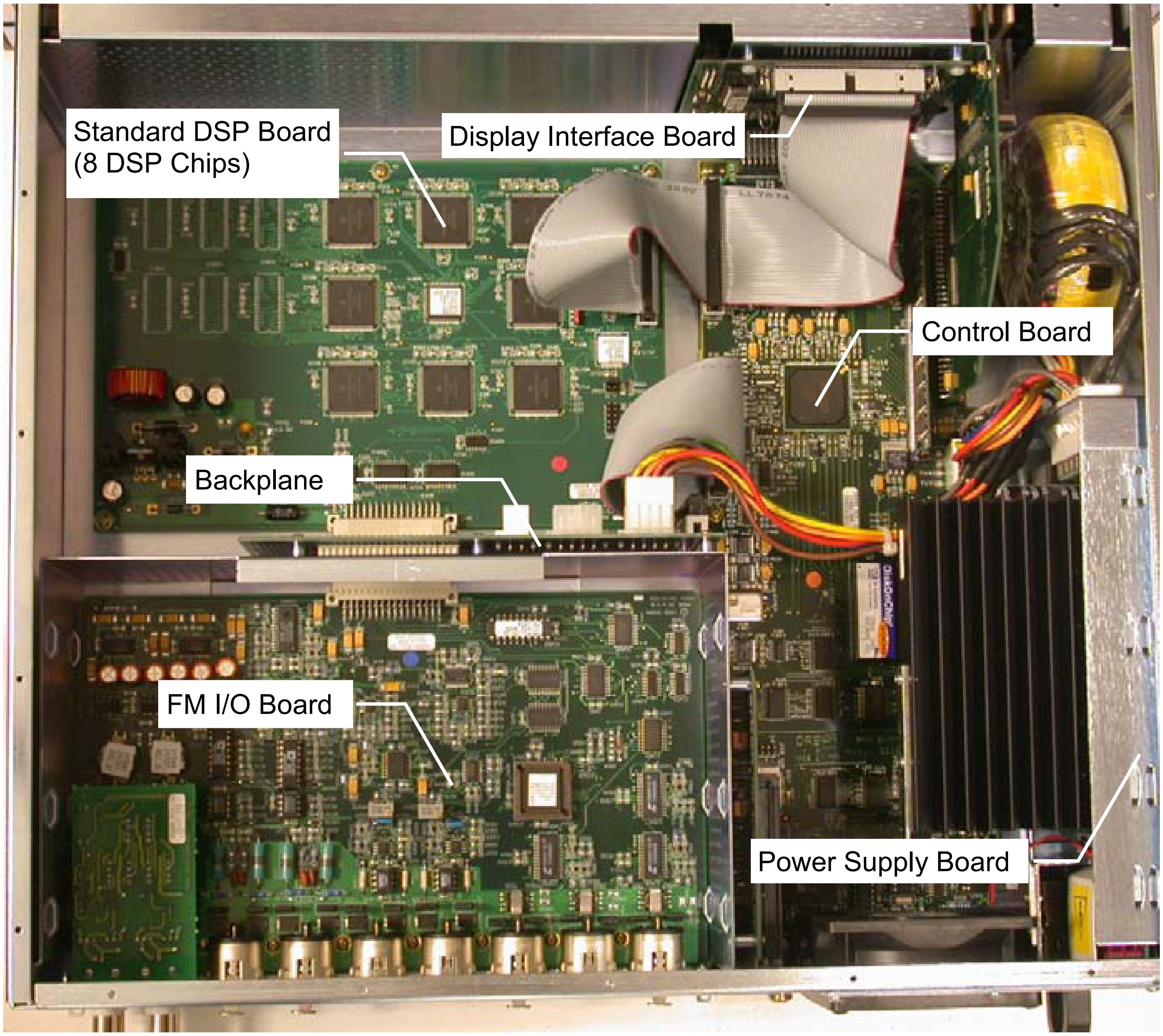
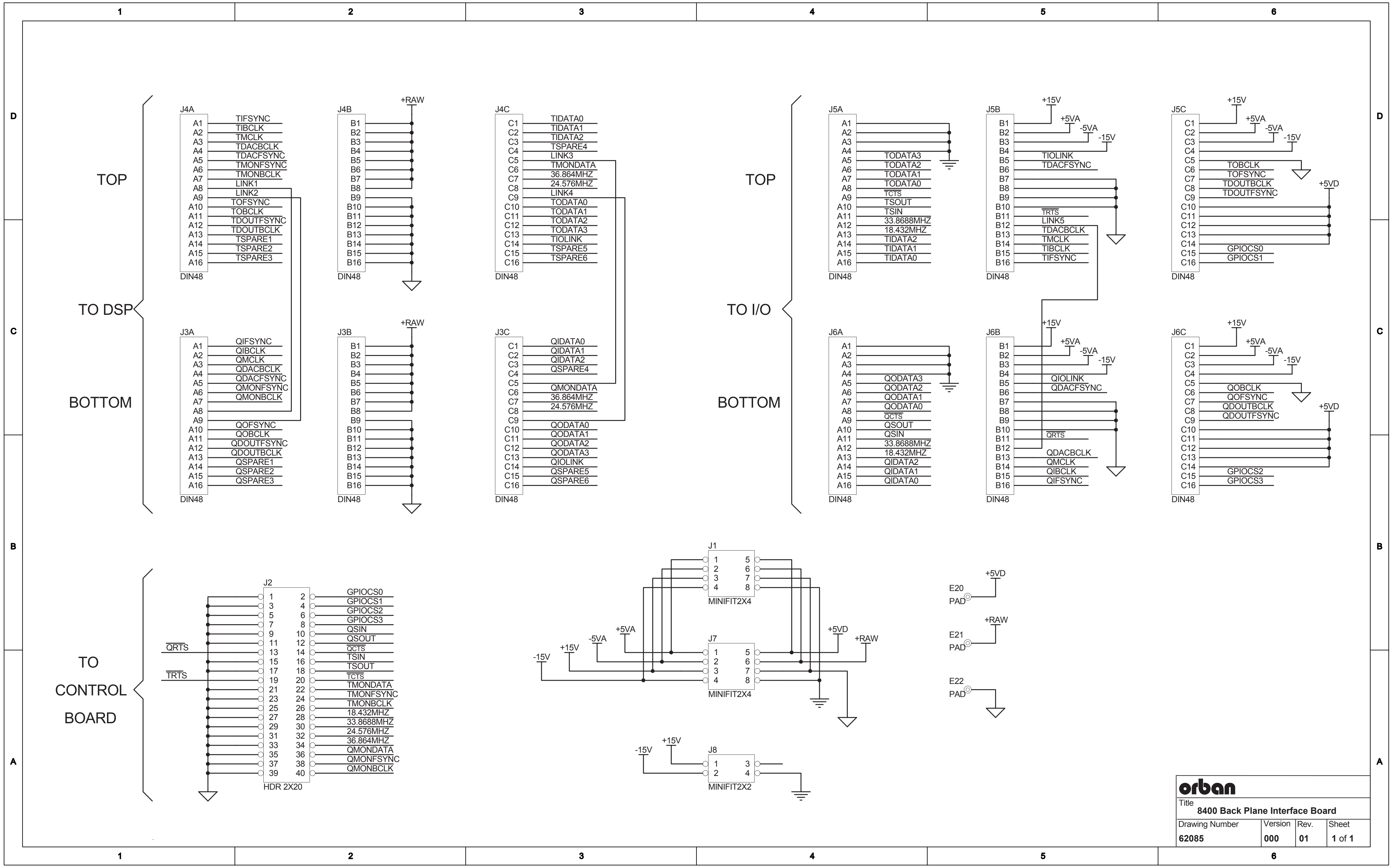
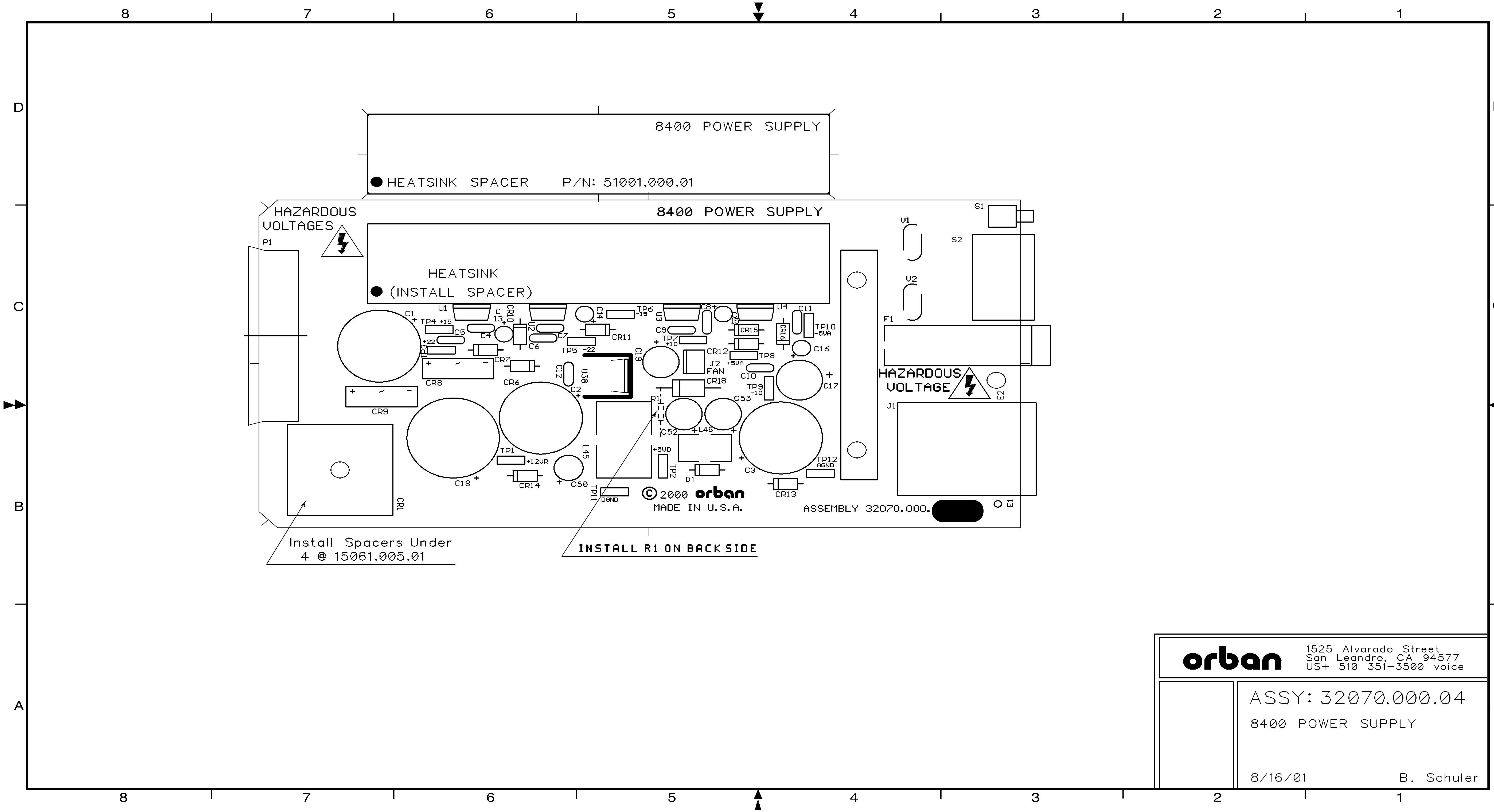
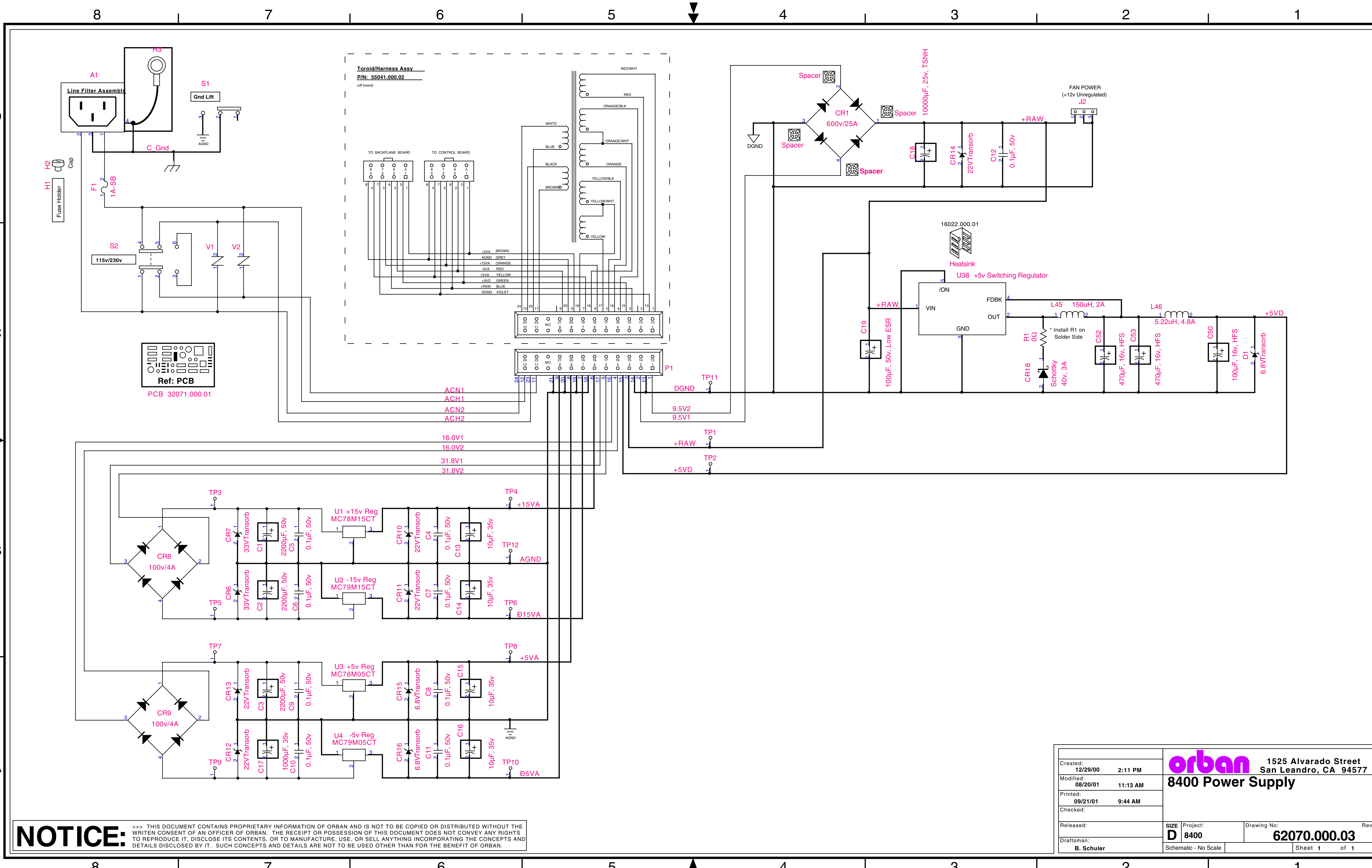


Figure 6-1: Main Circuit Board Locator


TOP
TO DSP
BOTTOM
TOP
TO I/O
BOTTOM
TO
CONTROL
BOARD
J4A
J4B
J4C
J5A
J5B
J5C
J3A
J3B
J3C
J6A
J6B
J6C
DIN48
J2
HDR 2X20
J1
MINIFIT2X4
J7
MINIFIT2X4
J8
MINIFIT2X2
E20
E21
E22
PAD
+RAW
+15V
-15V
+5VA
-5VA
+5VD
orban
Title
8400 Back Plane Interface Board
Drawing Number
62085
Version
000
Rev.
01
Sheet
1 of 1



8400 POWER SUPPLY
HEATSINK SPACER P/N: 51001.000.01
HAZARDOUS VOLTAGES
8400 POWER SUPPLY
HEATSINK
(INSTALL SPACER)
HAZARDOUS VOLTAGE
© 2000 orban
MADE IN U.S.A.
ASSEMBLY 32070.000.
Install Spacers Under
4 @ 15061.005.01
INSTALL R1 ON BACK SIDE
orban
1525 Alvarado Street
San Leandro, CA 94577
US+ 510 351-3500 voice
ASSY: 32070.000.04
8400 POWER SUPPLY
8/16/01
B. Schuler


Line Filter Assembly
Gnd Lift
C_Gnd
Fuse Holder
1A-SB
115v/230v
Ref: PCB
PCB 32071.000.01
Toroid/Harness Assy
P/N: 55041.000.02
TO BACKPLANE BOARD
TO CONTROL BOARD
CR1
600v/25A
Spacer
DGND
+RAW
FAN POWER
(+12v Unregulated)
J2
10000µF, 25v, TSNH
22VTransorb
0.1µF, 50v
16022.000.01
Heatsink
U38 +5v Switching Regulator
/ON
VIN
FDBK
OUT
GND
L45 150uH, 2A
L46
5.22uH, 4.8A
+5VD
100µF, 50v, Low ESR
* Install R1 on Solder Side
Schottky
40v, 3A
470µF, 16v, HFS
100µF, 16v, HFS
6.8VTransorb
P1
ACN1
ACH1
ACN2
ACH2
16.0V1
16.0V2
31.8V1
31.8V2
9.5V2
9.5V1
U1 +15v Reg
MC78M15CT
U2 -15v Reg
MC79M15CT
U3 +5v Reg
MC78M05CT
U4 -5v Reg
MC79M05CT
CR8
100v/4A
CR9
100v/4A
33VTransorb
2200µF, 50v
1000µF, 35v
10µF, 35v
+15VA
AGND
Ð15VA
+5VA
Ð5VA
1525 Alvarado Street
San Leandro, CA 94577
8400 Power Supply
Created: 12/29/00 2:11 PM
Modified: 08/20/01 11:13 AM
Printed: 09/21/01 9:44 AM
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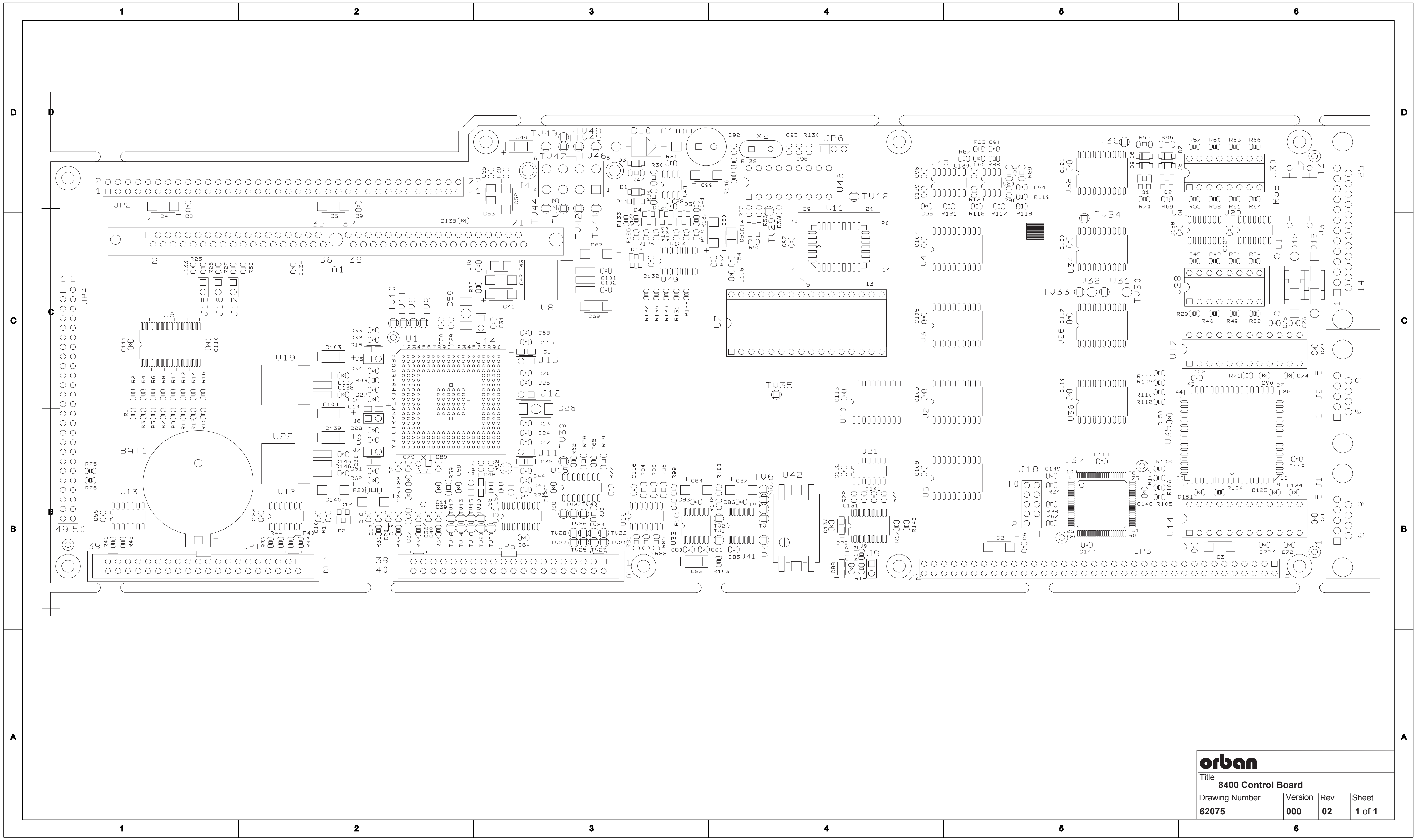
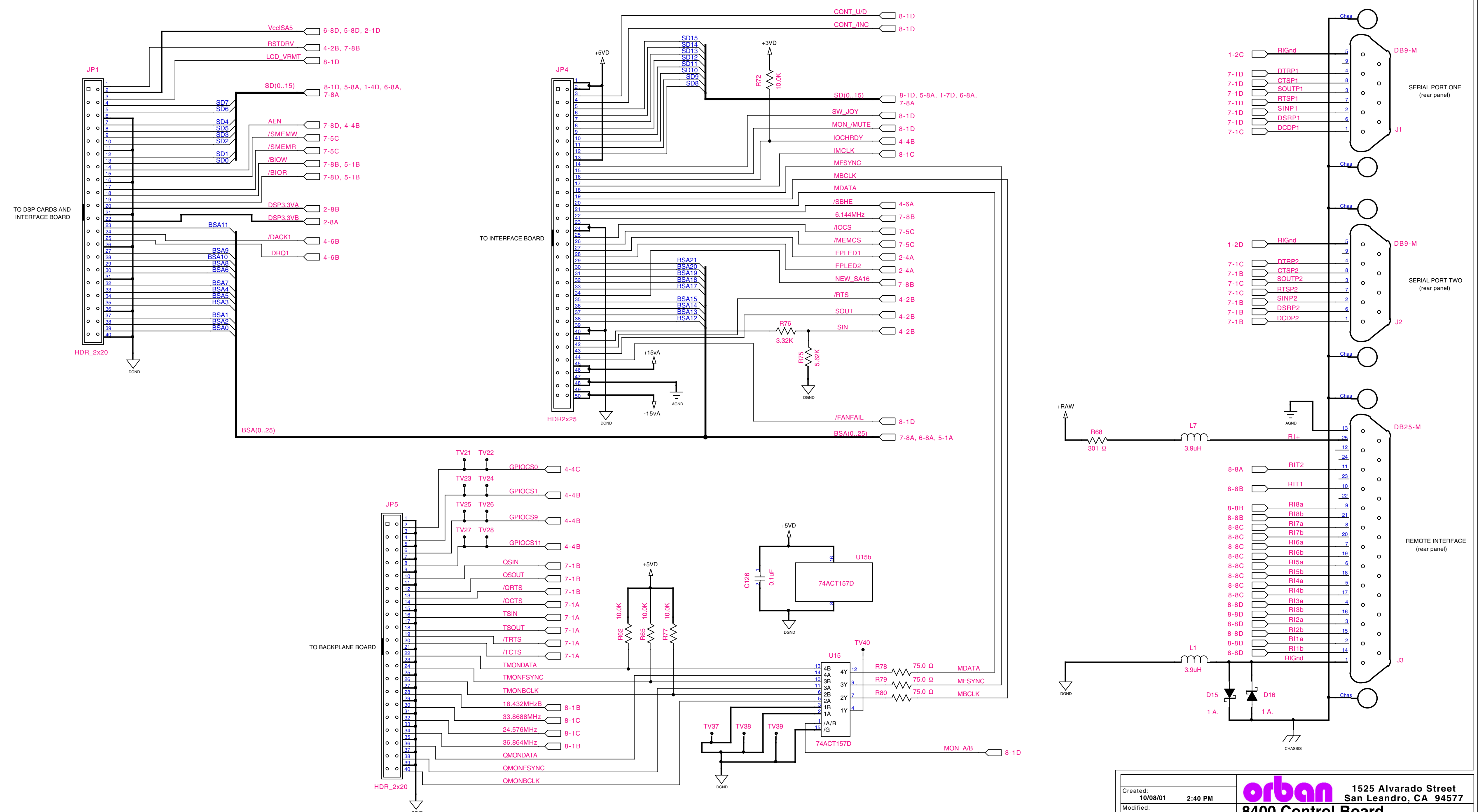
Released:
Draftsman: B. Schuler
SIZE D
Project: 8400
Drawing No: 62070.000.03
Rev.
Schematic - No Scale
Sheet 1 of 1


orban
Title
8400 Control Board
Drawing Number
62075
Version
000
Rev.
02
Sheet
1 of 1



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Printed:	10/08/01 2:50 PM
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Released:	
Draftsman:	B.Schuler

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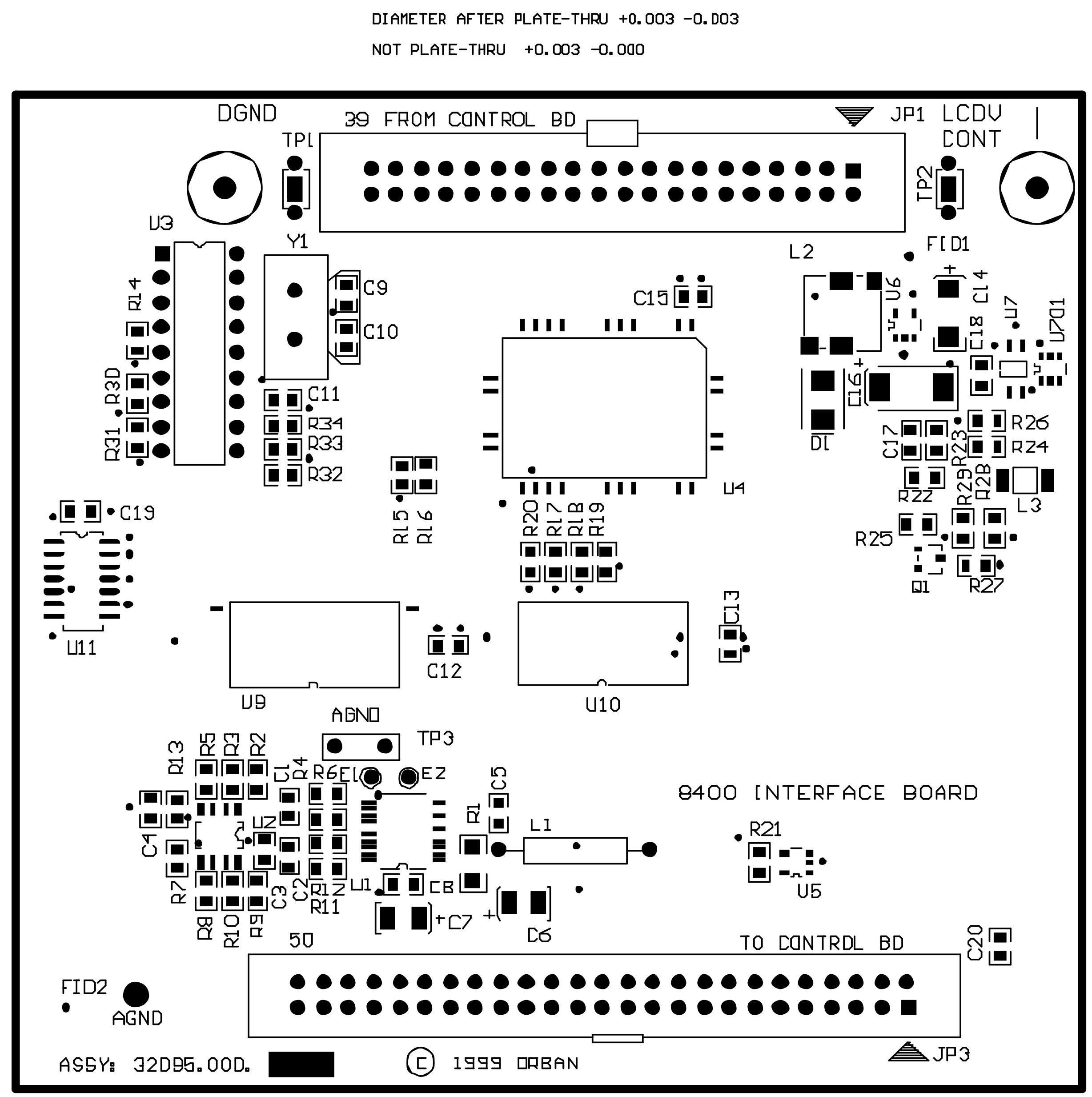
8400 Control Board

CONNECTORS

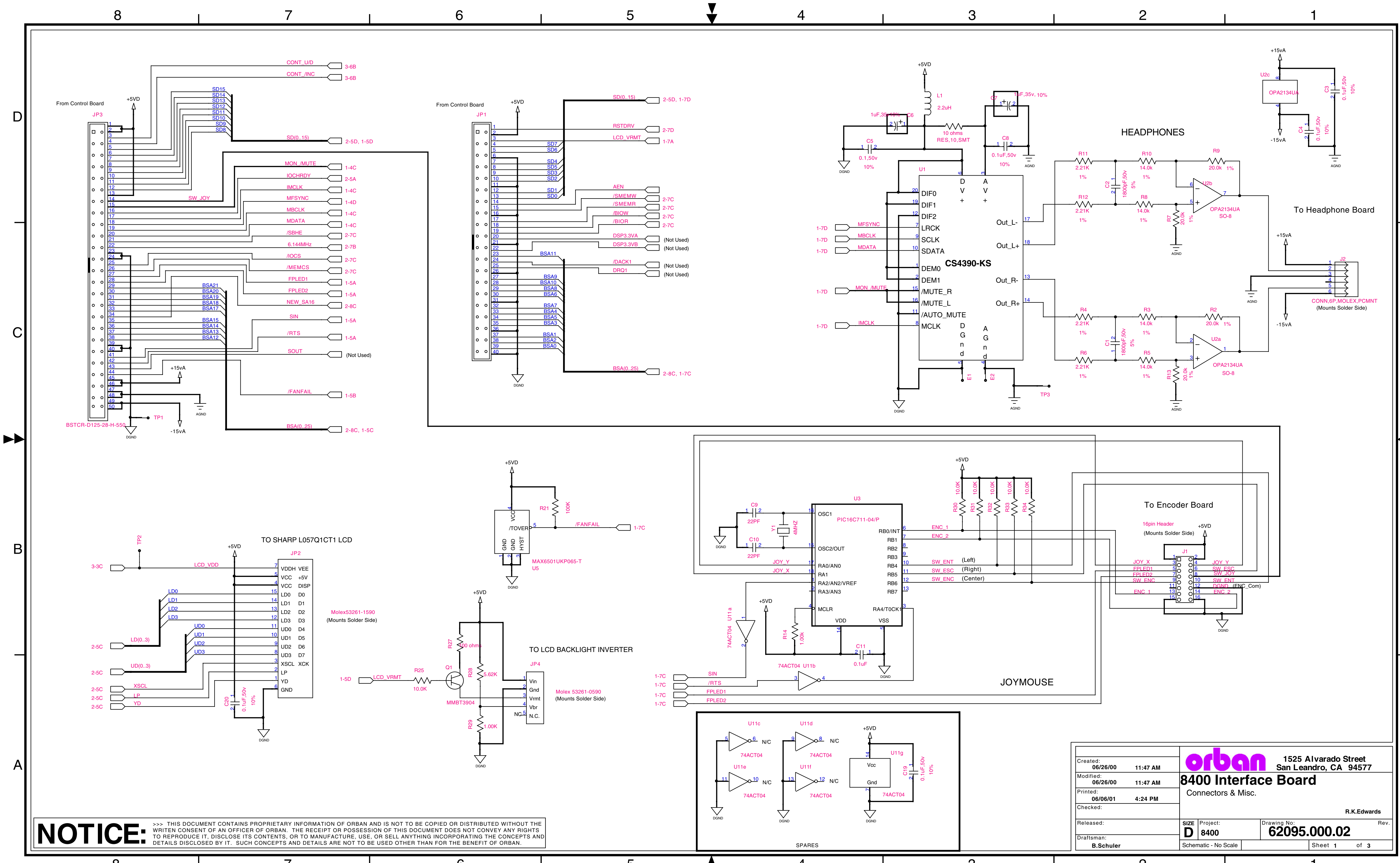
BC/RS

SIZE D Project: 8400 Drawing No: 62080.000.04 Rev.

Schematic - No Scale Sheet 1 of 8



orban			
Title: 8400 PCA interface			
Drawing Number	Version	Rev.	Sheet
62095	000	02	1 of 1



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Modified:	06/26/00 11:47 AM
Printed:	06/06/01 4:24 PM
Checked:	
Released:	
Draftsman:	B.Schuler

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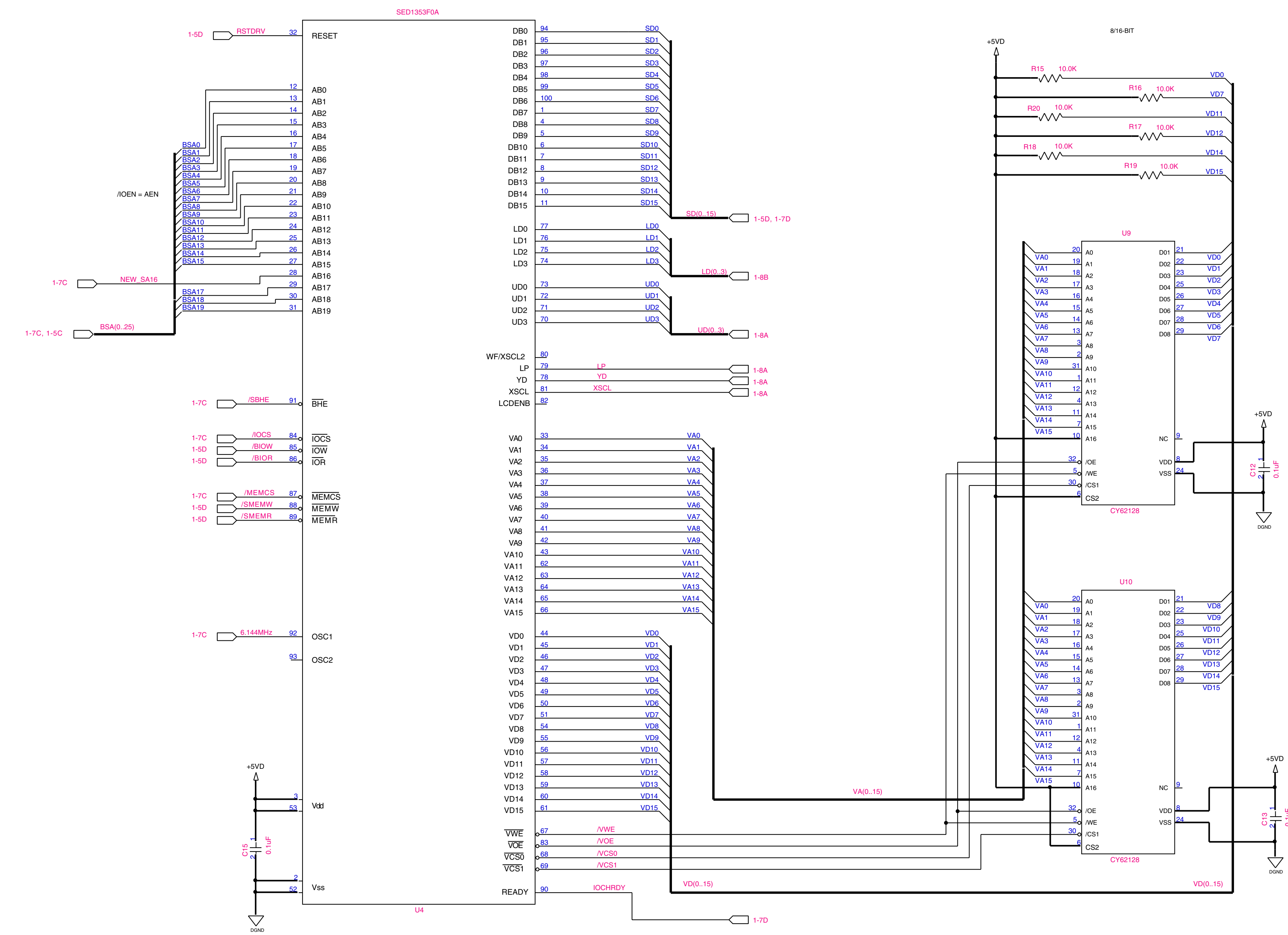
8400 Interface Board
Connectors & Misc.

R.K.Edwards

Size: D | Project: 8400 | Drawing No: 62095.000.02 | Rev.

Schematic - No Scale | Sheet 1 of 3



orban 1525 Alvarado Street San Leandro, CA 94577

8400 Interface Board

LCD Support

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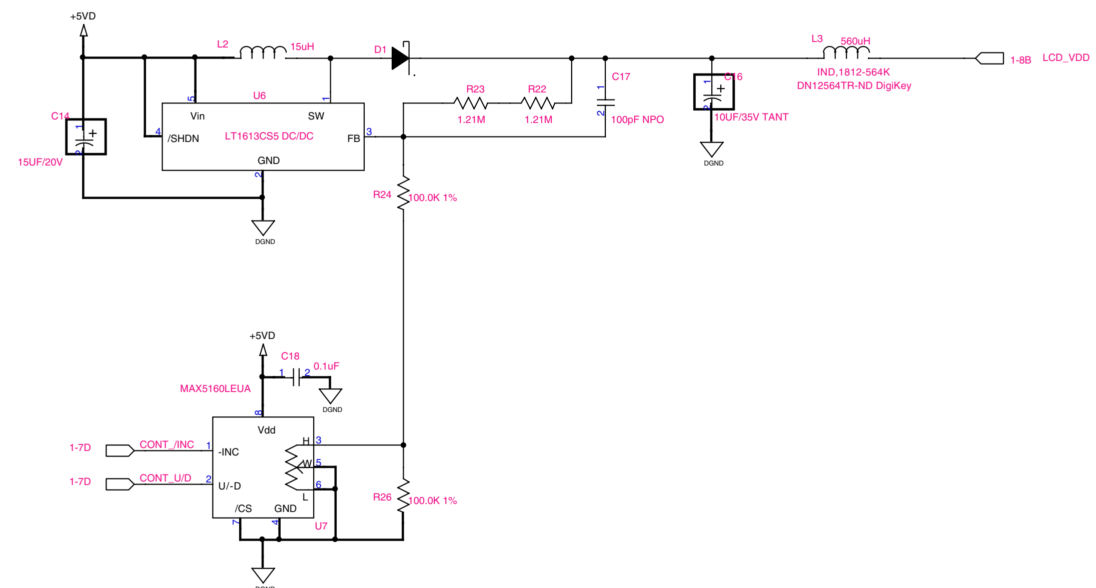
Released:	
Draftsman:	B.Schuler

R.K.Edwards

SIZE D | Project: 8400 | Drawing No: 62095.000.02 | Rev.

Schematic - No Scale | Sheet 2 of 3



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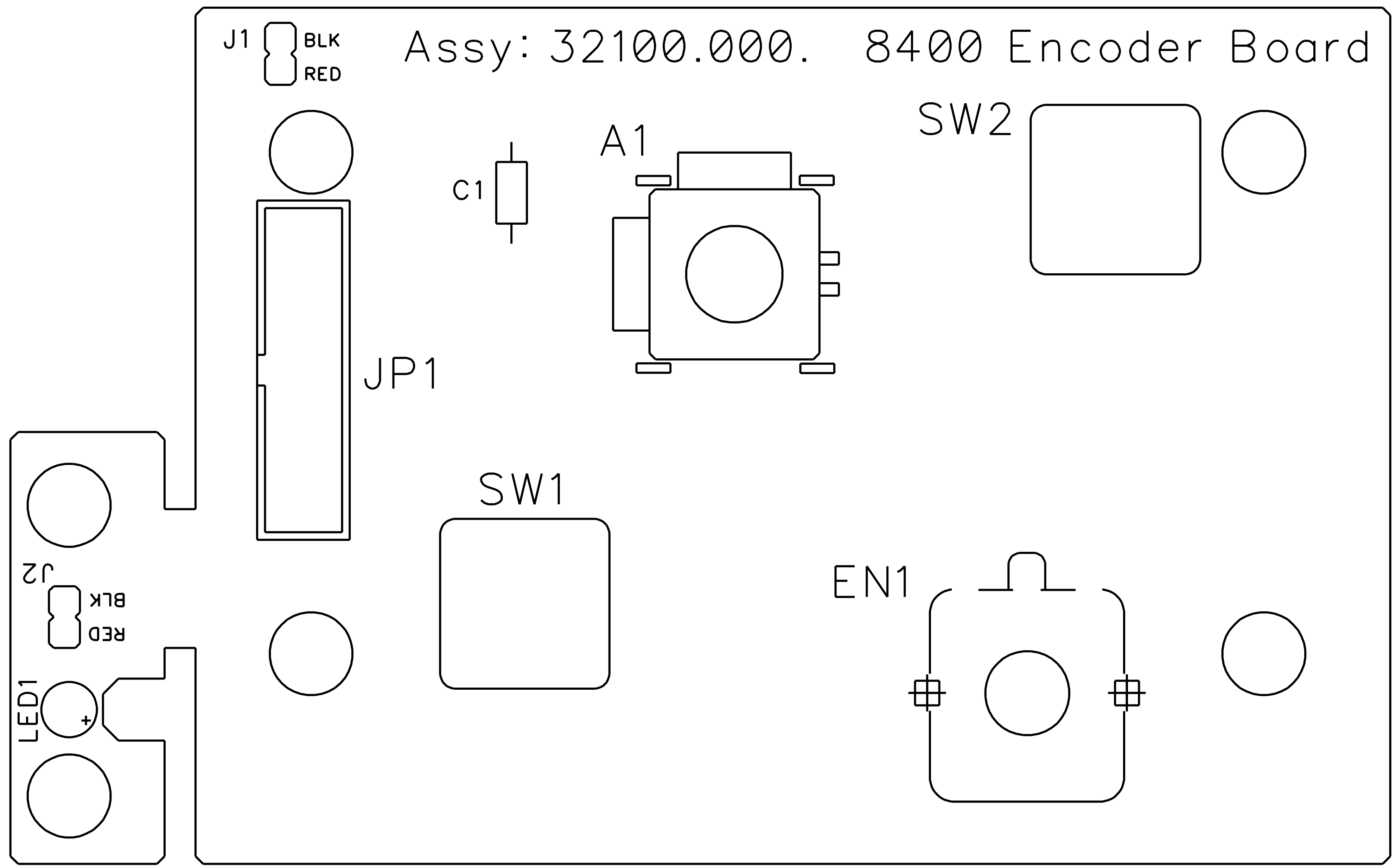
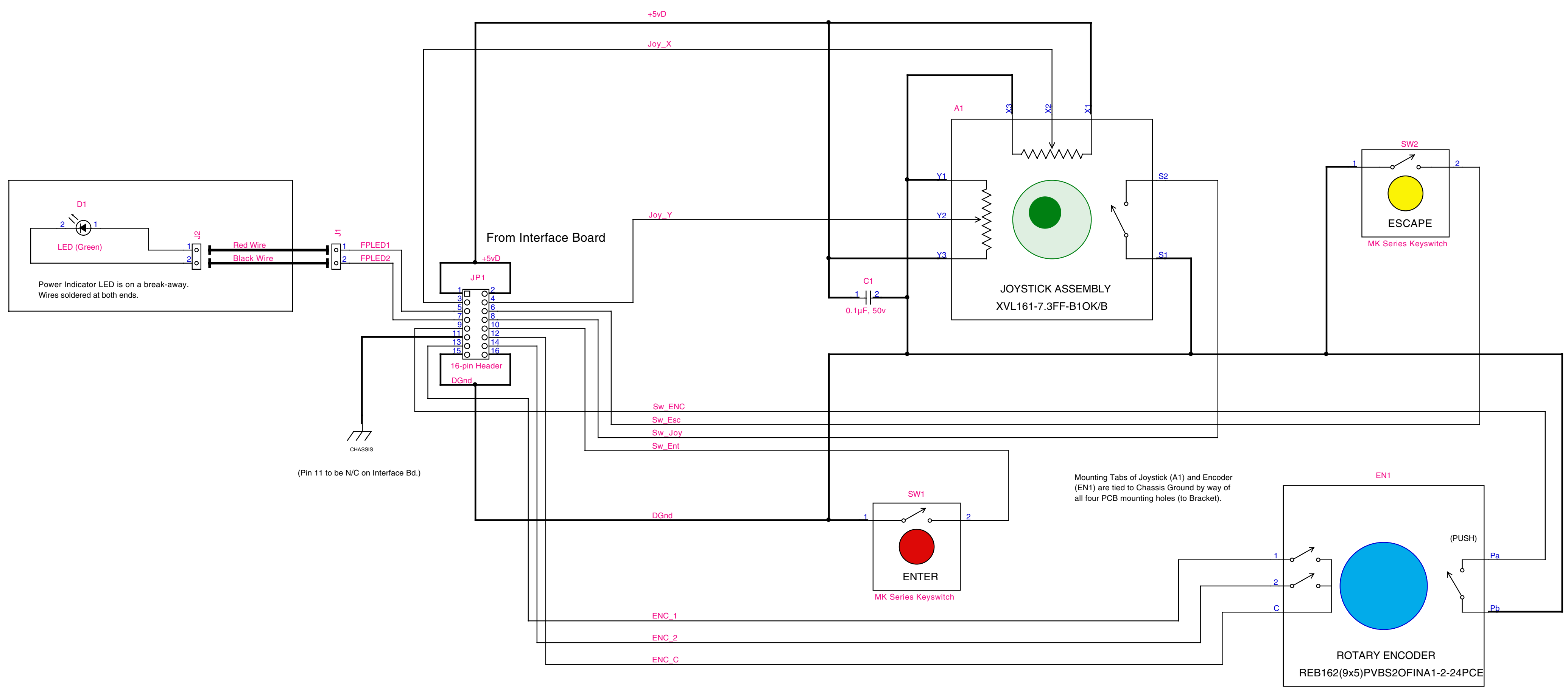
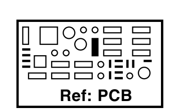
Modified: 06/26/00 11:47 AM	8400 Interface Board
Printed: 06/06/01 4:24 PM	LCD Contrast Adj Supply
Checked:	R.K.Edwards
Released:	SIZE D · Project: 8400 · Drawing No: 62095.000.02 · Rev.
Draftsman: B.Schuler	Schematic - No Scale · Sheet 3 of 3



Assy: 32100.000. 8400 Encoder Board
J1
BLK
RED
C1
A1
SW2
JP1
SW1
EN1
J2
BLK
RED
LED1
+





32101.000.03
(Part of panel # 32131.000.03)

Revision History:

ECO#	DATE	REV	DESCRIPTION	DONE	CHECKED
N/A	06/15/99	0A	First Prototype (original mechanical spec.)	-B.	
N/A	09/17/99	0B	Peter Edwards moves everything (mechanical change 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.	
2788	04/23/2K	03	Changes to Chassis Ground scheme	-B.	

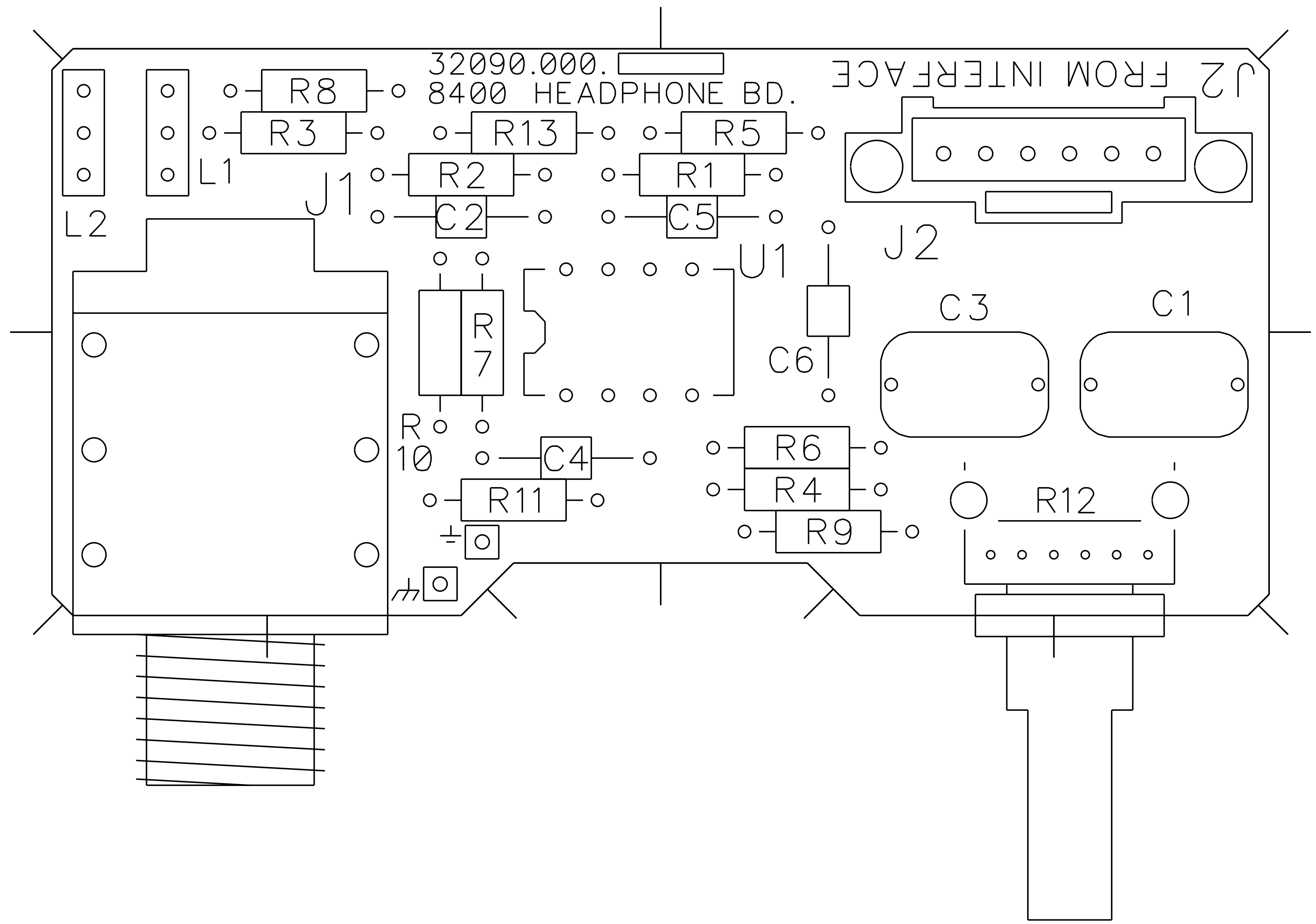
Created: 06/02/99 1:20 PM
Modified: 04/19/00 10:52 AM
Printed: 04/16/01 3:01 PM
Checked:
Released:
Draftsman: B. Schuler

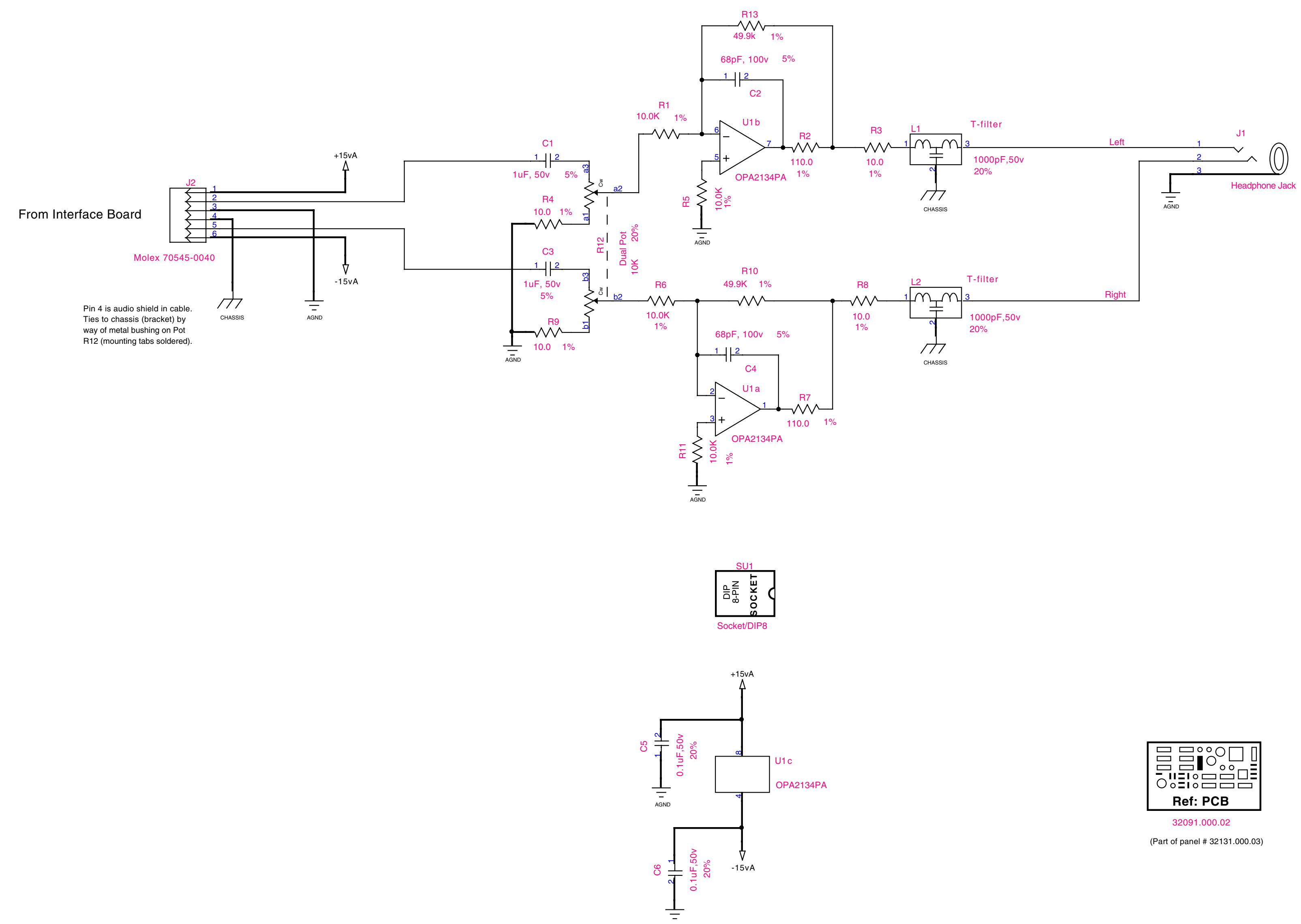
orban 1525 Alvarado Street San Leandro, CA 94577

8400 Encoder Board

SIZE D | Project: 8400 | Drawing No: 62100.000.03 | Rev.

Schematic - No Scale | Sheet 1 of 1


32090.000.
8400 HEADPHONE BD.
J2 FROM INTERFACE
R8
R3
R13
R5
R2
R1
C2
C5
L1
L2
J1
J2
U1
R7
C6
C3
C1
R10
C4
R6
R4
R11
R9
R12



orban 1525 Alvarado Street San Leandro, CA 94577

8400 Headphone Board

Created:	09/01/99	4:34 PM
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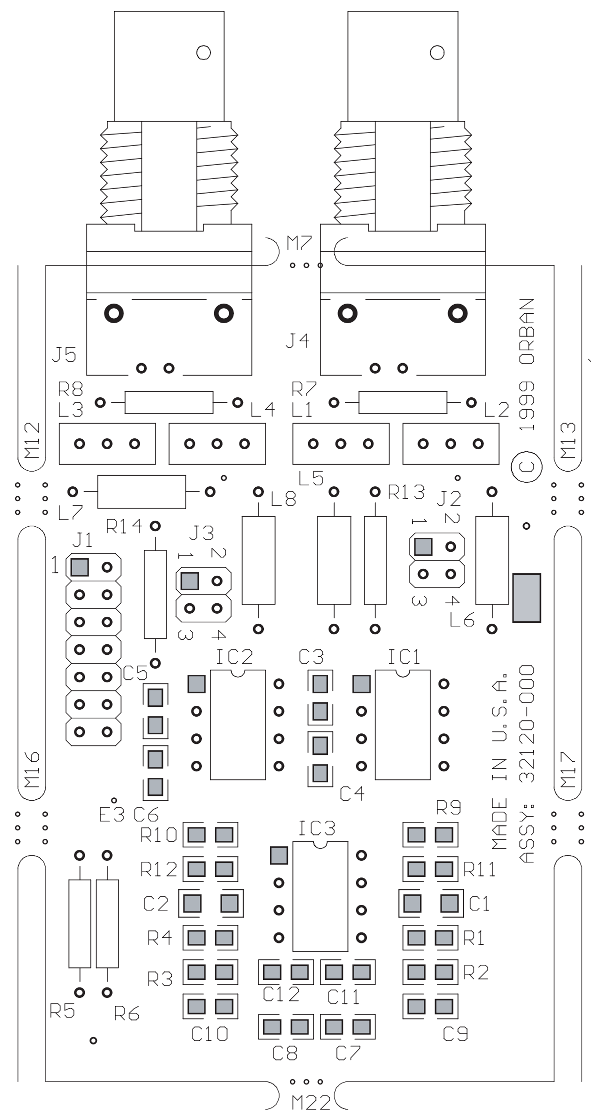
Printed:	04/16/01	3:00 PM
Checked:		
Released:		
Draftsman:	B. Schuler	

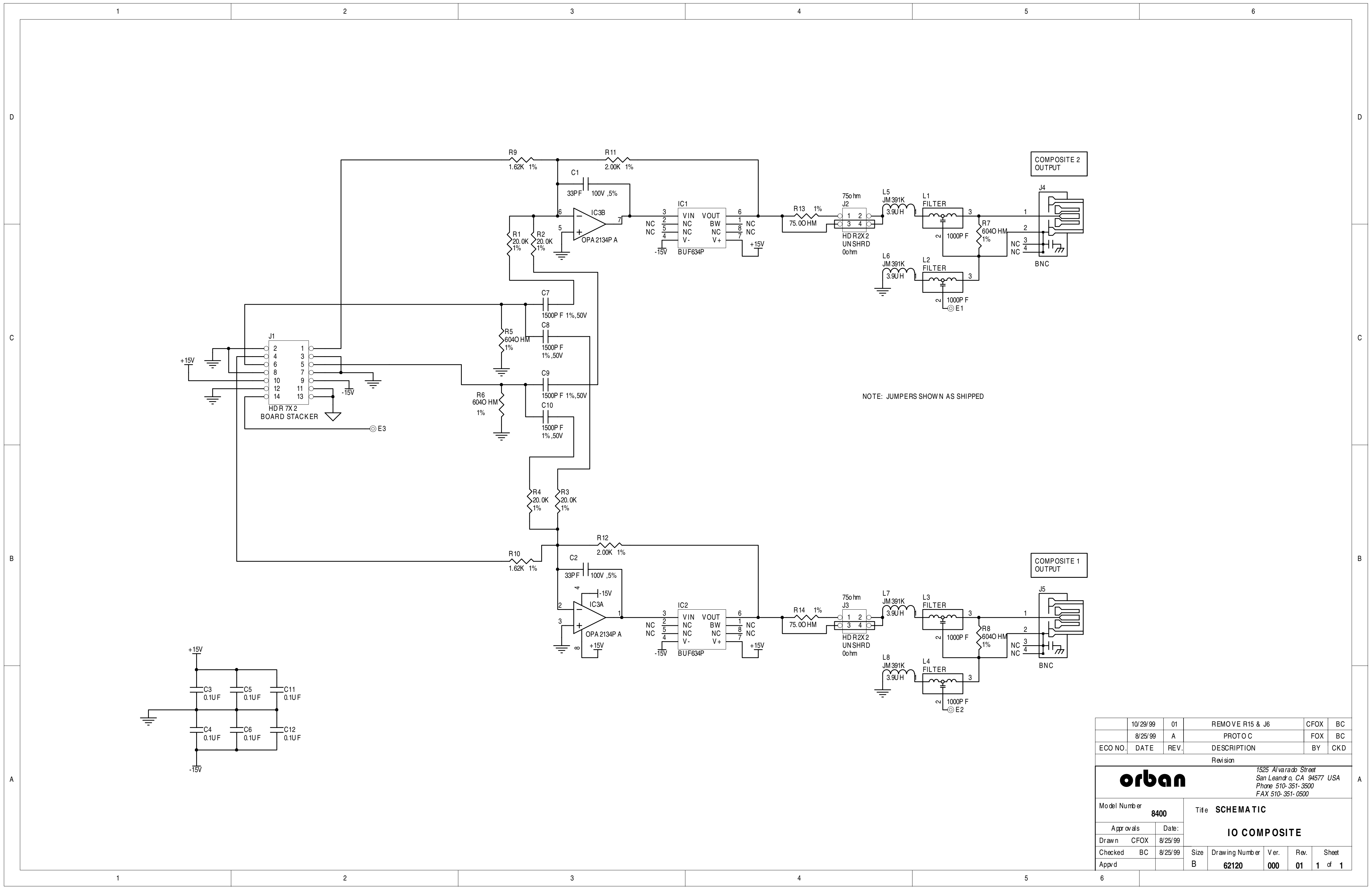
B. Coykendall

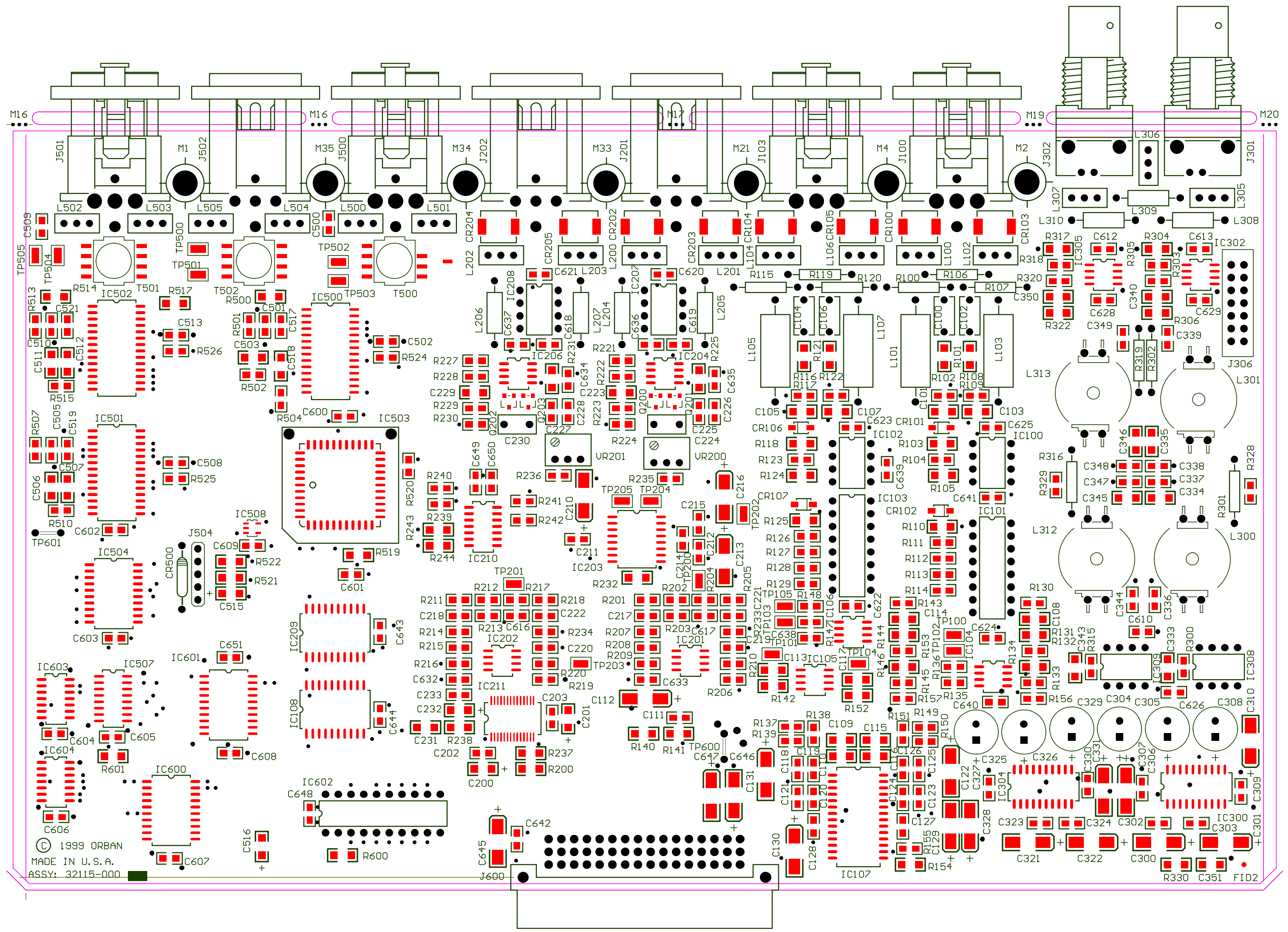
SIZE	Project:	Drawing No:	Rev.
D	8400	62090.000.02	

Schematic - No Scale — Sheet 1 of 1



ORBAN®		1525 Alvarado Street San Leandro Street, CA 94577 USA Phone 1-510-351-3500 Fax 1-510-351-0500				
MODEL NO.	8400	TITLE PCA I/O COMPOSITE				
APPROVALS	DATE					
DWN CFOX	8/30/99					
CKD JT	12/3/99	SIZE	DWG NO.	VER	REV	SHEET
APVR		D	32120	000	02	1 OF 1

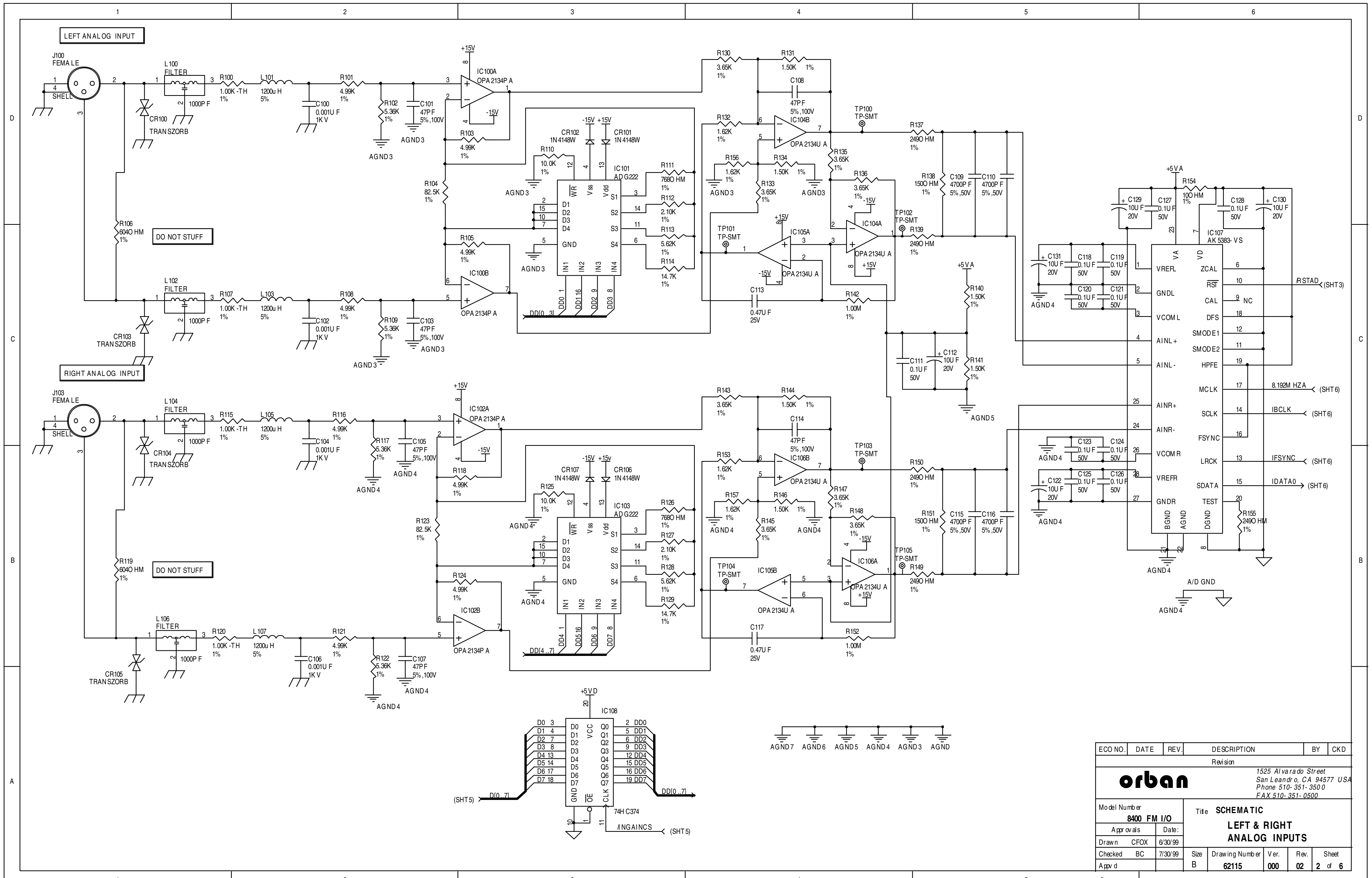

COMPOSITE 2 OUTPUT
COMPOSITE 1 OUTPUT
NOTE: JUMPERS SHOWN AS SHIPPED
HDR 7X2 BOARD STACKER
10/29/99 01 REMOVE R15 & J6 CFOX BC
8/25/99 A PROTO C FOX BC
ECO NO. DATE REV. DESCRIPTION BY CKD
Revision
orban
1525 Alvarado Street
San Leandro, CA 94577 USA
Phone 510-351-3500
FAX 510-351-0500
Model Number 8400
Title SCHEMATIC
IO COMPOSITE
Approvals Date:
Drawn CFOX 8/25/99
Checked BC 8/25/99
Appr'd
Size B
Drawing Number 62120
Ver. 000
Rev. 01
Sheet 1 of 1

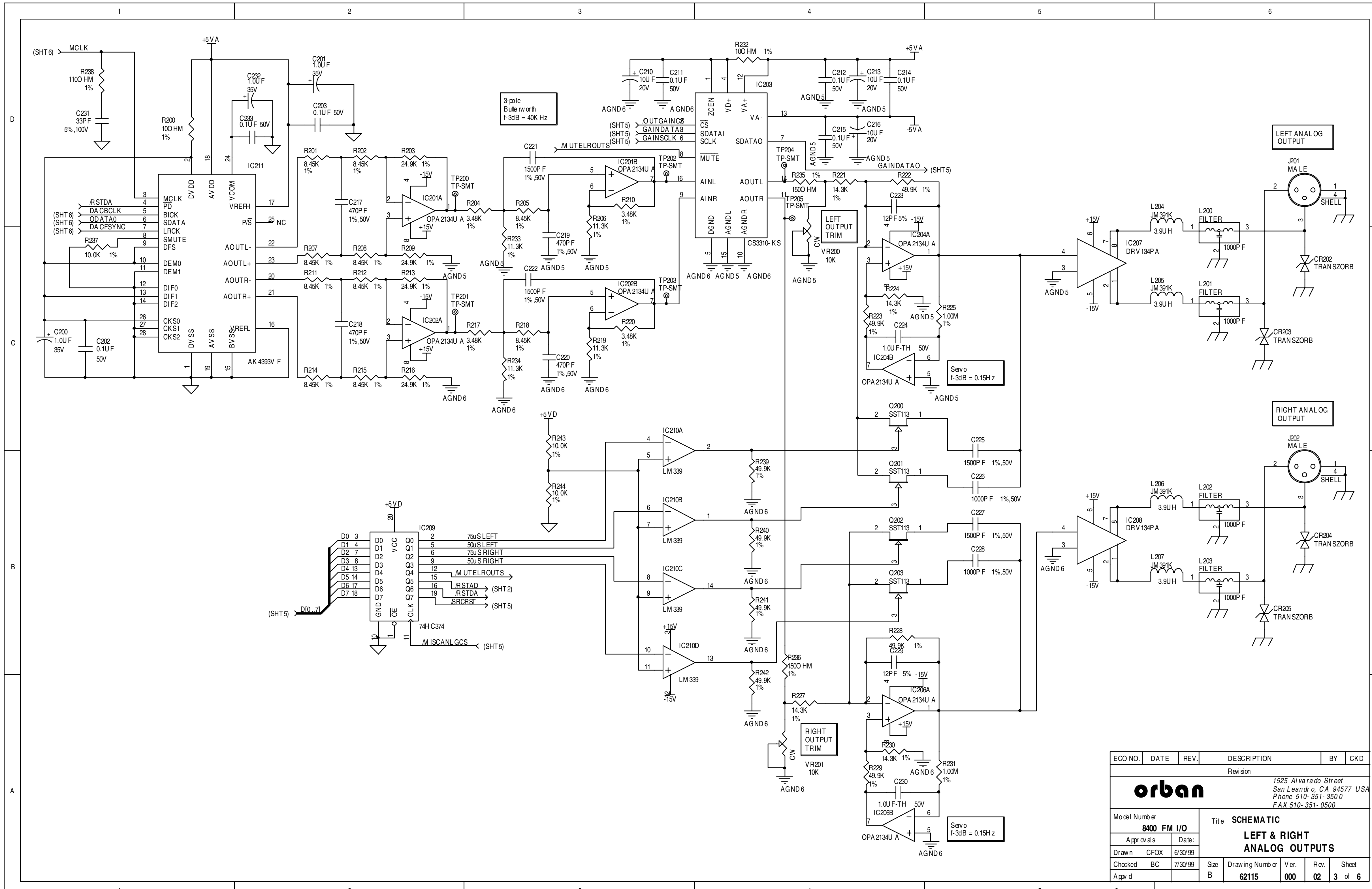


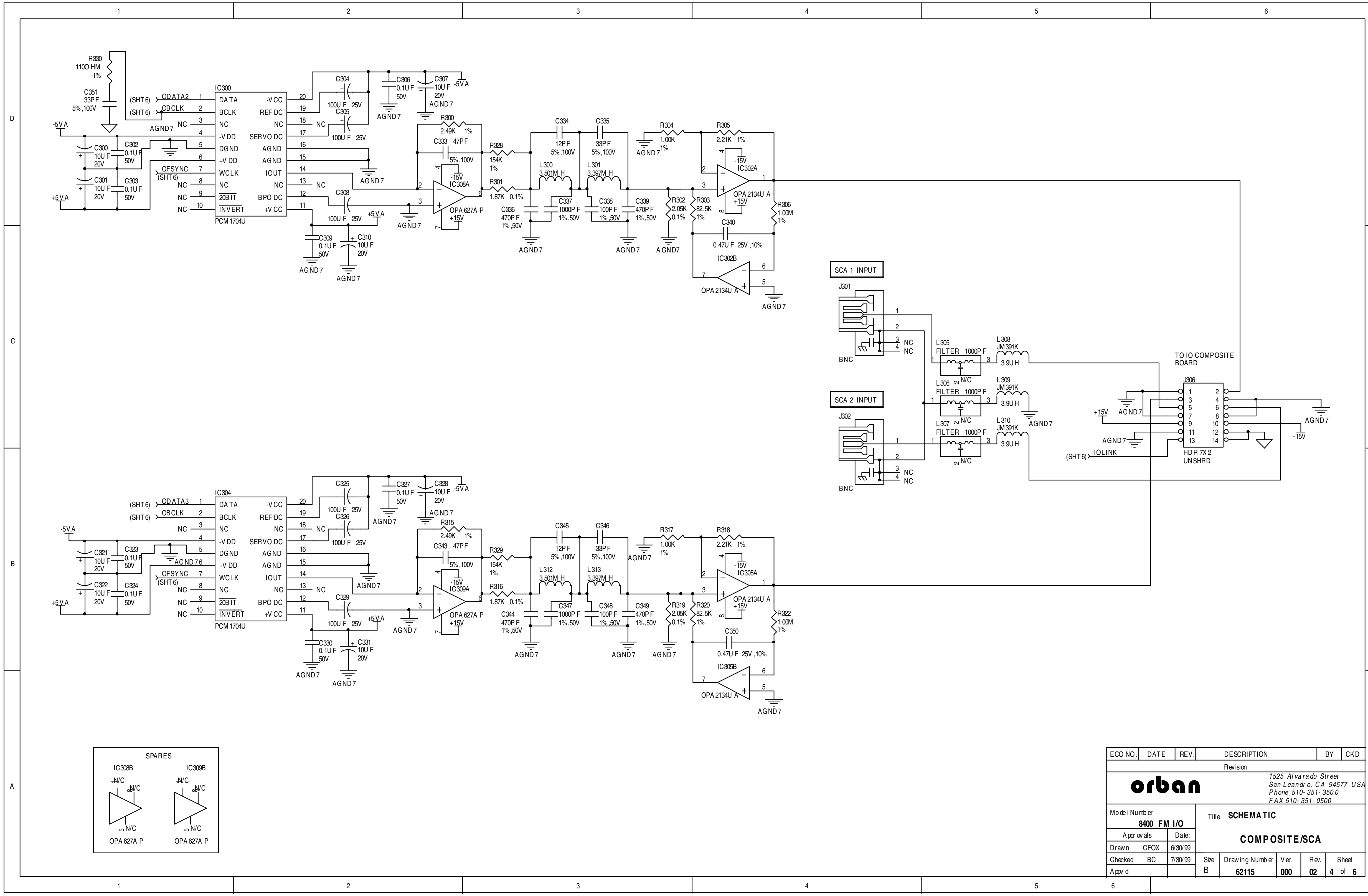
ORBAN®

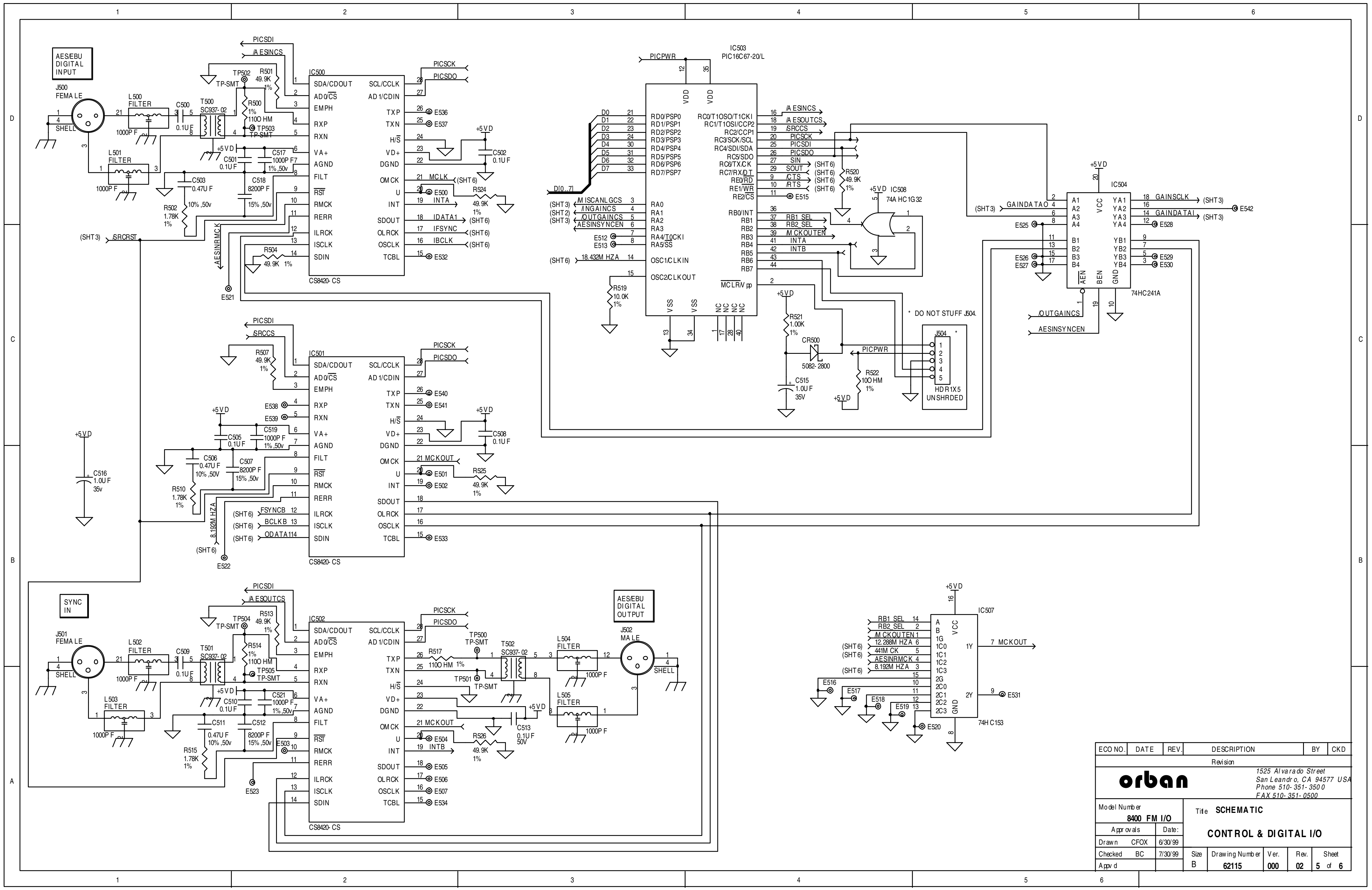
1525 Alvarado Street
San Leandro Street, CA 94577 USA
Phone 1-510-351-3500
Fax 1-510-351-0500

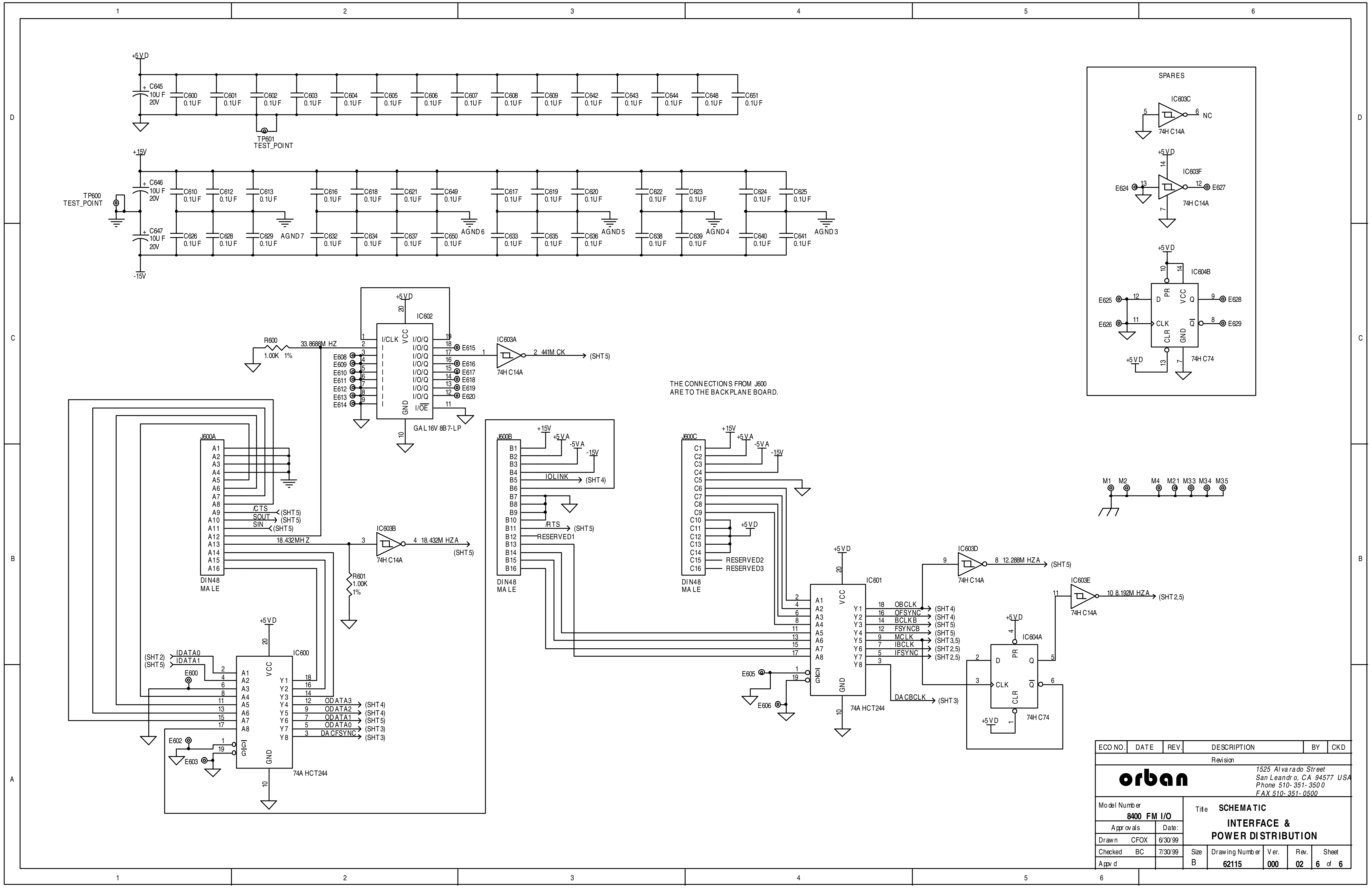
MODEL NO.	8400	TITLE PCB FM I/O				
APPROVALS	DATE					
DWN CFOX	7/28/99					
CKD JT	7/30/99	SIZE	DWG NO.	VER	REV	SHEET
APVR		D	32115	000	02	1 OF 1

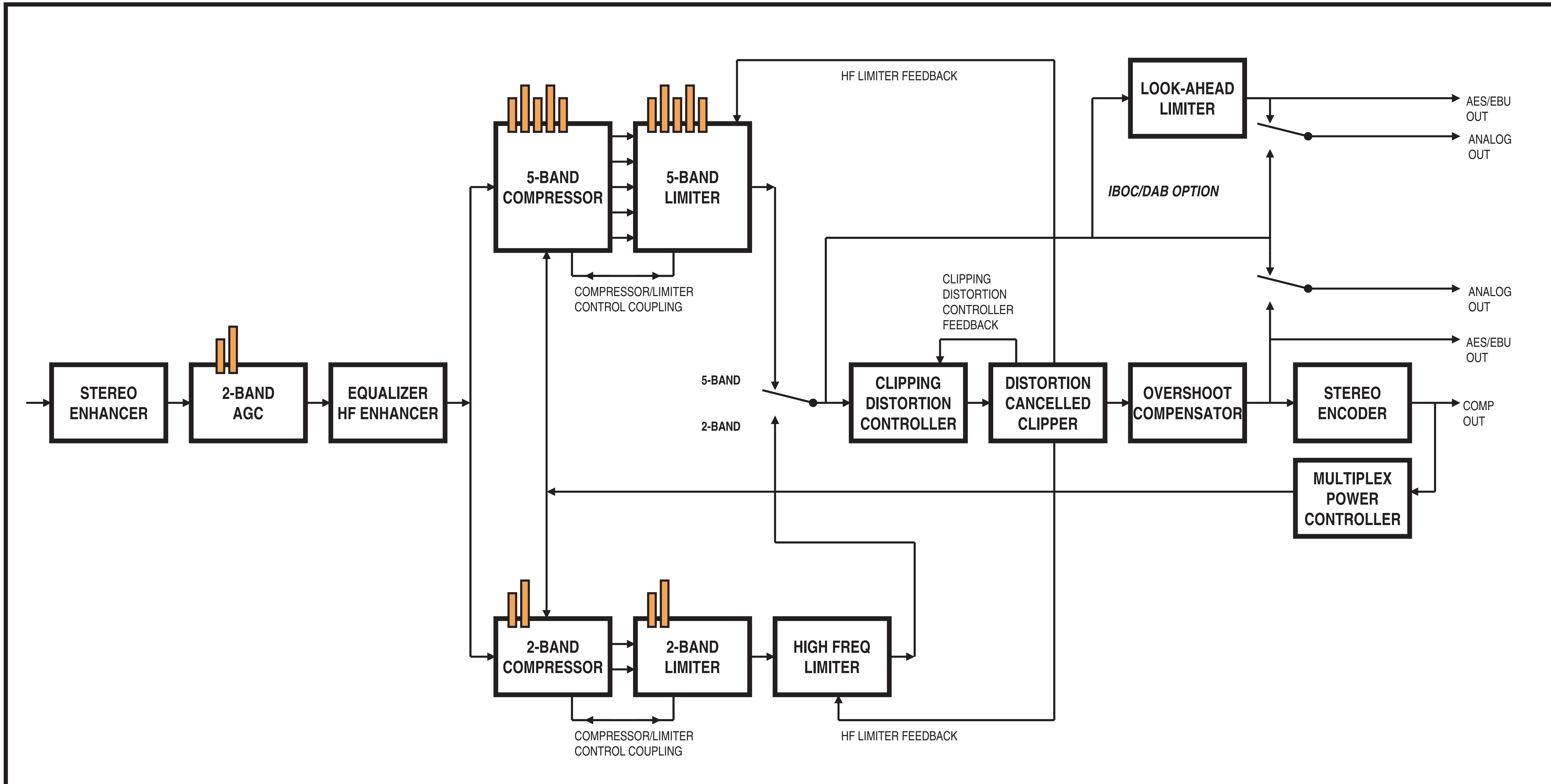

LEFT ANALOG INPUT
RIGHT ANALOG INPUT
DO NOT STUFF
orban
1525 Alvarado Street
San Leandro, CA 94577 USA
Phone 510-351-3500
FAX 510-351-0500
Model Number
8400 FM I/O
Title SCHEMATIC
LEFT & RIGHT
ANALOG INPUTS
Drawn CFOX 6/30/99
Checked BC 7/30/99
Drawing Number 62115
Ver. 000
Rev. 02
Sheet 2 of 6


3-pole
Butterworth
f-3dB = 40K Hz
LEFT ANALOG
OUTPUT
RIGHT ANALOG
OUTPUT
LEFT
OUTPUT
TRIM
RIGHT
OUTPUT
TRIM
Servo
f-3dB = 0.15Hz
AK4393VF
CS3310-KS
74HC374
LM339
OPA2134UA
DRV134PA
J201
MALE
J202
MALE
TRANSZORB
ECO NO. DATE REV. DESCRIPTION BY CKD
Revision
orban
1525 Alvarado Street
San Leandro, CA 94577 USA
Phone 510-351-3500
FAX 510-351-0500
Model Number
8400 FM I/O
Approvals Date:
Drawn CFOX 6/30/99
Checked BC 7/30/99
Appvd
Title SCHEMATIC
LEFT & RIGHT
ANALOG OUTPUTS
Size B
Drawing Number 62115
Ver. 000
Rev. 02
Sheet 3 of 6


SCA 1 INPUT
SCA 2 INPUT
TO IO COMPOSITE BOARD
SPARES
ECO NO. DATE REV. DESCRIPTION BY CKD
Revision
orban
1525 Alvarado Street
San Leandro, CA 94577 USA
Phone 510-351-3500
FAX 510-351-0500
Model Number 8400 FM I/O
Title SCHEMATIC
COMPOSITE/SCA
Approvals Date:
Drawn CFOX 6/30/99
Checked BC 7/30/99
Appv'd
Size B
Drawing Number 62115
Ver. 000
Rev. 02
Sheet 4 of 6


AES/EBU DIGITAL INPUT
SYNC IN
AES/EBU DIGITAL OUTPUT
IC500 CS8420-CS
IC501 CS8420-CS
IC502 CS8420-CS
IC503 PIC16C67-20/L
IC504 74HC241A
IC507 74HC153
IC508 74AHC1G32
* DO NOT STUFF J504.
orban
1525 Alvarado Street
San Leandro, CA 94577 USA
Phone 510-351-3500
FAX 510-351-0500
Model Number 8400 FM I/O
Title SCHEMATIC
CONTROL & DIGITAL I/O
Drawn CFOX 6/30/99
Checked BC 7/30/99
Size B
Drawing Number 62115
Ver. 000
Rev. 02
Sheet 5 of 6


SPARES
IC603C
74HC14A
IC603F
IC604B
74HC74
IC602
GAL16V8B7-LP
R600
33.8688M HZ
IC603A
2.441M CK
(SHT5)
THE CONNECTIONS FROM J600
ARE TO THE BACKPLANE BOARD.
J600A
DIN48
MALE
J600B
J600C
IC603B
18.432MHZ
18.432M HZA
IC600
74AHCT244
IDATA0
IDATA1
ODATA3
ODATA2
ODATA1
ODATA0
DACFSYNC
IOLINK
RTS
RESERVED1
RESERVED2
RESERVED3
IC601
OBCLK
OFSYNC
BCLKB
FSYNCB
MCLK
IBCLK
IFSYNC
DACBCLK
IC603D
12.288M HZA
IC603E
8.192M HZA
IC604A
TP600
TEST_POINT
TP601
TEST_POINT
AGND7
AGND6
AGND5
AGND4
AGND3
ECO NO.
DATE
REV.
DESCRIPTION
BY
CKD
Revision
orban
1525 Alvarado Street
San Leandro, CA 94577 USA
Phone 510-351-3500
FAX 510-351-0500
Model Number
8400 FM I/O
Title SCHEMATIC
INTERFACE &
POWER DISTRIBUTION
Approvals
Date:
Drawn CFOX 6/30/99
Checked BC 7/30/99
Appv'd
Size B
Drawing Number 62115
Ver. 000
Rev. 02
Sheet 6 of 6


HF LIMITER FEEDBACK
LOOK-AHEAD LIMITER
AES/EBU OUT
ANALOG OUT
5-BAND COMPRESSOR
5-BAND LIMITER
IBOC/DAB OPTION
COMPRESSOR/LIMITER CONTROL COUPLING
CLIPPING DISTORTION CONTROLLER FEEDBACK
ANALOG OUT
AES/EBU OUT
5-BAND
2-BAND
STEREO ENHANCER
2-BAND AGC
EQUALIZER HF ENHANCER
CLIPPING DISTORTION CONTROLLER
DISTORTION CANCELLED CLIPPER
OVERSHOOT COMPENSATOR
STEREO ENCODER
COMP OUT
MULTIPLEX POWER CONTROLLER
2-BAND COMPRESSOR
2-BAND LIMITER
HIGH FREQ LIMITER
COMPRESSOR/LIMITER CONTROL COUPLING
HF LIMITER FEEDBACK