Operating Manual

OPTIMOD 6200/6200S

Digital Audio Processor

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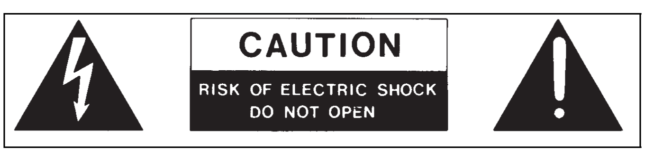
IMPORTANT NOTE: Refer to the unit's rear panel for your Model #.

Model Number:	Description:
6200/U	OPTIMOD-DAB 6200, Stereo Encoder, Digital I/O Protection Structure, Two-Band Structure set to 115V (for 90-130V operation)
6200/E	OPTIMOD-DAB 6200, Stereo Encoder, Digital I/O Protection Structure, Two-Band Structure set to 230V (for 200-250V operation)
6200/S	OPTIMOD 6200S, Stereo Encoder, Digital I/O Protection Structure, Two-Band Structure (for 90-250V operation)

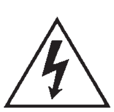
MANUAL:

Part Number:	Description:
96104-000-04	6200 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 THIS FIRST!

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 6200/6200S

Digital Audio Processor

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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 or damaging static.

The OPTIMOD 6200 is protected by U.S. patents 4,208,548; 4,460,871; and U.K. patent 2,001,495. Other patents pending.

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1525 ALVARADO STREET, SAN LEANDRO, CA 94577 USA
Phone: (1) 510/351-3500; Fax: (1) 510/351-0500; E-Mail: custserv@orban.com; Site: www.orban.com
P/N: 96104.000.04

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Section 1
Introduction

Reading this Manual

This manual supports both 6200 OPTIMOD-DAB and 6200S OPTIMOD digital audio processors. In general, when text refers to 6200 OPTIMOD-DAB, the information also applies to 6200S. Whenever there is a difference, it is noted. At the time of this writing, 6200S differs from 6200 in size and a few specifications (noted in Section 6), and has no rear panel remote control interface connector.

6200 OPTIMOD-DAB Digital Audio Processor

Orban's all-digital 6200 OPTIMOD-DAB Audio Processor can help you achieve the highest possible quality digital audio broadcast processing. Because all processing is performed by high-speed mathematical calculations within Motorola DSP56009 Digital Signal Processing chips, the processing has cleanliness, quality, and stability over time and temperature that is unmatched by analog processors.

OPTIMOD-DAB is descended from the industry-standard OPTIMOD audio processors for radio and television. Thousands of these broadcast-specific processors are on the air all over the world. They have proven that the "OPTIMOD sound" can attract and keep an audience even in the most competitive commercial environment.

Because OPTIMOD-DAB 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-DAB. A small investment of your time now will yield large dividends in audio quality.

The rest of Chapter 1 explains how OPTIMOD-DAB fits into the DAB and DTV broadcast plants, or can be used for webcasting. Chapter 2 explains how to install it. Chapter 3 tells how to properly operate OPTIMOD-DAB. Chapter 4 through Chapter 6 provides reference information.

OPTIMOD-DAB 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-DAB 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-DAB, which is then defeated.

User-Friendly Interface

- A liquid crystal display (LCD) makes setup, adjustment and programming of OPTIMOD-DAB easy. Dynamically labeled soft keys provide a context-sensitive interface. Dedicated LEDs show all metering functions of the processing structure in use.

- Push one of the dedicated buttons to “Recall” a preset, to “Modify” processing, or to access the system’s Setup controls.

Flexible Configuration

- OPTIMOD-DAB **precisely controls peak levels** to prevent overmodulation. The maximum level of the digital samples is controlled to better than 1%.

- The OPTIMOD-DAB 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 32kHz, 44.1kHz, and 48kHz sample rates.

- The analog inputs are transformerless, balanced 10kΩ instrumentation-amplifier circuits, and the analog monitor outputs are transformerless balanced, and floating with 30Ω impedance to ensure highest transparency and accurate pulse response.

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

- OPTIMOD-DAB **controls the audio bandwidth** as necessary to accommodate the transmitted sample frequency. OPTIMOD-DAB’s high frequency bandwidth can be switched instantly (typically in 500Hz increments) between 4.0kHz and 20kHz. 20kHz is used for highest-quality systems. 15kHz meets the requirements of the proposed AM IBOC systems that use 32kHz sample frequency. 10kHz bandwidth meets the requirements of auxiliary Eureka-147 transmissions using a 24kHz sample frequency. Lower audio bandwidths meet the requirements of auxiliary speech grade services and the proposed digital shortwave systems.

- OPTIMOD-DAB is equipped with a second serial port to interface to a DAB transmitter. This serial port is intended to **support PAD (Program-Associated Data) services**. (This support is not available with Firmware Version 2.0 or lower.)

- OPTIMOD-DAB 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 active at a time. Just as there are many possible ways of configuring a processing system using analog components (such as equalizers, compressors, limiters, and clippers), there are many possible processing structures achievable by OPTIMOD-DAB. OPTIMOD-DAB realizes its processing structures as a series of high-speed mathematical computations made by Digital Signal Processing (DSP) chips.

- OPTIMOD-DAB features three processing structures: **Five-Band** (or Multi-Band) for a consistent, "processed" sound, free from undesirable side effects; **Two-Band** for a tastefully controlled sound that preserves the frequency balance of the original program material; and **Protect**, which provides up to 25dB of safety limiting with minimal side effects.

- OPTIMOD-DAB **can increase the density and loudness** of the program material by multi-band compression and look-ahead limiting — improving the consistency of the station's sound and increasing loudness and definition remarkably, without producing unpleasant side effects.

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

- OPTIMOD-DAB's processing structures are all **phase-linear** to maximize audible transparency.

- OPTIMOD-DAB can be changed from one processing structure to another with a smooth cross-fade.

- OPTIMOD-DAB's Two-Band structure includes the CBS Loudness Controller for DTV applications. This system measures subjective loudness (as perceived by an average listener) and then closes a feedback loop to limit loudness to a preset level. It effectively controls loud commercials, which are the primary irritant in broadcast television audio.

Controllable

- All compression, limiting, and clipping can be defeated by remote control to permit broadcast system test and alignment or "proof of performance" tests.

- Both 6200 and 6200S can be remote-controlled by an external computer running Orban remote software (included) and connected directly or via modem to one of OPTIMOD's RS-232 serial ports. 6200 OPTIMOD-DAB (and not Model 6200S) can also be remote-controlled by 5-12V pulses applied to eight programmable, optically isolated ports.

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

Presets in OPTIMOD-DAB

There are two distinct kinds of presets in OPTIMOD-DAB: Factory Processing Presets and User Presets.

Factory Processing Presets

There are 36 Factory Processing Presets — 26 for radio and 10 for television. These are our “factory recommended settings” for various program formats or types. The description indicates the processing structure and the type of processing. Each Factory Processing Preset on the Recall Preset list is really a library of 20 separate presets, selected by pressing the *Modify* button and using the LESS-MORE control to adjust OPTIMOD-DAB for more or less processing.

Factory Processing Presets are stored in OPTIMOD-DAB’s non-volatile memory and cannot be erased. You can change the settings of a Factory Processing 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 Processing Preset to suit your requirements, and store those changes.

You may store up to 32 User Presets. They are indicated on the Recall Preset list by a number designation from 01 to 32, followed by a description.

You may enter in any description you wish, up to 16 characters. User Presets cannot be created from “scratch.” Start by recalling a Factory Preset. You can then immediately store this in a User Preset, give it whatever name you wish, then make changes to the settings. Or you can recall a Factory Preset, make the changes first, 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 backed up in EEPROM, which is a form of rewritable memory that does not require battery backup. So your presets are safer than they would be if the memory required battery backup.

You can also modify an existing User Preset.

Input/Output Configuration

OPTIMOD-DAB is designed to simultaneously accommodate:

- Digital AES/EBU left/right inputs and outputs.
- Analog left/right inputs and outputs.

Digital AES/EBU Left/Right Input/Output

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

> To ensure best control of peak modulation, operate the output at 48kHz. This guarantees that the output samples are synchronous with the peak-controlled samples produced by the processing.

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. A second digital input is available to genlock the 6200's output sample frequency to house sync, if required.

OPTIMOD-DAB is designed to simultaneously accommodate digital and analog inputs and outputs. You select whether OPTIMOD-DAB uses the digital or analog input on the System Setup I/O Calib 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 System Setup.

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 +27dBu (0dBu = 0.775Vrms).

The left and right analog outputs are on XLR-type male connectors on the rear panel. Output impedance is 30Ω; balanced and floating.

The outputs are intended for monitoring. Their output level is fixed at +14dBu full scale. They can drive 600Ω or higher impedances.

Level control of the analog inputs is via software control through System Setup. (See step 2 on page 2-24 and step 3 on page 2-26.)

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 two fundamentally different ways. They can either pass unprocessed audio for application to the 6200's input, or they can pass the 6200's peak-controlled output. The two applications have fundamentally different performance requirements. In general, a link that passes unprocessed audio should have very low noise and low nonlinear distortion, but its transient response is not important. A link that passes processed audio doesn't need as low a noise floor as a link passing unprocessed audio. However, its transient response is critical. In DAB applications such a link *must* be uncompressed digital and *must* use digital inputs and outputs to achieve 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 6200 to the transmitter. For example, the MPEG Layer 2 algorithm can increase peak levels up to 4dB 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 6200'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 6200 uses multi-band 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 6200's multi-band 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 and they do not require downward sample rate conversion.

> Downward sample rate conversion can cause overshoot due to spectral truncation and asynchronous re-sampling of the 48kHz peak-controlled samples.

If the link does not have an AES/EBU input, you must drive its analog input from the 6200's monitor output. This is not recommended because the 6200's monitor output will overshoot in the analog domain because of the physics of the system.

> Peak control in the 6200 occurs at a 48kHz sample frequency. This is sufficient to prevent any samples from exceeding the threshold of limiting. However, after reconstruction, the analog output may overshoot the nominal 100% level because these overshoots "fall between the samples," so the processing cannot be aware of them. If you use this output to feed the analog input of a digital STL, the new samples in the STL will not be synchronous with the samples inside the 6200. Therefore, they may well fall on the overshoots, causing loss of peak modulation control. It is therefore very important to use a link with an AES/EBU input to ensure correct peak control.
>
> The same sort of thing can happen if you use the output sample rate converter, because the output samples are no longer synchronous with the peak-controlled samples in the processing. Always use 48kHz output sample rate to achieve best peak control.
>
> If you *must* use an analog input, you may bypass any anti-aliasing filters in digital links driven by the 6200 because the 6200's output spectrum is tightly controlled. This ensures the most accurate possible transient response, given the limitations of asynchronous sampling described above.

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 6200's AES/EBU digital output, which will provide the most accurate interface to the STL. The digital input and output accommodate sample rates of 32kHz, 44.1kHz, and 48kHz.

Microwave STLs

In general, an analog microwave STL provides high 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. However, the noise and non-linear distortion characteristics of such links are likely to be notably poorer than 16-bit digital even if propagation conditions are ideal.

As discussed above, asynchronous sampling problems will cause overshoots if *any* analog path (even a perfectly transparent one) passes the 6200's processed output to the transmitter. Lack of transparency in the analog path will cause even more overshoot. Unless carefully designed, 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 6200 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 6200 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 6200 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 6200.

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

Further, it is not unusual for a microwave STL to bounce because of a large infrasonic peak in its frequency response caused by an under-damped automatic frequency control (AFC) phase-locked loop. This bounce can increase the STL's peak carrier deviation by as much as 2dB, reducing average modulation. Many commercial STLs have this problem.

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

Analog landline (PTT/post office line)

Analog landline quality is extremely variable, ranging from excellent to poor. Whether landlines should be used or not depends upon the quality of the lines locally available, and upon the availability of other alternatives. 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 DAB 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.

Location of OPTIMOD-DAB

At the Transmitter is Best

The best location for OPTIMOD-DAB is as close as possible to the transmitter so that OPTIMOD-DAB's AES/EBU output can be connected to the transmitter through a circuit path that introduces no change in OPTIMOD-DAB's output bitstream. A high-quality AES/EBU cable is ideal.

Where Access to the Transmitter Plant is Not Possible

Sometimes it is not possible to locate OPTIMOD-DAB at the transmitter. Instead, it must be located on the studio side of the link connecting the audio plant to the transmitter. If the transmitter plant is not accessible, all audio processing must be done at the studio, and you must tolerate any damage that occurs later.

If an uncompressed digital link is available, this is an ideal situation because such a link will pass OPTIMOD-DAB's output with little or no degradation. However, such a link is not always available.

> If only a 32kHz sample rate link is available, the sample rate conversion necessary to downsample the audio will cause overshoots when the 6200 is operated at 20kHz bandwidth because the sample rate converter removes spectral energy. In this case you can minimize overshoot by operating the 6200 at 15kHz bandwidth.

Unless the path is a digital path using no lossy compression, this situation will yield lower performance than if OPTIMOD-DAB is connected directly to the transmitter, because artifacts that cannot be controlled by OPTIMOD-DAB will be introduced by the link to the transmitter. These artifacts can result in 2-4dB lower average modulation level, and can also add noise and audible non-linear distortion. In the case of lossy digital compression this deterioration will be directly related to the bit rate. In the case of an analog path, the deterioration will depend on the amount of linear and non-linear distortion in the path. In addition, there will be an unavoidable amount of overshoot caused by asynchronous re-sampling (see page 1-8).

One strategy is to apply the same lossy compression to OPTIMOD-DAB's output signal that the DAB transmitter would apply. If a digital link is available with sufficient bit rate to pass this compressed signal, it can then be passed directly to the DAB transmitter without further processing if synchronization issues can be resolved. Consult with the manufacturer of your DAB transmitter to see if this can be done.

Where only an analog or lossy digital link is available, feed the audio output of OPTIMOD-DAB directly into the link. If available, the transmitter's protection limiter should be adjusted so that audio is normally just below the threshold of limiting: The transmitter protection limiter should respond only to signals caused by faults or by spurious peaks introduced by imperfections in the link.

Where maximum quality is desired, it is wise to request that all equipment in the signal path after the studio be carefully measured and aligned and qualified to meet the appropriate standards for bandwidth, distortion group delay and gain stability. Such equipment should be measured at reasonable intervals.

OPTIMOD-DAB at the Transmitter: Gain Control before the STL

The audio received at OPTIMOD-DAB's input should have the highest possible quality. To achieve the full audible benefit of OPTIMOD-DAB processing, use a studio-transmitter link (STL) that is as flat as the bandwidth of OPTIMOD-DAB as used in your plant (usually 20kHz). Ideally, you should use a 20-bit (or better) uncompressed digital link with at least 44.1kHz sample frequency.

Because the audio processor controls peaks, it is not important that the audio link feeding OPTIMOD-DAB's input terminals be phase-linear. However, the link should have low noise, the flattest possible frequency response from 20-20,000Hz, and low nonlinear distortion.

If the audio link between the studio and the transmitter is noisy (or, if digital, is limited to 16 bits or less), performing the AGC function at the studio site can minimize the audibility of this noise. AGC applied before the audio link improves the signal-to-noise ratio because the average level on the link will be greater. Further, many STLs require level control to prevent the STL from being overloaded.

To apply such level control and compression, we recommend the Orban Model 8200ST Compressor/Limiter/HF Limiter/Clipper before the STL transmitter. The 8200ST performs the function of OPTIMOD-DAB's internal broadband automatic gain control (AGC), while simultaneously protecting the STL. If this is done, defeat OPTIMOD-DAB's broadband AGC by accessing the ST CHASSIS function within the Setup menu and setting it to yes.

In DTV applications, the AGC function in Two-Band presets is normally off. We do not recommend using additional AGC ahead of the STL in this application. Instead, either align STL operating levels to allow sufficient headroom to pass unprocessed audio, or use an Orban Transmission Limiter (which is ordinarily operated below threshold) to protect the link from operator error.

Using Lossy Data Reduction in the Studio

Many stations are now using lossy data reduction algorithms like MPEG-1 Layer 2 or Dolby AC2 to increase the storage time of digital playback media. In addition, source material is often supplied through a lossy data reduction algorithm, whether from satellite or over landlines.

Sometimes, several encode/decode cycles will be cascaded before the material is finally presented to OPTIMOD-DAB'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, there is at least one other mechanism that can cause the noise to become audible at the radio. OPTIMOD-DAB's multi-band 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.

Interfacing to the Transmitter

Sync Input

In the Eureka-147 system several programs are combined into one "ensemble multiplex." This requires synchronization of the sample rates applied to the transmitter. DTV also requires synchronization. OPTIMOD-DAB provides a second AES/EBU input to accept "house sync," which allows OPTIMOD-DAB's output to be synchronized to a master sync generator. Regardless of whether its analog or digital inputs are used, its AES/EBU output will be synchronized to the AES/EBU signal at its SYNC INPUT. Because OPTIMOD-DAB's digital input is equipped with a sample rate converter, the SYNC INPUT allows an asynchronous digital input to be applied to OPTIMOD-DAB while ensuring that OPTIMOD-DAB's output is in sync with the master sync generator.

If there is no signal present at the SYNC input, the 6200 can still sync its output to the signal present at the AES/EBU input.

Sample Rate and Audio Bandwidth

Most DAB audio is at a 48kHz sample rate. However, several of the proposed AM IBOC systems operate at 32kHz, requiring 15kHz audio bandwidth. The Eureka-147 system offers a 24kHz sample rate option, requiring 10kHz audio bandwidth. The proposed shortwave systems require audio bandwidths as low as 4.5kHz for speech-grade services.

OPTIMOD-DAB's bandwidth can be adjusted from 4kHz to 20kHz to provide correctly anti-aliased audio for any of these systems. Provided that any anti-aliasing filters following OPTIMOD-DAB's output are phase-linear and that they have integer-sample time delays, these will pass the band-limited OPTIMOD-DAB output without introducing overshoot because they remove no further spectrum and do not cause their output samples to become asynchronous with the peak-controlled samples at the 6200's output.

OPTIMOD-DAB always operates at 48kHz sample rate internally. Its output is equipped with a sample rate converter that can output at 32kHz, 44.1kHz, or 48kHz. These rates can be synchronized to the AES/EBU SYNC INPUT.

We expect that transmitters that transmit sample rates below 32kHz will provide internal sample rate conversion, and that most will probably accept audio at 48kHz sample rate regardless of the final sample rate of the transmission. Any sample rate conversion may cause the transmitted sample to become asynchronous to the peak-controlled samples emerging from the 6200 and may therefore introduce overshoot. Fortunately, as the audio bandwidth becomes lower this becomes less of the problem because the 48kHz sample rate within the 6200 oversamples the audio and therefore becomes less likely to let peaks "slip between the samples."

Subframe Delay

OPTIMOD-DAB provides an adjustable time delay of up to 96 milliseconds. This allows the installer to force the total delay through the processing to equal one frame. The definition of "frame" depends on the system in which the 6200 is installed.

> The selections are off (approximately 14ms delay), 24 milliseconds, 30 fps, 29.97 fps (NTSC color video), 25 fps (most PAL video), 24 fps (film), and 96 milliseconds.

Program-Associated Data (PAD)

OPTIMOD-DAB provides two serial ports. One is designed for use with a PC-based remote control. The other is designed to interface to a DAB transmitter to provide Program-Associated Data (particularly the Dynamic Range Control signal).

> Version 2.0 (and lower) of the 6200 firmware does not yet support PAD because Orban is still working with transmitter vendors and customers to define exactly what is required and desired.

Setting Modulation Levels

In a perfect world, one could set the peak modulation level at the 6200's output to 0dBfs. However, there are at least two potential problems that may make it desirable to set the modulation level slightly lower.

First is asynchronous re-sampling, which we have discussed at length earlier in this chapter. (See page 1-8, for example.) If any digital processing that causes its output samples to be asynchronous to its input samples is used after the 6200's output, this can cause the peak levels of individual samples to increase above the nominal threshold of limiting. This increase is typically less than 0.5dB.

Second is headroom in lossy data compression systems. A well-designed perceptual encoder will accept samples up to 0dBfs and will have internal headroom sufficient to avoid clipping. However, there is no guarantee that *receiver* manufacturers will implement perceptual decoders with sufficient headroom to avoid clipping overshoots. Such overshoots are the inevitable side effect of increasing the quantization noise in the channel, and can be as large as 3-4dB. Most perceptual encoder algorithms are designed to have unity gain from input to output. So if peak levels at the input frequently come up to 0dBfs, peak levels at the output will frequently exceed 0dBfs (and will be clipped) unless the decoder algorithm is adjusted to be less than unity gain.

The canny station engineer will therefore familiarize him/herself with the performance of real-world receivers and will reduce the peak modulation of the transmissions if it turns out that most receivers are clipping due to perceptual encoding overshoots.

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 6200 can be as much as 14ms (or more, if the installer purposely adds frame-makeup delay). This delay, although unlikely to be audible as a distinct echo, can cause bone conduction comb filtering of the DJ/presenter's voice in his/her ears. This is almost always very uncomfortable to them.

OPTIMOD-DAB's Monitor Output can be switched so that it is driven after the multiband compressor but before the look-ahead peak limiter, which is where the majority of the delay occurs. When driven by the multi-band compressor alone, the input/output delay is approximately 3-4ms (depending on whether the analog or digital input is used and whether sample rate conversion is used). This delay can still be uncomfortable to some, but many DJ/presenters find it acceptable.

Such problems can be completely avoided if the DJ/presenter's headphones are driven directly from the program line or, better, by an inexpensive compressor connected to the program line. If the DJ/presenter 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. Such alarms could be configured to cut off audio to the DJ/presenter's phones when an audio or carrier failure occurs.

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:

> **Note:** Normal 6200 processing may not allow the full modulation level as required by EAS standards. It is therefore necessary to temporarily defeat the 6200's processing during the broadcast of EAS tones and data. Placing the 6200 in Bypass mode can defeat the processing. The BYPASS GAIN control allows a fixed gain trim through the 6200. See "Test Modes," on page 3-35 for more information.

1. Place the 6200 in Bypass mode locally.

A) Press *Setup* button.

B) Press TEST soft key.

C) Hold down the MODE soft key, turn the control knob to display test: bypass, then release the soft key.

D) Begin EAS broadcast.

After the EAS broadcast, resume normal processing:

E) Press *Setup* button.

F) Press TEST soft key.

G) Hold down the MODE soft key, turn the control knob to display operate, then release the soft key. This will restore the processing preset in use prior to the test mode.

Alternately, you may press *Recall* button to exit bypass test.

2. Place the 6200 in Bypass mode by remote control. Then program any two Remote Interface inputs for "test: bypass" and "exit test," respectively. (OPTIMOD-DAB 6200 only)

A) Press *Setup* button.

B) Press REMOTE soft key.

C) Press Remote Interface soft key.

D) Select the desired Remote Interface input (1–8), using *Next* and *Prev* buttons to display additional pages.

E) Hold down either soft key below the desired Remote Interface input and turn the control knob to display test: bypass, then release the soft key.

F) With a different Remote Interface input hold down either soft key below the desired input and turn the control knob to display exit test, then release the soft key.

G) Connect two outputs from your station remote control system to the Remote Interface connector on the rear panel of the 6200, according to the wiring diagram in Section 2.

H) Place the 6200 in Bypass mode by remote control.

a) Switch the 6200 into Bypass mode by a momentary command from your station's remote control to the input programmed as test: bypass.

b) Begin EAS broadcast.

c) When the EAS broadcast is finished, switch the 6200 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 for the duration of the EAS broadcast.

3. Place the 6200 in Bypass mode by PC remote control.

Refer to 6200PC document for specific steps.

Security Pascode for PC Control

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

A front panel soft key, labeled END PC CONTROL can be pressed to end remote control of the 6200; this feature effectively prevents simultaneous remote and local control.

See 6200PC document for more detail.

DTV Applications

The following discussion is based on the best information available in early 2000, and is likely to be subject to change as the industry gets more experience with DTV broadcasts.

Using the 6200 in the United States DTV System

The United States DTV system specifies the Dolby AC3 lossy audio compression system as the standard for transmitting anywhere from one to six channels of digital audio. Part of the AC3 bitstream is "metadata" — data about the data. There are three important pieces of metadata in the AC3 bitstream.

- The first is Dialog Normalization, which, in essence, sets the receiver's volume control to complement the dynamic range of the program material being transmitted.

- The second is Line-Mode Dynamic Range Control, which allows the receiver to perform a wideband compression function if the listener chooses.
- The third is RF-mode Dynamic Range Control, which applies heavier processing.

The obvious question that arises is how these signals are to be generated in a real-world operational facility. And, indeed, in which situations they *should* be generated.

We should remember that the marketing landscape is littered with "features" that seemed to be a good idea at the time, but which proved to be of little or no interest to consumers. Digital technology has vastly decreased the cost of adding new features to consumer electronics, and many consumer manufacturers have responded with a blizzard of features that are confusing, hard-to-understand, or just plain useless.

For example, CDs have always offered the ability to deliver auxiliary data. According to the original CD hype, you would see the lyrics of the songs scroll by as you played them. In addition, you would see still pictures of the band members by connecting your CD player to your television set. Where are these features now? The answer, of course, is that the public did not find them compelling enough to justify the additional production expense to add them to the CD data stream, or to justify the increase in manufacturing cost necessary to add the video outputs or the LCD screens to the CD players.

Another example is the SAP channel in BTSC stereo television. Very few viewers understand it, yet a number of them manage to turn it on by accident. Then they can't understand why the sound becomes low-fidelity mono, and why everyone is suddenly speaking Spanish! Consequently, many consumer manufacturers buried the SAP control very deep in the menu structure of receivers or VCRs to prevent this confusion from occurring in the future.

Concerning the AC3 metadata, we believe that only a small minority of viewers will ever understand the concept of dynamic range control. Dolby Laboratories wisely specified that dynamic range compression would be the receiver default, because they realized that most consumers would never want full dynamic range audio.

Experience has shown that a vast majority of viewers are not interested in wide dynamic range. Instead, they want two things. First, dialog should be comfortably intelligible, and second, commercials should not be irritatingly loud by comparison to program material.

Home theater owners may want the opportunity to watch feature films while hearing a wide dynamic range signal. However, even these viewers usually consume television in a much more passive way when viewing garden-variety programs. If television is to be an acceptable part of the domestic environment, the sound cannot overwhelm household members not interested in viewing (not to mention neighbors, particularly in multi-family dwellings). For a variety of reasons, the dynamic range of sound essential to the intelligibility of the program should not exceed 15dB in a domestic listening environment. Underscoring and ambient sound effects will, of course, be lower than this.

The issue of loud commercials is particularly important — the FCC has been concerned with loud commercials ever since the mid 1960's, and has twice actively investigated the problem since then as a result of viewer complaints. It is against FCC rules to broadcast irritatingly loud commercials.

Audio Processing for Consistency and Loudness Control

In current NTSC practice, all audio is applied to a transmission audio processor that automatically controls the average modulation and the peak-to-average ratio. This ensures that the audio will be comfortably listenable. The audio processor also has another crucial function — it smooths out transitions between one piece of program material and the next. Orban's OPTIMOD-TV 8282 (for analog services) incorporates the CBS Loudness Controller algorithm. This uses a complex algorithm that estimates the amount of perceived loudness in a given piece of program material. If the loudness exceeds a preset threshold, the controller automatically reduces it to that threshold. The main purpose of this circuit is to control the loudness of commercials that have been processed to produce irritating loudness without such control. The 6200's Two-Band structure also incorporates the CBS Loudness Controller algorithm, bringing automatic loudness control to DTV audio.

A Hybrid Technique for DTV Processing

Knowing how broadcasters do successful processing in the analog world, we believe that the most realistic approach to handling AC3 dialog normalization is a hybrid technique. Most program material can be passed through an audio processor with a loudness controller very much like the ones currently used for analog television. This material is typically either mono or two-channel stereo. It includes commercials, live news, game shows, talk shows, soap operas; and many documentaries, sports, and pop music videos and concerts. Processors used in analog TV control their maximum loudness level very well, so a single dialog normalization value will apply to all program material whenever the processor is online. The advantage of this strategy is that the processor will guarantee that all of this material is comfortably listenable, and that commercials are not excessively loud. With the possible exception of sports, this program material does not rely on extreme dynamic range to make its point, so we do not believe that compression damages the artistic integrity of this programming. No one needs more dynamic range on the local news!

Prime time dramatic shows, newer feature films, classical music concerts, and certain sports broadcasts all use dynamic range for dramatic impact, and therefore are candidates for full-blown exploitation of the AC3 metadata. Each show, film, and concert must have a dialog normalization value pre-assigned to it, ideally derived by referring to a calibrated loudness meter. The uncompressed audio is then applied to the AC3 encoder, along with Line-Mode and RF-Mode Dynamic Range Control signals to ensure that the receiver can apply compression if the viewer prefers a narrower dynamic range.

Unless they have been produced in 5.1-channel, commercials should be processed through the 6200 in the usual way. If the fixed dialog normalization value is correctly chosen for all material passed through the audio processor, commercials will automatically be limited in loudness to the average loudness of the dialog and will therefore be unobtrusive regardless of whether the listener is hearing compressed or uncompressed audio.

We suspect that it is impractical to pass through, without review, dialog normalization values created by program and commercial providers, because some commercial providers will inevitably try to game the system to make their commercials excessively loud. Instead, if dialog normalization is to be actively used in transmission, the broadcaster must strip its existing value from the program, and must then preview each piece of program material and replace the value with one that will ensure consistency from one piece of program material to the next. We think that very few local stations will want to devote the necessary resources to this activity. Instead, it's an obvious thing for the networks to do.

If the networks have done their job well, they will choose dialog normalization values that ensure consistency from source to source, and when the viewer changes channels. It is improbable that this can be done by automation. The best we will be able to do is to manually identify dialog or other baseline sounds, measure their loudness with a true loudness meter, and manually adjust the dialog normalization parameter so that these baseline sounds emerge with a standardized loudness. CBS's research into this area showed that no simple meter could do this accurately, including frequency-weighted meters with averaging characteristics. The errors in such measurements were so large that they were not useful in controlling the levels of commercials well enough to eliminate viewer complaints.

The CBS loudness meter divides the signal into seven octave bands and weights the gains of the bands according to the 70-phon equal-loudness curve of the ear. It then averages the output of each band with a 15-millisecond time constant. The averaged outputs of the bands are then added and the sum is applied to a 200-millisecond time constant. This is applied to the meter, which is assumed to have instantaneous response so that it clearly shows the effect of the two previous time constants.

In tests, this meter agreed with average listeners within 2dB. However, it's important to note that listeners disagreed amongst themselves by as much as 4dB when asked to assess the subjective loudness of a given piece of program material. So any loudness meter can only work for an average listener, and may show considerably greater errors when compared to any given listener.

Advantages of Multiband Compression in DTV Audio

Multiband compression is useful in performing an "automatic equalization" function to change the frequency balance of the audio on a program-adaptive basis. In a multiband compressor, frequency bands containing excessive energy are automatically compressed more than other bands. This results in a re-equalization of the program material towards some target spectral balance. The 6200's two-band compressor controls excessive bass, which can otherwise cause muddy balances. The five-band compressor can perform more detailed automatic re-equalization that can be very useful for program material such as live news.

Availability of multiband compression is another argument for passing most program material through a conventional compressor with loudness control even in DTV service. Multiband compression smooths out not only loudness variations but also variations in equalization, which can be particularly valuable with program material that has to air in a timely manner, where there is no time budgeted for careful audio post-production. Material that airs with full Dynamic Range Control implemented should be refined so that it sounds polished and consistent without further processing. A considerable amount of televised material does not meet this criterion.

Using the 6200 in the 5.1 Channel Plant

The following is one potential scenario for installing the 6200 in a 5.1 channel plant. Obviously, other, more complicated switching arrangements could also be designed. The only crucial requirements are keeping the time delay of all channels equal and compensating for the delay of the 6200, which will ordinarily be padded to 1 frame in this application.

In a full 5.1 channel plant, the channels ordinarily used to transmit 2-channel material should be applied to the 6200's input and to one input of an AES/EBU switcher. A second input of the AES/EBU switcher receives the 6200's output. The output of the switcher drives the channel 1/2 input of the Dolby AC-3 encoder. The remaining 3.1 channels are applied directly to the AC-3 encoder.

The 6200 is configured for one-frame delay. The AC-3 encoder is configured with a preset that (1) reduces its delay by one frame, (2) defeats the AC-3 encoder's internal DRC compressor, and (3) puts the AC-3 encoder into two-channel mode. This preset is activated whenever the AES/EBU switcher chooses the output of the 6200 as the input source for the AC3 encoder's channel 1 / 2 input. As discussed earlier, this configuration is made active whenever the station is transmitting "garden variety" program material and wants the advantages of automatic loudness control and multiband compression.

If the AES/EBU switcher has a delay, it will be necessary to put a matching delay in the other 3.1 channels to ensure that phasing is maintained in 5.1 channel operation.

Bypassing the Loudness Controller

The Loudness Controller can reduce the impact of sound effects in dramatic programming, so the broadcaster may wish to defeat it during such programming and turn it on during breaks and commercials. To do this, temporarily edit your on-air preset to turn off the loudness controller, and save this edited preset as a new user preset. To turn the loudness controller off and on, recall the appropriate preset.

The 6200 in 2-Channel DTV Applications

Some DTV systems (such as DBS satellite) specify two-channel audio and do not provide for metadata. The general comments above regarding 6200 audio processing apply, but the system can be much simpler because no bypassing of the 6200 is required. We recommend padding the 6200's delay to one frame of the system in use (which can be done from the 6200's Setup menu) and then ensuring that the overall audio delay matches the video.

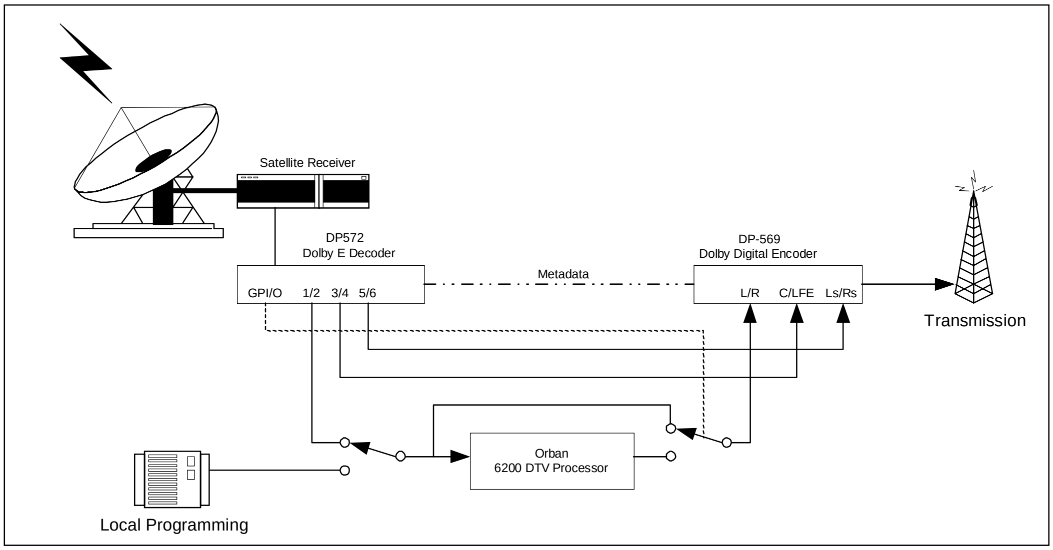


Figure 1-1: Possible DTV system setup using 6200
(drawing courtesy of Dolby Laboratories)

Webcasting Applications

This section was written in early 1999. As the state of the art in webcasting is changing with ferocious rapidity, we expect it to become outdated quickly. Please check Orban's web site, www.orban.com, for newer information.

Using the 6200 in Streaming Applications

You need an audio source connection (either analog, AES/EBU digital, or SPDIF digital). The digital input can accept any sample rate from 32 to 48kHz.

The output of the 6200 can be set to either the professional AES/EBU or the consumer SPDIF standard. Many sound cards with digital inputs accept SPDIF signals. It is not advisable to use a sound card with only an analog output, because these will almost certainly distort the shape of the carefully peak-controlled waveform, increasing the peak-to-average ratio. Because the peak level is limited by digital full-scale, this means that the average level must be reduced, reducing loudness accordingly.

Ordinarily, you will set the digital output level of the 6200 to –0.5dBfs — unless you observe clipping or other distortion (refer to "Decoder headroom" below). Set the output sample rate to 32, 44.1, or 48kHz, according to the requirements of your sound card and encoder. Set the 6200's output word length to 16 bits and turn dither on, unless your encoder accepts higher word lengths. (The 6200 can also output 18 or 20-bit words.)

If you are encoding at a sample rate lower than 44.1kHz, set the 6200's lowpass filter cutoff frequency to approximately 45% of the encoder's sample frequency. You may have to read your encoder documentation carefully to determine the sample frequency, because this bears no relationship to the output bit rate after lossy audio compression by the encoder. The encoder may do a sample rate conversion on its input before encoding.

Using the 6200 in Non-Streaming Applications, to Prepare Audio Files for Download

The 6200's inputs and outputs will be connected the same way that they are for streaming applications. There are two scenarios. In the first, you pass external program material through the 6200 and then record the processed program onto your hard disk. This can be done with any sound card having a digital input. The resultant files are then processed offline through the appropriate encoding application(s).

The second scenario supports processing of files already existing on the computer's hard disk or other storage device. It requires a sound card capable of full duplex operation, such that its digital input and output can be used simultaneously. Few consumer-grade sound cards support this mode of operation, but most cards designed for audio production will do so. The processor is then used in loop-through mode, accepting unprocessed playback from the computer and emitting processed audio, which is then recorded as a separate file through the sound card's input onto the hard disk.

Decoder Headroom

It is possible that highly processed audio (like the output of the 6200) will cause certain decoders to overshoot and clip, even if no clipping occurs at the encode side. This is a design defect in the decoder. Nevertheless, you should carefully test and qualify the latest decoding software (and possibly hardware) that complements the encoding algorithm you are using. If you observe clipping or other distortion, you will have to reduce the 6200's output level until the decoder no longer distorts.

You should periodically re-test suspect decoders and hardware as new revisions become available to ascertain when (or if) the problem has been fixed.

Loudness

You can expect a large increase in loudness from 6200 processing by comparison to unprocessed audio. (An exception is recently mastered CDs, which may have already been aggressively processed for loudness when they were mastered.) In radio broadcasting, it is generally believed that loudness relative to other stations attracts an audience that perceives the station as being more powerful than its competition. We expect that the same subliminal psychology will hold in webcasting too.

Choosing your Encoder

The state of the art in encoder technology changes weekly. We can give no specific recommendations. However, be aware that different encoders are optimized for different bit rates, and you should match your encoder to your potential audience. An encoder appropriate for a dial-up rate of 20kb/sec may not be optimum for ISDN, DSL, or T-1 rates. This makes it necessary to use more than one algorithm to optimally serve audiences with these disparate connection speeds.

MP3

MPEG-1 Layer 3 has become a de-facto standard for distribution of non-streaming, high fidelity audio on the Internet. The 6200 is well matched to MP3, and can effectively preprocess audio intended for MP3 playback.

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 6200

Allow about 2 hours for installation.

Installation consists of: (1) unpacking and inspecting the 6200, (2) optional resetting of jumpers for input termination, (3) checking the line voltage setting, fuse, and power cord, (4) mounting the 6200 in a rack, (5) connecting inputs, outputs and power, (6) setting the Ground Lift switch, (7) optional connecting of remote control leads — 6200 OPTIMOD-DAB only — and (8) optional connecting of computer interface control leads.

When you have finished installing the 6200, proceed to “System 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 6200 are:

1 Operating Manual
1 Line Cord
2 Fuses (F1 = ½A or 250mA
    depending upon specified mains voltage)
4 Rack-mounting screws, 10-32 x ½ — with washers, #10

B) Save all packing materials! If you should ever have to ship the 6200 (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).

We do not sell our customer’s names to anyone.

2. Change standard factory input termination, if required.

[Skip this step if your installation does not require 600Ω termination on the analog left/right inputs.]

The analog left/right inputs are shipped from the factory with balanced bridging (10kΩ) input impedance. However, the 6200 analog inputs can be changed to 600Ω input impedance.

To change the input impedance of the analog left/right inputs:

A) Make sure that power is off before removing the covers.

B) Remove the 18 screws fastening the top cover to the chassis. Remove the top cover to expose the internal circuit board.

C) Move jumpers J301 and J303 according to Figure 2-1.

Jumper J301 sets the left channel termination and jumper J303 sets the right channel termination.

D) Replace the top cover and the 18 screws fastening it to the chassis.

Take care not to strip these self-tapping screws by tightening them with excessive force.

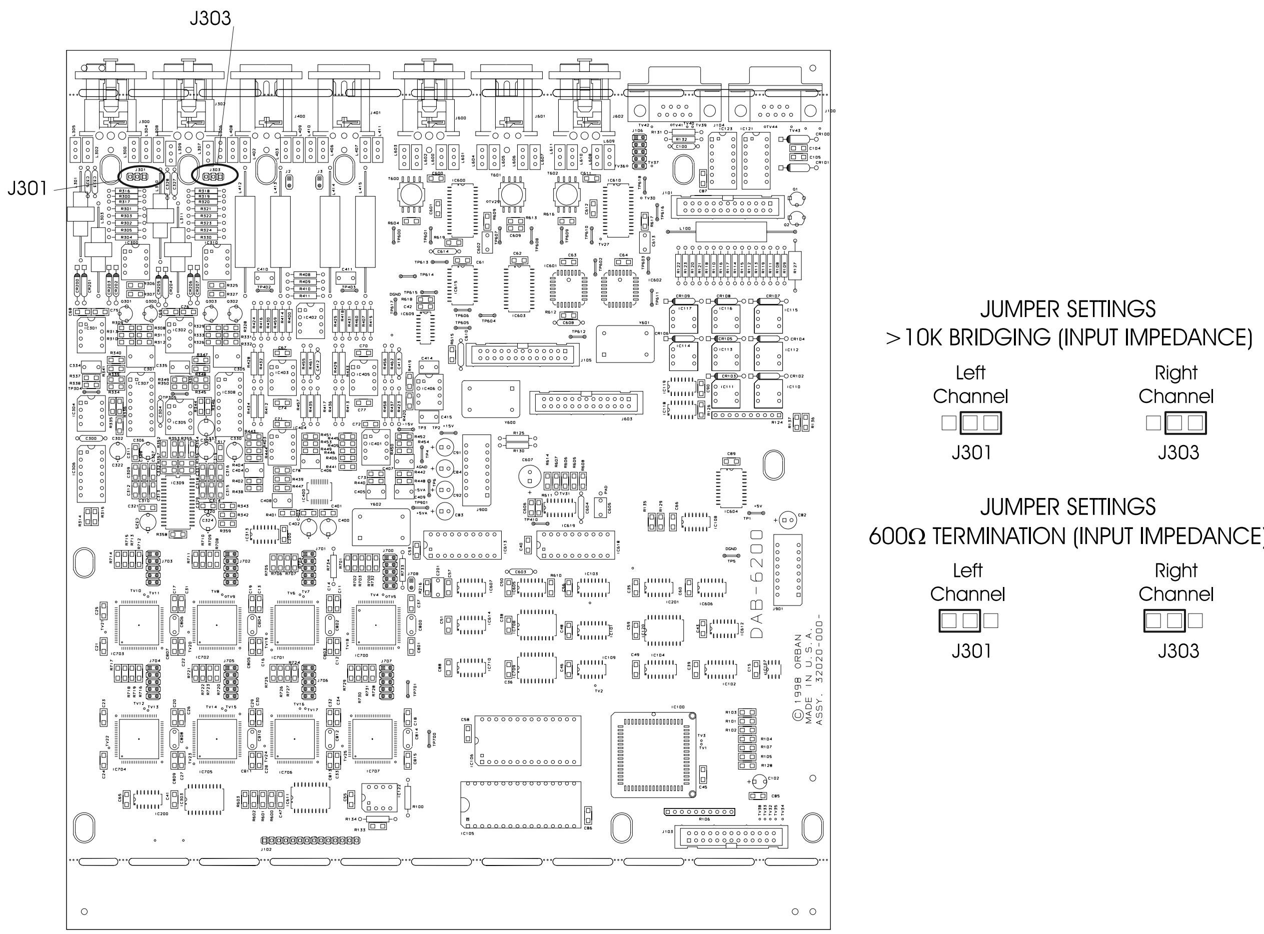


Figure 2-1: Jumper Settings

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

A) DO NOT connect power to the unit yet!

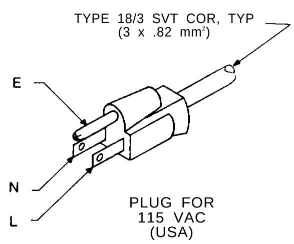
B) Check the Voltage Selector. This is on the rear panel.

The 6200 is shipped configured for either 90-130V or 200-250V, 50Hz or 60Hz operation, as indicated on the rear panel. Refer to the unit's rear panel for your Model Number and the inside of the front cover of this manual for your Model Number's line voltage setting. To change the operating voltage, set the Voltage Selector to 115V (for 90-130V) or 230V (for 200-250V) as appropriate.

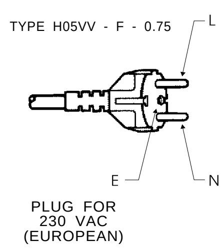
C) Check the value of the fuse and change the fuse if the value is incorrect.

Important: If the fuse is not installed, select the appropriate fuse from the accessory kit and install it.

For safety, fuse F1 must be Slow-Blow, ½-amp for 115V, or ¼-amp (250mA) "T" type for 230V.



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



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

Figure 2-2: 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 6200's Model Number. The green/yellow wire is connected directly to the 6200 chassis.

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

4. Set Ground Lift switch.

The GND LIFT (Ground Lift) switch is located on the rear panel.

The Ground Lift switch is shipped from the factory in the "CHASSIS GROUND" position, (to connect the 6200's circuit ground to its chassis ground). We can think of no reason to set it to LIFT, but if you develop a system grounding problem the switch is available as a resource to help you fix your problem.

5. Mount the 6200 in a rack.

The 6200 OPTIMOD-DAB requires two standard rack units (3.5 inches/8.8 cm). Model 6200S requires one standard rack unit (1¾ inches/4.4 cm).

There should be a good ground connection between the rack and the 6200 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.

6. Connect remote control. (optional, 6200 OPTIMOD-DAB only)

The 6200 OPTIMOD-DAB has extensive remote control provisions.

Optically isolated remote control connections are terminated in a type DB-25 female connector located on the rear panel. It is wired according to Fig. 2-3. To select the desired function, apply a 6-24V AC or DC pulse between the appropriate Remote terminals. The (–) terminals can be connected together and then connected to ground at pin 1 to create a Remote Common. If you use 48V, connect a 1kΩ ±10%, 2-watt carbon composition resistor in series with the Remote Common or the (+) terminal to provide current limiting. A current-limited +9VDC source is available on pin 25.

> 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. SYNC TALLY
11. SIGNAL TALLY
12. N/C
13. POWER COMMON
14. REMOTE 1-
15. REMOTE 2-
16. REMOTE 3-
17. REMOTE 4-
18. REMOTE 5-
19. REMOTE 6-
20. REMOTE 7-
21. REMOTE 8-

22 - 24. N/C

25. +9VDC

REMOTE INTERFACE

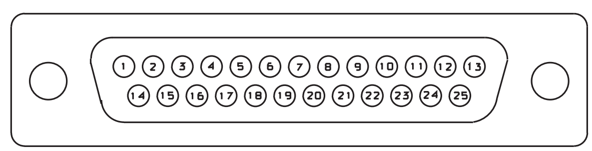


Figure 1-3: Wiring the 25-pin Remote Control Connector

7. Connect computer interface. (optional)

The RS-232 connector is wired according to Fig. 2-4.

If you want to connect the 6200 directly to a computer, use a “null modem” cable. If you want to connect it to a modem, use a conventional computer-to-modem cable.

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.

(Most competently designed serial cables are already well shielded to prevent the cable from radiating EMI to the environment.)

For complete 6200 PC installation steps, refer to the separate 6200 PC document.

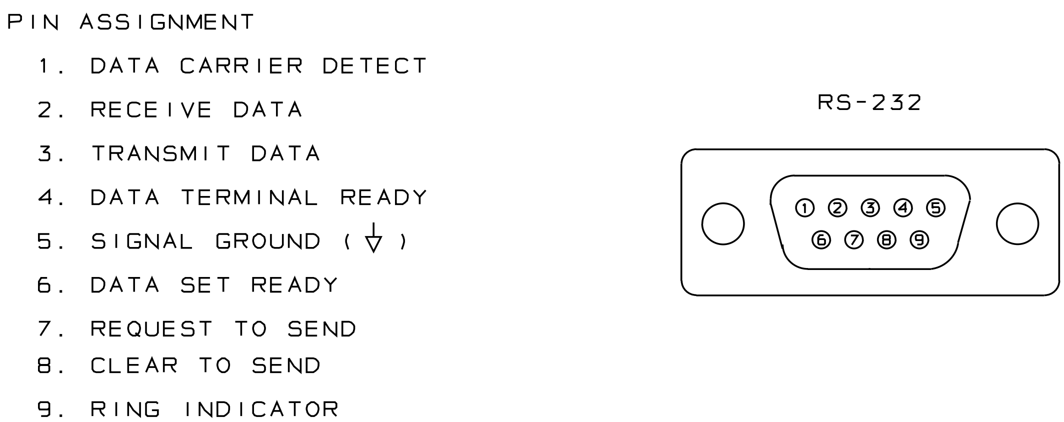


Figure 2-4: Wiring the RS-232 Computer Interface Connector

8. Connect inputs and outputs.

See the hookup and grounding information on the following pages.

Audio Input and Audio Output Connections Page 2-8
AES/EBU Digital Input and Output Page 2-10
Grounding Page 2-10

6200 Rear Panel

The **Voltage Selector** 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 3AG Slow-Blow, ½-amp for 115V, or ¼-amp (250mA) “T” type 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 6200’s Model Number.

The **GND LIFT (Ground Lift) Switch** can be set to connect the 6200’s circuit ground to its chassis ground (in the “CHASSIS GROUND” position). In the LIFT position, it breaks that connection. Although we have provided this switch to ensure versatility in any installation, we can conceive of no installation where it would be set to LIFT.

An **RS-232 (PC Remote) Computer Interface** on **Port 1** is provided to connect the 6200 to IBM PC-compatible computers, directly or via modem, for remote control and metering. In addition, an **RS-232 (PAD) interface** on **Port 2** is provided to connect the 6200 to a DAB transmitter to communicate with it, providing PAD (Program-Associated Data) to the transmitter and receiving frame synchronization data from it. Both remote computer interfaces use DB-9 connectors.

For 6200 OPTIMOD-DAB units only, an optically isolated **Remote Control Interface** is provided to connect the 6200 to your existing transmitter remote control. The 6200 remote control supports user-programmable selection of up to eight inputs for any one of the following parameters: user presets, factory presets, test presets, stereo, mono left, mono right, analog input, digital input, digital output parameters, low-pass filter. (See Remote Interface Programming on page 2-36.) The 6200 remote control accepts a DB-25 connector.

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.

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 one pin is considered LOW, the other pin is automatically HIGH.

Analog Audio Input

- Nominal input level between –14dBu and +8dBu will result in normal operation of the 6200.

(0dBu = 0.775Vrms. For this application, the dBm@600Ω scale on voltmeters can be read as if it were calibrated in dBu.)

- The peak input level that causes overload is dependent on the setting of the AI CLIP (Analog Input Clipping) level control. It is adjustable from +5.0dBu and +27.0dBu.

- The electronically balanced input uses an ultra low noise and distortion differential amplifier for best common mode rejection, and is compatible with most professional and semi-professional audio equipment, balanced or unbalanced, having a source impedance of 600Ω or less. The input is EMI suppressed.

- Input connections are the same whether the driving source is balanced or unbalanced.

- Connect the red (or white) wire to the pin on the XLR-type connector (#2 or #3) that is considered HIGH by the standards of your organization. Connect the black wire to the pin on the XLR-type connector (#3 or #2) that is considered LOW by the standards of your organization.

- In low RF fields (like a studio site), do not connect the cable shield at the 6200 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 6200 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 Monitor Output

- Electronically balanced and floating outputs simulate a true transformer output. The source impedance is 30Ω. The output is capable of driving loads of 600Ω or higher; the output level is fixed at +14dBu = 100% modulation. 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 6200'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.

AES/EBU Digital Input and Output

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 48kHz.

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 0dB relative to the maximum digital word. The maximum digital input will make the 6200 input meters display 0dB. The reference level is adjustable using the DI REF VU and DI REF PPM level controls.

Best peak control will occur if the output is run at 48kHz. See page 1-8 for a discussion.

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 6200 has balanced inputs.
- 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 6200 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 to 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 6200's GND LIFT (Ground Lift) switch to its "CHASSIS GROUND" connect position.

- In high RF fields, the system is usually grounded through the equipment rack in which the 6200 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.

6200 Front Panel

- **Screen Display** labels the four soft keys and provides control setting information.

- Screen **Contrast** button adjusts the optimum viewing angle of the screen display.

- Four **Soft Keys** provide access to all 6200 functions and controls. The functions of the soft keys change with each screen, according to the labels at the bottom of each screen.

- **Next** and **Prev** (← and →) buttons are used to horizontally scroll through the screen to accommodate menus that cannot fit in the available space.

  These flash when such a menu is in use. Otherwise they are inactive.

- **Control Knob** is used to change the setting that is selected by the soft keys.

- **Recall** button allows you recall a Factory or User Preset.

- **Modify** button allows you to edit a Factory or User Preset. (If you edit a Factory Preset, you must save it as a new User Preset to retain your edit.)

- **Setup** button accesses the technical parameters necessary to match the 6200 to your transmission system.

- **Escape** button provides an escape from current screen and returns user to the next previous screen. Repeatedly pressing *Escape* will always return you to the Idle screen.

- **Input** meters show the peak input level applied to the 6200's analog or digital inputs with reference to 0 = digital full-scale.

- **AGC** meter shows the gain reduction of the slow AGC processing that precedes the multi-band compressor. Full-scale is 25dB gain reduction.

  Because the AGC is a two-band unit with Orban's patented bass coupling system, this meter actually reads the gain reduction of the AGC Master band.

- **Gate** LED indicates gate activity, lighting when the input audio falls below the threshold set by the gate threshold control (with the Modify screen's FULL CONTROL, GATE THR control). When this happens, the compressor's recovery time is drastically slowed to prevent noise rush-up during low-level passages.

- **Gain Reduction** meters show the gain reduction in the multi-band compressor. Full-scale is 25dB gain reduction.

  If the Multi-Band structure is operational, all the meters light. If the Two-Band structure is operational, the two leftmost meters light. If the Protect structure is operational, only the leftmost meter lights.

- **Limiter** meters show the amount of broadband look-ahead peak limiting in the left and right channels, which are not coupled because of the fast release time of this circuit. Full-scale is 12dB gain reduction.

Installation of Studio Level Controller (optional)

Refer to “Gain Control before the STL” on page 1-11.

[Skip this section if you are not using a studio level controller ahead of the 6200. Continue with “System 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, then 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 Figure 2-5.

a) Place jumper JA in the “CLIPPER ON” position.

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.

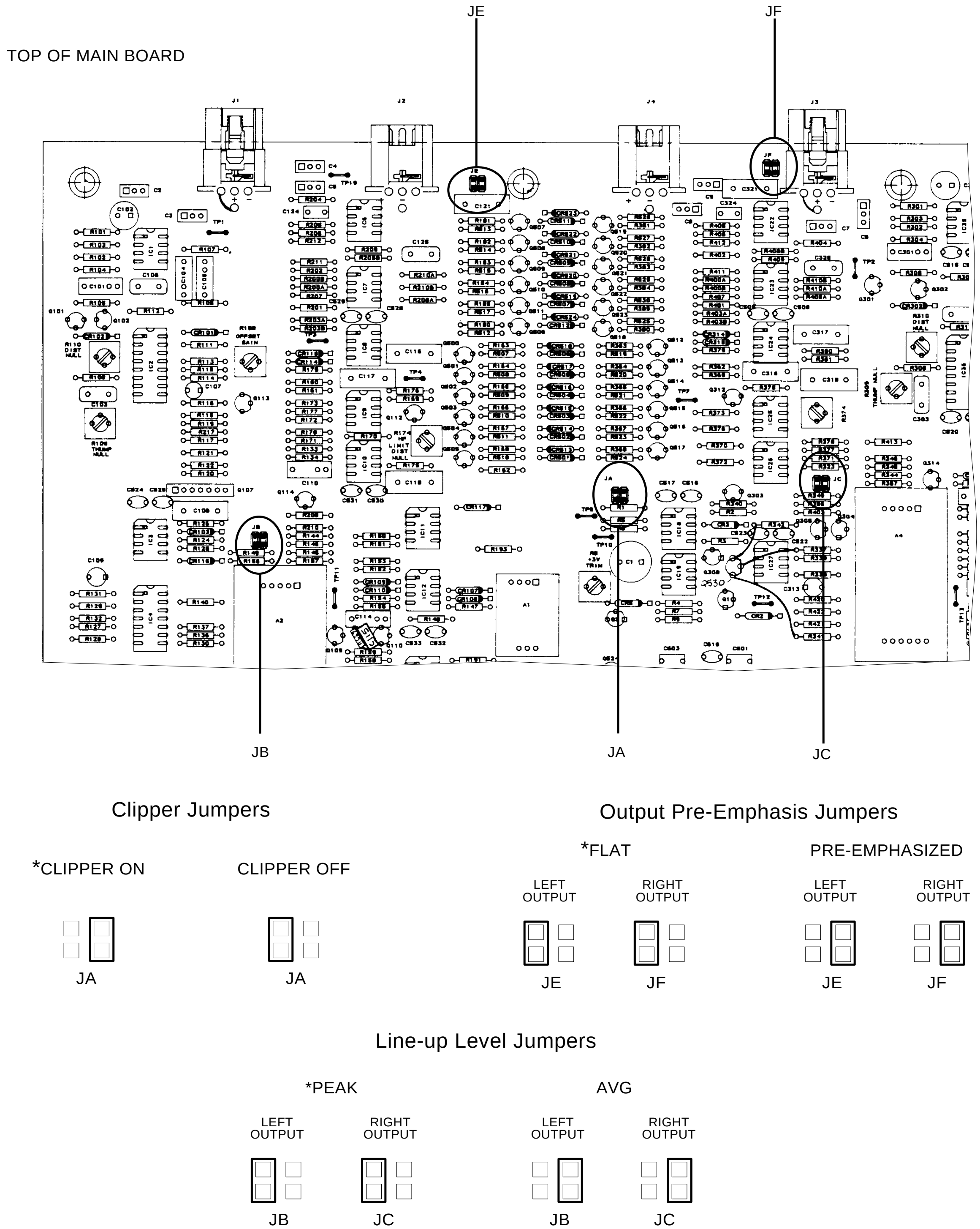


Figure 2-5: 8200ST Jumper Settings (*Factory Configuration)

3. Set 8200ST Output Level with tone.

A) Press the *TONE* button on the 8200ST.

The *TONE* lamp should light and the modulation meters should indicate "0." If they do not, re-strap jumpers JB and JC to "peak." (Refer to Figure 2-5.) 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 overmodulation caused by overshoots on certain audio signals.

4. Set controls for normal operation with program material.

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

A) Set controls as follows:

HF LIMITER:	Set to match the pre-emphasis of the transmission system
L OUT:	Do not change
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 8-15dB gain reduction for most formats.

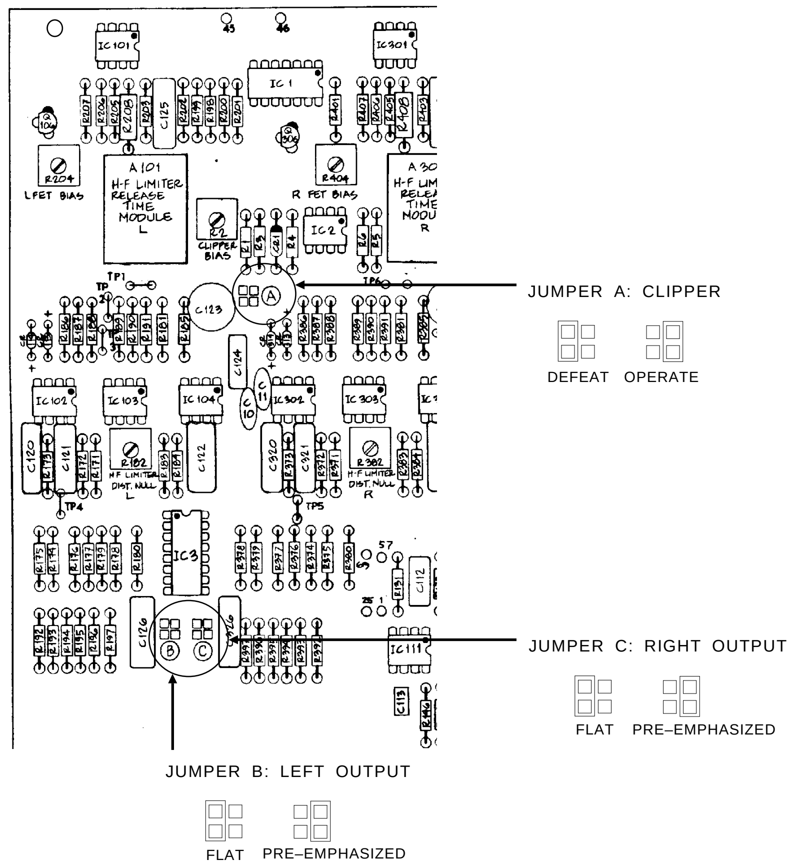


Figure 2-6: 464A Jumper Settings

If you are using Orban 464A Co-Operator

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, then 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, then lift it off.

B) Refer to Figure 2-6.

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 +3dB for 100% modulation of the STL. However, you may wish to calibrate them to indicate 0dB 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 0dB 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 5-10dB.

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, then 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 Figure 2-7 for jumper locations.

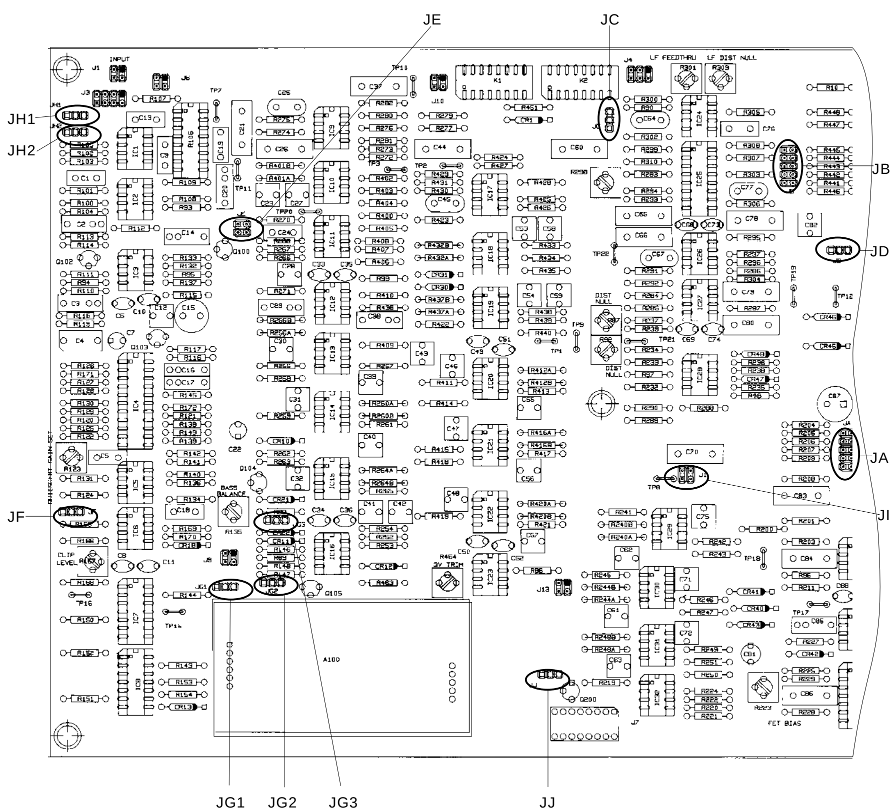


Figure 2-7: 4000 Jumper Locations

C) Activate the high frequency limiter.

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

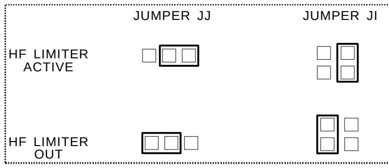


Figure 2-8: 4000 HF Limiter Jumpers

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.)

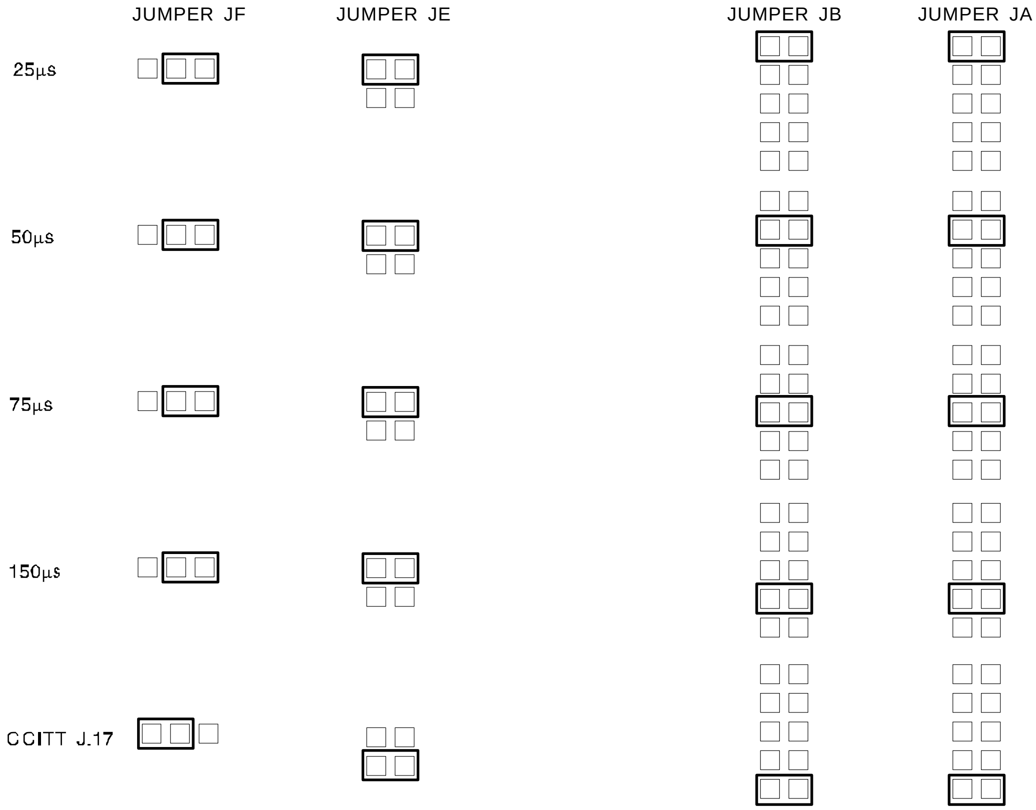


Figure 2-9: 4000 Pre-Emphasis Jumper

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.

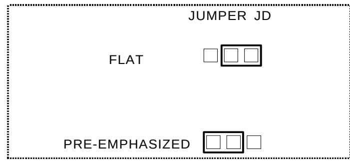


Figure 2-10: 4000 Pre-Emphasis Jumper

F) Set the two channels for stereo coupling.

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

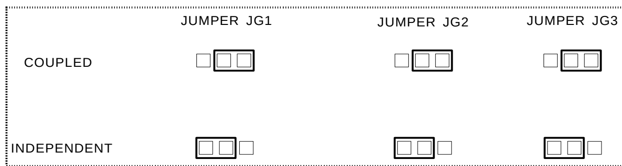


Figure 2-11: 4000 Stereo Coupling Jumper

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 2-22.)]

Some facilities have specific standards for transmission line-up. For example, a transmission standard may state that +4dBu at 400Hz produces 50% modulation of a microwave link. Or PPM6 might allow 8dB 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), we 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)–14db = gain of the 4000

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

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

For facilities using PPM meters, we suggest:

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

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

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

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 7dB 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 2-21.)]

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.

System Setup

For the following adjustments, use the soft keys to select parameters. *While holding down the appropriate soft key*, use the front panel control knob to adjust the parameter settings, as desired.

> If a parameter is not visible and the *Next* and *Prev* buttons are flashing, use these buttons to scroll horizontally to access the desired parameter.

Use the *Contrast* knob to adjust the display for best clarity.

1. Press Setup button to access Setup screen.

A) Press the *Next* button to access miscellaneous Setup parameters.

2. Set the MAX LPF frequency as appropriate to your application.

[4.0 kHz] to [20.0 kHz]

The 6200's audio bandwidth can be set in three places: (1) in Setup, (2) in the EQ page of the Modify screen, and (3) by remote control. The 6200's bandwidth is always the *lowest* of these three settings. The frequency in Setup is a technical parameter that determines the *highest* bandwidth available. The installing engineer should set it to be congruent with the sample rate of the digital system being driven by the 6200. For example, if the 6200 is driving a system with a 32kHz sample rate, set the MAX LPF to 15.0 kHz. That way, a setting of 20kHz elsewhere will not cause excessive bandwidth and aliasing because the 6200 will automatically override it with the MAX LPF setting.

3. Set the ST CHASSIS mode as appropriate for your installation.

[yes] or [no]

This control tells the 6200 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. Setting ST CHASSIS to yes defeats the 6200's AGC.

> Most of the processing structures in the 6200 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 6200 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, answer no to the question (so that the AGC function in the 6200 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 6200.

Temporarily set the ST CHASSIS to "no" so that the Analog and Digital Input reference level alignment steps (below) will work correctly. After you have finished with these steps, you will set the ST CHASSIS parameter appropriately for your installation.

4. Adjust MONO/ST selector.

[STEREO], [MONO-L], [MONO-R], or [MONO-S] for Mono-Sum

This determines which input channel drives the 6200 processing. Normal operation is STEREO. However, you can also drive both channels with the left input channel, the right input channel, or the mono sum of both input channels.

Analog I/O Setup

[Skip this step if you will not be using the 6200's analog input or output, but will only be using the AES/EBU digital input and output. Continue to step 1 on page 2-30.]

For the following I/O CALIB parameters, use the soft keys to select input/output parameters. *While holding down the appropriate soft key*, use the front panel control knob to adjust the parameter settings, as desired.

If a parameter is not visible and the *Next* and *Prev* buttons are flashing, use these buttons to scroll horizontally to access the desired parameter.

The screen as shown above lists all of the controls available for adjustment.

1. Adjust Input selector.

[Skip steps 1-4 if you will not be using the 6200's analog input. Continue to step 5 on page 2-28.]

A) Press *Setup* button to access Setup screen.

B) Press I/O CALIB soft key, then press ANLG IN CALIB soft key to access Analog Input/Output calibration controls.

C) Set the INPUT soft key control to analog.

The 6200 will automatically revert to analog input if no valid input is available at the AES/EBU input.

2. Adjust AI CLIP control.

[+5.0dBu to +27dBu] in 0.5dB steps

This step matches the level at which the 6200's A-D (Analog-to-Digital) converter clips to the absolute maximum peak level that your installation supplies to the 6200's analog input.

This setup maximizes the 6200's signal-to-noise ratio. If the clip level is set too low, the 6200'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.

If you are adjusting the 6200 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) page 2-25.

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 2-25, 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 2-25, or
- Calibrate with an Orban 464A Co-Operator, step (D), page 2-26,

as appropriate.

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

A) Calibrate while on air with normal programming.

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

a) Adjust the AI CLIP so that program peaks indicate approximately –10dB on the input meters.

Observe the meters on the 6200 screen for a long period; be sure to observe live announcer voice. If this setting is misadjusted, distortion will result.

0dB indicates input clipping on the 6200. These meters should never peak as high as 0dB with program material.

B) Calibrate with unprocessed audio:

[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 6200's AI CLIP so that the program peaks reach to approximately –2dB on the input meters.

0dB indicates input clipping on the 6200. These meters should never peak as high as 0dB with program material.

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 6200's AI CLIP to indicate –2dB 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 6200 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) 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 464A's *METER CAL* controls so that the 0dB segment on the 464A's PEAK OUTPUT LEVEL meter just illuminates on program peaks.

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

e) Adjust the 6200's AI CLIP so that the program peaks indicate approximately –2dB on the meter on the screen.

f) Return the 464A to the normal settings.

3. Adjust Analog Input Calibration.

A) Adjust AI REF (Analog Input Reference) level.

[–9dBu to +13dBu (VU), or –1 to +21dBu (PPM)] in 0.5dB steps

The AI REF VU and AI REF PPM settings track each other with an offset of 8dB. 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 6200's gain reduction range to the level to which your studio operators peak their program material on the studio meters. This assures that the 6200'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, we are calibrating to the normal indication of the studio meters; this is quite different from the actual peak level or actual average or RMS level.

If you know the reference VU or PPM level that will be presented to the 6200, set the AI REF level to this level, but do verify it with the steps shown directly below.

B) Press the *Recall* button.

C) Turn the control knob until you see next: GENERAL–OPEN.

D) Press the RECALL NEXT soft key.

E) Calibrate using Tone — feed a tone at your reference level to the 6200.

[Skip this step if you are using Program material to calibrate the 6200 to your standard studio level. Skip to step (F).]

If you are not using a studio level controller, 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 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.

a) Verify ST CHASSIS is set to no.

Refer to step 3 on page 2-23 above.

b) Hold down the AI REF soft key and use the 6200's front panel control knob to adjust for 10.0dB on the 6200's AGC meter.

This control has no effect on the AES/EBU digital input.

c) When finished, reset ST CHASSIS to yes, if required (e.g., if that was its setting prior to setting AI REF level).

d) Skip to step 4.

F) Calibrate using Program — feed normal Program material to the 6200.

[Skip this step if you are using Tone to calibrate the 6200 to your standard studio level — see step (E) above.]

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).

a) Verify ST CHASSIS is set to no.

Refer to step 3 on page 2-23 above.

b) Hold down the AI REF soft key and use the 6200's front panel control knob to adjust for an average of 10dB gain reduction on the 6200's AGC meter.

Also, observe the Gate indicator. It should go out when program is present.

If the AGC gain reduction meter averages less than 10dB gain reduction (higher on the meter), or if the Gate indicator stays on when program material is present, re-adjust the AI REF level to a lower level.

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

This control has no effect on the AES/EBU digital input.

c) When finished, reset ST CHASSIS to yes, if required (e.g., if that was its setting prior to setting AI REF level).

4. Adjust R CH BAL (Right Channel Balance).

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

[–2dB to +2dB] on right channel only, 0.1dB 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 6200 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 6200 input.

A) Press *Setup* button to access Setup screen.

B) Press I/O CALIB soft key, then press ANLG IN CALIB soft key.

C) Press the *Next* button to page to the R CH BAL settings, then adjust the parameter, as necessary.

5. Adjust the source of the analog output.

[Skip this step if you will not be using the 6200's analog output.]

A) Press *Setup* button.

B) Press I/O CALIB soft key, then ANLG OUT CALIB soft key.

C) Set the MON OUT control to pre limt (pre limiter) or post lim (post limiter), as desired.

The analog output pair is intended primarily for monitoring because the AES/EBU digital output will give much better peak modulation control. The analog output can be fed either before or after the look-ahead limiter. Since the look-ahead limiter contributes most of the time delay through the system, listening from the output of the multi-band compressor (ahead of the look-ahead limiter) can be much more comfortable to a DJ or presenter monitoring his/her voice through earphones.

6. End I/O CALIB programming.

[Skip this step if you will be using the AES/EBU digital input and output. Continue to step 1 on page 2-30.]

When you are finished adjusting input/output parameters, repeatedly press the *Escape* button to return to the Idle G/R screen.

7. Select the program format of your station.

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

After this step, you can always select a different processing preset, modify presets to customize your sound, and store these presets as User Presets.

A) Press the *Recall* button.

B) Use the knob to scroll through the list of Factory Programming Presets.

C) When you have found an appropriate preset, press the RECALL NEXT soft key.

Your chosen preset is now on the air.

Digital I/O Setup

[Skip these steps if you will not be using the 6200's digital input and output. Refer to page 2-24 to set the 6200's analog I/O.]

For the following I/O Calibration parameters, use the soft keys to select input/output parameters. *While holding down the appropriate soft key*, use the front panel control knob to adjust the parameter settings, as desired.

If a parameter is not visible and the *Next* and *Prev* buttons are flashing, use these buttons to scroll horizontally to access the desired parameter.

1. Adjust Input selector.

[Skip steps 1-4 if you will not be using the 6200's digital input. Continue to step 6 on page 2-32.]

A) Press *Setup* button to access Setup screen.

B) Press I/O CALIB soft key, then press DIG IN CALIB soft key to access Analog Input/Output calibration controls.

C) Set the INPUT control to digital.

The 6200 will automatically revert to analog input if no valid input is available at the AES/EBU input.

2. Set DI MODE.

[normal], [J.17].

If your STL is pre-emphasized with J.17 pre-emphasis, set DI MODE to J.17. This will apply J.17 de-emphasis to the digital input.

Ordinarily only NICAM links will be pre-emphasized to J.17.

Otherwise set DI MODE to normal.

3. Adjust Digital Input Calibration.

A) Adjust DI REF (Digital Input Reference) level.

[–30 to –10dBfs (VU), or –22 to –2dBfs (PPM)] in 0.5dB steps.

The DI REF VU and DI REF PPM settings track each other with an offset of 8dB. 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 6200's gain reduction range to the level to which your studio operators peak their program material on the studio meters. This assures that the 6200'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, we are calibrating to the normal indication of the studio meters; this is quite different from the actual peak level or actual average or RMS level.

If you know the reference VU or PPM level that will be presented to the 6200, set the DI REF level to this level, but do verify it with the steps shown directly below.

B) Press the *Recall* button.

C) Turn the knob until you see next: GENERAL–MEDIUM.

D) Press the RECALL NEXT soft key.

E) Calibrate using Tone — feed a tone at your reference level to the 6200.

[Skip this step if you are using Program material to calibrate the 6200 to your standard studio level. Skip to step (F).]

If you are not using a studio level controller, 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 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.

a) Verify ST CHASSIS is set to no.

Refer to step 3 on page 2-23 above.

b) Hold down the DI REF soft key and use the 6200's front panel control knob to adjust for 10.0dB on the 6200's AGC meter.

This control has no effect on the AES/EBU digital input.

c) When finished, reset ST CHASSIS to yes, if required (e.g., if that was its setting prior to setting the DI REF level).

d) Skip to step 4.

F) Calibrate using Program — feed normal Program material to the 6200.

[Skip this step if you are using Tone to calibrate the 6200 to your standard studio level — see step (E) above.]

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).

a) Verify ST CHASSIS is set to no.

Refer to step 3 on page 2-23 above.

b) Hold down the DI REF soft key and use the 6200's front panel control knob to adjust for an average of 10dB gain reduction on the 6200's AGC meter.

Also, observe the Gate indicator. It should go out when program is present.

If the AGC gain reduction meter averages less than 10dB gain reduction (higher on the meter), or if the Gate indicator stays on when program material is present, re-adjust the DI REF level to a lower level.

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

This control has no effect on the AES/EBU digital input.

c) When finished, reset ST CHASSIS to yes, if required (e.g., if that was its setting prior to setting the DI REF level).

4. Adjust R CH BAL (Right Channel Balance).

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

[–2dB to +2dB] on right channel only, 0.1dB 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 6200 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 6200 input.

A) Press *Setup* button to access Setup screen.

B) Press I/O CALIB soft key, then press DIG IN CALIB soft key.

C) Press the *Next* button to page to the R CH BAL settings, then adjust the parameter, as necessary.

5. Set 6200 USER BITS to receive or block user bits.

[pass], [block].

If you want to transparently pass incoming user bits through to your 6200 digital output, set USER BITS to pass. Otherwise, set this parameter to block.

A) Press *Setup* button to access Setup screen.

B) Press I/O CALIB soft key, then press DIG IN CALIB soft key.

C) Press the *Next* button to page to the USER BITS settings, then adjust the parameter, as necessary.

6. Adjust Digital Output level.

[Skip these steps if you will not be using the 6200's digital output. Continue to step 13 on page 2-34.]

A) Press *Escape* button.

B) Press DIG OUT CALIB soft key.

C) Adjust DO 100% to determine the maximum peak digital output level with reference to full-scale.

7. Set the DO RATE (Digital Output sample rate).

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

48kHz is preferred because its samples are synchronous with the peak controlled samples in the processing.

Selecting a 32kHz output sample rate will automatically set 15kHz as the highest available audio bandwidth.

DO RATE will also affect the available range of test tone frequencies. When DO RATE is set to 32 kHz, the highest TONE FREQ setting is 15000 Hz. When DO RATE is set to 44.1 or 48 kHz, TONE FREQ range extends to 20000 Hz.

8. Set the DO SYNC (Digital Output Sync) mode to internal or external.

[internal] or [external]

The internal sync setting synchronizes the output words to the 6200 internal clock. All inputs are subject to sample rate conversion.

The external setting synchronizes the output rate to the input rate. If a valid AES/EBU signal is present at the AES/EBU SYNC INPUT on the rear panel, it will be used for synchronization. If lock is unavailable at the SYNC INPUT, or if the SYNC INPUT exceeds ±4% of the selected output rate, the AES/EBU INPUT is used. If *both* the SYNC INPUT *and* the AES/EBU INPUT exceed ±4% of the selected output rate, the unit automatically switches to internal. The unit automatically returns to external sync after 1 second of continuously valid AES/EBU signal lock at a valid sample-rate. Furthermore, external lock is only permitted when the output rate matches the sync (or AES/EBU input) rate.

DO SYNC may be external while INPUT is analog. In this case the analog input provides the audio while the SYNC or AES/EBU input provides the sync for the digital output.

9. Set the SYNC DELAY through the processing.

[off], [24 ms], [30 fps], [29.97 fps], [25 fps], [24 fps], and [96 ms]

The 6200 can add time delay to make its input/output delay one frame, using a variety of different standards. The selections are off (approximately 14ms delay), 24 milliseconds, 30 fps, 29.97 fps (NTSC color video), 25 fps (most PAL video), 24 fps (film), and 96 milliseconds.

10.Set the desired output word length.

[16bits], [18bits] and [20bits]

The largest valid word length in the 6200 is 20 bits, limited by the sample rate converters. (Internal word length is 24 bits.) The 6200 can also truncate its output word length to 18 or 16 bits. If the input material is insufficiently dithered for these lower word lengths, the 6200 can also add dither. (See the next step.)

A) Press the *Next* button to page to the Digital Output Calibration screen with Word Len displayed, then adjust the parameter, as necessary.

11. Adjust DITHER to on or off, as desired.

[on] or [off]

The 6200 can add "high-pass" dither before any truncation of the output word. The amount of dither automatically tracks the setting of the WORD LEN 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 3dB 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, you will not need to add dither because the source material has already been correctly dithered. However, particularly if you use the Noise Reduction feature, the processing can sometimes attenuate input dither so that it is insufficient to dither the output correctly. In this case, you should add dither within the 6200.

12. Set STAT BITS (Status Bits) mode.

[AES/EBU], [SPDIF]

This control determines whether the status bits supplied at the digital output are in Consumer (SPDIF) or Professional (AES/EBU) mode.

13. End I/O CALIB programming.

When you are finished adjusting input/output parameters, repeatedly press the *Escape* button to return to the Idle G/R screen.

14. Select the program format of your station.

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 6200 to automatically change presets on a time/date schedule, modify presets to customize your sound, and store these presets as User Presets.

A) Press the *Recall* button.

B) Use the knob to scroll through the list of Factory Programming Presets.

C) When you have found an appropriate preset, press the RECALL NEXT soft key.

Your chosen preset is now on the air.

Security and Pascode Programming

[Skip this step if you do not wish to change the security level or program pascodes at this time.]

The 6200 has a relatively simple security system. The same pascode is used for local and remote security. If you plan to use Orban's PC Remote software you *must* set a pascode so the software can connect. If you do not plan to use the PC Remote software, setting a pascode is optional.

A timer that locks out the front panel after a certain period of control inactivity achieves local protection of the 6200. The LOCKOUT soft key activates this feature.

1. Program the pascode.

A) Press *Setup* button.

B) Press SECURITY soft key.

C) Press SET PASCODE soft key to access Set Pascode screen.

D) Turn the knob until the desired number appears.

E) Press *Next* button to program the next number in the pascode.

F) Repeat this sequence to program a pascode with up to eight numbers.

A pascode may have one to eight numbers.

G) When you have finished programming your pascode, press the SAVE PASCODE soft key to save it. Write it down and keep it in a safe place.

2. Program local lockout.

A) Press *Escape* button to reveal the LOCKOUT soft key.

B) Set the desired lockout time (if any).

You can program the lockout delay time (in hours:minutes) from 15 minutes to 8 hours, or off. This is the time delay between the last access to a local front panel control and when the front panel automatically locks itself out, requiring the pascode to re-establish communication with the 6200.

Remote Control Interface Programming

Important Note: Model 6200S does not have a rear panel Remote Interface connector; it does not support remote interface programming. However, 6200S can be controlled remotely with 6200 PC software: If you plan to use Orban PC Remote software, you may need to change the Modem Initialization String. Refer to step 6 below and the 6200 PC Manual.

[Skip this step if you do not wish to program the remote control interface at this time. Skip to step 6 if you only want to change the Modem Initialization String.]

1. Press Setup button.

2. Press REMOTE soft key.

3. Press REMOTE INTERFACE soft key to access the remote control interface for programming.

The control interface is eight opto-isolated inputs, which allow you to direct the 6200 to perform certain functions when a voltage (6-24V) is presented to the input.

This screen is used to program the function of each of the eight control interfaces.

4. Program one or more remote control interfaces.

To program a given remote input, hold down its associated soft key while turning the knob. As you turn the control knob, the functions listed below will appear in the selected (highlighted) field. A momentary pulse or voltage will switch the function.

Preset Name: Switches that preset on the air. Any test preset, factory or user programming preset may be recalled by the control interface.

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

do rate 32, 44.1, or 48kHz: Sets the 6200's digital output sample rate to 32, 44.1, or 48kHz.

stereo, mono-left, mono-right, or mono-sum: Selects the input routing to the processing.

analog in, or digital in: Selects the analog or digital inputs. (The processing will always revert to analog if a valid digital input is not available.)

LPF 4.0 kHz, to LPF 20.0 kHz (21 different steps): Sets the system audio bandwidth (low-pass filter) from 4.0 to 20.0kHz. (Choices higher than the MAX LPF frequency in System Setup are locked out — see step 2 on page 2-23.)

word length 16, 18, or 20: Selects 16-, 18-, or 20-bit output words. See step 10 on page 2-33.

dither on or off: sets dither status. See step 11 on page 2-34.

no function: remote interface disabled.

5. End remote control interface programming.

A) When you are finished programming the remote control interface, press the *Escape* button once to return to the Remote screen.

6. Set Modem Initialization String.

[Skip this step if you are not using 6200 PC Remote Control software.]

A) Press SET MODEM INIT soft key to access Modem Initialization screen.

The 6200 powers up with a default initialization string [AT&FE0S0=4].

Always begin by using this default. If you cannot get a reliable connection with the default, then you may edit this string as necessary. Use the *Next* and *Prev* buttons to move through the characters and use the knob to set a character as desired.

To verify the string, complete the 6200 PC hardware and installation steps (refer to 6200 PC Operating Manual), then proceed:

With the modem connected to the 6200 (modem plugged into the 6200's RS-232 connector), its AA light comes on within 10 seconds and stays on. If the AA light doesn't come on and/or the received data light (often labeled RD or RX) flashes at regular intervals, something is wrong with the initialization string.

In this case, reset the MODEM INIT string for the following parameters: recall factory preset, echo off, and auto-answer (Refer to the modem's manual).

If the modem initialization string is set correctly, and 6200 PC and the 6200 still do not negotiate successfully, make sure the modem is set for V.42 operation only.

7. Complete Installation.

Installation is finished; repeatedly press the *Escape* button to return to the Idle G/R screen.

Section 3 Operation

6200 Front Panel

- **Screen Display** labels the four soft keys and provides control setting information.
- Screen **Contrast** button adjusts the optimum viewing angle of the screen display.
- Four **Soft Keys** provide access to all 6200 functions and controls. The functions of the soft keys change with each screen, according to the labels at the bottom of each screen
- **Next** and **Prev** (← and →) buttons are used to horizontally scroll the screen to accommodate menus that cannot fit in the available space.

  These flash when such a menu is in use. Otherwise they are inactive.
- **Control Knob** is used to change the setting that is selected by the soft keys.
- **Recall** button allows you recall a Factory or User Preset.
- **Modify** button allows you to edit a Factory or User Preset. (If you edit a Factory Preset, you must save it as a new User Preset to retain your edit.)
- **Setup** button accesses the technical parameters necessary to match the 6200 to your transmission system.
- **Escape** button provides an escape from current screen and returns user to the next previous screen. Repeatedly pressing *Escape* will always return you to the Idle screen.
- **Input** meters show the peak input level applied to the 6200's analog or digital inputs with reference to 0 = digital full-scale.
- **AGC** meter shows the gain reduction of the slow AGC processing that precedes the multi-band compressor. Full-scale is 25dB gain reduction.

  Because the AGC is a two-band unit with Orban's patented bass coupling system, this meter actually reads the gain reduction of the AGC Master band.
- **Gate** LED indicated gate activity, lighting when the input audio falls below the threshold set by the gate threshold control (with the Modify screen's FULL CONTROL, GATE THR control). When this happens, the compressor's recovery time is drastically slowed to prevent noise rush-up during low-level passages.
- **Gain Reduction** meters show the gain reduction in the multi-band compressor. Full-scale is 25dB gain reduction.

  If the Multi-Band structure is operational, all the meters light. If the Two-Band structure is operational, the two leftmost meters light. If the Protection Limiter structure is operational, only the leftmost meter lights.

- **Limiter** meters show the amount of broadband look-ahead peak limiting in the left and right channels, which are not coupled because of the fast release time of this circuit. Full-scale is 12dB gain reduction.

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 look-ahead limiter in the 6200 operates with very fast attack and release times and is closer to a clipper than a conventional peak limiter.

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 processing, there is a direct trade-off between loudness and distortion. You can improve one only at the expense of one or both of the other two. Thanks to Orban's psychoacoustically optimized designs, this is less true of Orban processors than of any others. Nevertheless, all intelligent processor designers must acknowledge and work within the laws of physics as they apply to this trade-off.

> In AM and FM processing, we have long said that there is a direct trade-off between loudness, *brightness*, and distortion, However, because DAB systems don't use pre-emphasis there is no problem getting the audio to sound bright, and the trade-off is only between loudness and distortion.

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. But 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. 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-DAB — from Bach to Rock

OPTIMOD-DAB can be adjusted so that the output sounds:

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

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.

Customizing the 6200's Sound

The subjective setup controls on the 6200 give you the flexibility to customize your station's sound. But, 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 6200 controls to suit your format, taste, and competitive situation.

When you start with one of our Factory Presets, there are two levels of subjective adjustment available to you to let you customize the Factory Preset to your requirements: LESS-MORE and FULL CONTROL

Less-More

The single LESS-MORE control changes many different subjective setup control settings simultaneously according to a table that we have created in the 6200'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 processing. We believe that most 6200 users will never need to go beyond the LESS-MORE level of control, because 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.

The LESS-MORE control has a different effect in the "radio" presets than it does in the "television" presets. In the "radio" presets the air sound will become louder as you go from less to more, but (as with any processor) processing artifacts will increase. In the "television" presets (except for those few based on "radio" presets), the LESS-MORE control sets the average amount of dynamic range control provided by the processing. As you go from less to more, the loudness of loud sounds will stay about the same but the loudness of quieter sounds will increase. Because of the 6200's sophisticated gating circuits, very quiet material like background sounds, quiet underscoring, hiss, and hum will not be pumped up.

Please note that, in the "radio" presets, 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.

Full Control

If you want to create a signature sound for your station that is out of the ordinary, or if your taste differs from the people who programmed the LESS-MORE tables, FULL CONTROL 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.

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. Because the LESS-MORE control also adjusts certain invisible system parameters that cannot be accessed by the user — even at the FULL CONTROL level—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 FULL CONTROL level.

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 6200.

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-DAB — 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!

About the Processing Structures

In the 6200, a processing structure is a program that operates as a complete audio processing system. Only one processing structure can be active 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 many possible processing structures achievable by the 6200. Unlike an analog system, where creating a complete processing system involves physically wiring its various components together, the 6200 realizes all of its processing structures as a series of high-speed mathematical computations made by Digital Signal Processing (DSP) integrated circuit chips. In the 6200 all structures operate simultaneously so there is no delay in switching between them, which is done with a smooth cross-fade.

Factory Programming Presets

Factory Programming Presets are our "factory recommended settings" for various program formats or types. The Factory Programming Presets are designed as starting points to help you get on the air quickly without having to understand anything about adjusting the 6200's sound. 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. Because it is so easy to fine-tune the sound at the LESS-MORE level, we think 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 6200'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 Multi-Band sections that follow.

Factory Programming Presets

The presets have been named similarly to their radio counterparts in Orban's OPTIMOD-FM 8200 with firmware version 3.00 and to their television counterparts in Orban's OPTIMOD-TV 8282 (with firmware version 1.2). (There are two more presets available in OPTIMOD-DAB than in 8200 Version 3.00) The basic audio texture of corresponding 6200, 8200, and 8282 presets is quite similar, although the OPTIMOD-DAB presets will usually have a less restricted high frequency sound because the digital channel does not use pre-emphasis and therefore requires no high frequency limiting. This will help engineers implementing In-Band On-Channel (IBOC) DAB systems match the sound of the digital and analog signals as closely as possible. This is important because several of the proposed systems attempt to conceal digital errors by cross-fading to analog in the receiver when the digital signal becomes unusable.

> Unlike the 8200 and 8282, none of the OPTIMOD-DAB presets use phase rotation. So some care will have to be applied in cross-fading to avoid momentary audible comb filtering because of the different phase responses of the analog and digital channels. In practice, this means that the cross-fade should be quite fast — perhaps 50 milliseconds.

Of the 23 Multi-Band presets, six are duplicates because we felt that they were appropriate for more than one format. So there are actually 17 distinct and different Multi-Band presets. Each 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.

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 multi-band 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!** These 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 FULL CONTROL and make small changes to the BASS, MID EQ, and HF EQ controls. Remember to do this after you have decided on a LESS-MORE setting that’s right for you. Once you have edited a preset using FULL CONTROL, LESS MORE is no longer available for that edited preset.

> 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.

FACTORY PROGRAMMING PRESETS (RADIO)		
Preset Names	**Source Preset**	**Normal Less-More**
PROTECTION 0dB	PROTECTION 0dB	2.0
PROTECTION 5dB	PROTECTION 5dB	5.0
2B CLASSICAL	2B CLASSICAL	5.0
2B PROCESSED	2B PROCESSED	7.0
GENERAL-DENSE	GENERAL-DENSE	5.0
GENERAL-MEDIUM	GENERAL-MEDIUM	5.0
GENERAL-OPEN	GENERAL-OPEN	5.0
URBAN/RAP-DENSE	URBAN/RAP-DENSE	7.0
URBAN/RAP-MEDIUM	URBAN/RAP-MEDIUM	7.0
URBAN/RAP-OPEN	URBAN/RAP-OPEN	7.0
ROCK-DENSE	ROCK-DENSE	7.0
ROCK-MEDIUM	ROCK-MEDIUM	7.0
ROCK-OPEN	ROCK-OPEN	7.0
ADLT CONTEMP-MED	GENERAL-MED	5.0
ADLT CONTMP-OPEN	GENERAL-OPEN	5.0
COUNTRY-MEDIUM	GENERAL-MED	5.0
COUNTRY-OPEN	GENERAL-OPEN	5.0
POP-DENSE	GENERAL-DENSE	5.0
POP-MEDIUM	POP-MEDIUM	5.0
POP-OPEN	POP-OPEN	5.0
JAZZ	JAZZ	5.0
INSTRUMENTAL	JAZZ	5.0
OLDIES-DENSE	OLDIES-DENSE	7.0
OLDIES-OPEN	OLDIES-OPEN	7.0
FOLK/TRADITIONAL	POP-MEDIUM	5.0
NEWS/TALK	NEWS/TALK	5.0
SPORTS	SPORTS	5.0

Table 3-1: Factory Programming Presets (Radio)

PROTECTION: The PROTECTION presets are designed for stations wanting the highest possible fidelity to the source, such as a station broadcasting concert music at night when its audience is likely to listen in a concentrated and critical way. Refer to "The Protection Limiter Structure," on page 3-13, for a full discussion.

2B: The 2B presets provide an open, easy-to-listen-to sound that is similar to the source material if the source material is of good quality. These presets are useful for Classical or "fine arts" programming that demands high fidelity to the original program source. For a full discussion on setting up and using two-band presets, refer to "The Two-Band Structure," page 3-20.

GENERAL: The GENERAL presets are a compromise between ROCK and POP. They have a gentle bass and treble lift, along with enough presence energy to help vocals to stand out. These presets are also used for ADULT CONTMP (Adult Contemporary) and COUNTRY, and are a useful candidate for AOR formats.

URBAN/RAP: The URBAN/RAP presets are similar to the ROCK presets but with more bass. They use the 3-pole (18dB/octave) shape on the bass equalizer. They are appropriate for Urban, Rap, Black, R&B, Dance and other similar formats.

ROCK: The ROCK presets are designed for a bright high end and punchy low end (although not as exaggerated as the URBAN/RAP presets).

There is enough presence energy to ensure that vocals stand out. A modest amount of high frequency coupling (determined by the B3>B4 CPL and B4>B5 CPL settings) allows reasonable amounts of automatic HF equalization (to correct dull program material), while still preventing exaggerated frequency balances and excessive HF density.

These presets are appropriate for general rock and contemporary programming.

For Contemporary Hit Radio (CHR) we recommend the DENSE or MEDIUM versions. For Album-Oriented Rock (AOR) we recommend the MEDIUM or OPEN versions, although you might prefer the more conservative ADULT CONTMP (Adult Contemporary) presets here.

ADULT CONTMP (Adult Contemporary): Derived from the GENERAL presets, these presets are a compromise between ROCK and POP. They have a gentle bass and treble lift, along with enough presence energy to help vocals to stand out.

COUNTRY: The COUNTRY presets use the GENERAL source presets. These presets are a compromise between ROCK and POP. They have a gentle bass and treble lift, along with enough presence energy to help vocals to stand out.

POP: POP is a more conservative preset designed for a mellow, open high end. There is substantial high frequency coupling (determined by the B3>B4 CPL and B4>B5 CPL settings) to ensure that the high frequencies do not become dense. This is an ideal preset for formats designed primarily for women listeners (who, by and large, dislike hyped treble) or for any preset designed for long time-spent-listening formats because of its open, clean sound, which leads to very low listener fatigue. Because of its conservative nature, this preset is also used for the FOLK/TRADITIONAL preset.

JAZZ: JAZZ is quite similar to POP, and is specifically tailored toward stations that play mostly instrumental music. It has a relatively mellow high end and produces very low listening fatigue.

INSTRUMENTAL: Derived from the JAZZ source preset, INSTRUMENTAL is quite similar to POP, and is specifically tailored toward stations that play mostly instrumental music. It has a relatively mellow high end and produces very low listening fatigue.

OLDIES: OLDIES is similar to ROCK except high frequency coupling (the B3>B4 CPL and B4>B5 CPL settings) is less. This allows the preset to do substantially more automatic equalization than ROCK, making recordings of different eras more uniform.

OLDIES-OPEN might be a useful alternative to FOLK/TRADITIONAL if the recordings being played are very inconsistent in frequency balance.

FOLK TRADITIONAL: FOLK TRADITIONAL is derived from the POP-MEDIUM source preset. It is a more conservative preset designed for a mellow, open high end. There is substantial high frequency coupling (determined by the B3>B4 CPL and B4>B5 CPL settings) to ensure that the high frequencies do not become dense. This is an ideal preset for formats designed primarily for women listeners (who, by and large, dislike hyped treble) or for any preset designed for long time-spent-listening formats because of its open, clean sound, which leads to very low listener fatigue.

NEWS/TALK: This preset is quite different from the others above. It is based on the fast multi-band 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.

SPORTS: Similar to NEWS/TALK except the AGC REL (AGC Release Time) is slower and the GATE THR (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.

FACTORY PROGRAMMING PRESETS (TELEVISION)		
Preset Names	**Source Preset**	**Normal Less-More**
TV 2B-GEN PURPOSE	TV 2B-GEN PURPOSE	5.0
TV 2B-GP NO LC	TV 2B-GEN PURPOSE	5.0
TV 2B-FINE ARTS	2B CLASSICAL	5.0
TV LIVE NEWS	TV LIVE NEWS	5.0
TV LIVE SPORTS	TV LIVE SPORTS	5.0
TV 5B-GEN PUR W/NR	TV 5B-GEN PURPOSE	5.0
TV 5B-GEN PURPOSE	TV 5B-GEN PURPOSE	5.0
TV 5B-NEWS	TV 5B-NEWS	5.0
TV 5B-SPORTS	TV 5B-SPORTS	5.0
TV 5B-OPTICAL FILM	TV 5B-OPTICAL FILM	5.0

Table 3-2: Factory Programming Presets (Television)

TV 2B-GEN PURPOSE (TV Two-Band General Purpose): This preset is designed to accommodate most dramatic programming, providing gentle gain control that limits dynamic range to a level that provides the general audience with consistently intelligible dialog. It sounds very similar to Orban's analog OPTIMOD-TV (Model 8182A) when that unit is adjusted for "General" programming according to the instructions in its operating manual. This preset retains the spectral balance of its input as much as possible. The loudness controller prevents irritating loudness excursions but can also reduce the impact of sound effects in dramatic programming. A station may therefore wish to activate this preset during commercial breaks, game shows, and the like, and activate TV 2B-GP NO LC (TV Two-Band General Purpose; no Loudness Controller) for dramatic programming. TV 2B-GEN PURPOSE is usually not the best choice for live news, sports, or films with optical soundtracks. The Five-Band presets (see below) can automatically equalize such program material when its spectral balance is inappropriate and can also apply single-ended dynamic noise reduction.

TV 2B-GP NO LC (TV Two-Band General Purpose; no Loudness Controller): is identical to TV 2B-GEN PURPOSE except that the Loudness Controller is defeated.

TV 2B-FINE ARTS (TV Two-Band Fine Arts) is identical to TV 2B-GP NO LC except that it rides gain more slowly than the general-purpose presets. The Loudness Controller was turned off to prevent its compromising the dynamic impact of music. If you need the Loudness Controller function, you can create a User Preset identical to TV 2B-FINE ARTS but with the Loudness Controller activated.

TV LIVE NEWS (Two-Band Live News) rides gain more quickly than the general-purpose presets. Its gate threshold is lower, so it will bring up low level input material more quickly. It is designed for live news programs where input levels may be quite unpredictable. Being a Two-Band algorithm with loudness control, it controls loudness well but does not automatically re-equalize substandard audio (which is quite common in live news broadcasts). You may therefore prefer the Five-Band Live News preset.

TV LIVE SPORTS (Two-Band Live Sports): is similar to TV LIVE NEWS except the release time is slower to resist pumping up crowd noise and the threshold of the single-ended noise reduction system is higher.

TV 5B-GEN PUR W/NR (TV Five-Band General Purpose with Noise Reduction): provides effective dynamic range control and “automatic re-equalization” of most dramatic material. It applies single-ended noise reduction to the material, which will reduce unwanted noise like hiss, hum, or stage rumble. However, it will also reduce ambience. If the program material is carefully produced (as are most contemporary feature-film soundtracks), you may wish to use TV 5B-GEN PURPOSE (which does not apply noise reduction), or, if the material is so well produced that it would not benefit from “automatic re-equalization,” use TV 2B-GP NO LC.

TV 5B-GEN PURPOSE (TV Five-Band General Purpose without Noise Reduction): is identical to TV 5B-GEN PUR W/NR except that the single-ended dynamic noise reduction system is off.

TV 5B-NEWS (TV Five-Band News): rides gain more quickly than the general-purpose presets. Its AGC release time is faster so it will bring up low level material more quickly. It is designed for live news programs where input levels may be quite unpredictable. It also automatically re-equalizes substandard audio (which is quite common in live news broadcasts).

TV 5B-SPORTS (TV Five-Band Sports): is similar to TV 5B-NEWS, except the AGC release time is slower to resist pumping up crowd noise.

TV 5B-OPTICAL FILM (TV Five-Band Optical Film): is designed to make the best of the low-quality audio provided with optical film sound tracks (particularly 16mm). The gate threshold is quite high to avoid pumping up hiss, thumps, and other optical artifacts. The threshold of the single-ended dynamic noise reduction system is also high so that this system can reduce artifacts as far as possible. Release times are slow, because we assumed that material encoded on optical film has already been carefully level-controlled to accommodate the very limited dynamic range of the medium, and that little gain riding is therefore required from the 6200.

The Protection Limiter Structure

The Protection Limiter structure is designed for stations wanting the highest possible fidelity to the source, such as a station broadcasting concert music at night when its audience is likely to listen in a concentrated and critical way. Unlike the other structures, the Protection Limiter structure is not designed to reduce the dynamic range, to increase program density, to increase loudness, or to increase the consistency of sound from different sources. Its only function is to protect the transmitter from over-deviation while preserving the spectral and dynamic quality of the source material.

The Protection Limiter structure is designed for operation below the threshold of limiting most of the time. There are virtually no user controls — the parameters of the structure have been chosen to make it audibly undetectable.

There are two Factory Presets for the Protection Limiter structure. PROTECTION 0dB sets the limiting threshold so that limiting almost never occurs, while PROTECTION 5dB 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 5dB.

Setting Up the Protection Limiter

To set up the Protection Limiter, recall preset PROTECTION 0dB 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 5dB if you want about 5dB 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 than 0dB or 5dB.

There are only two FULL CONTROL parameters for the Protection Limiter: PROT DRIVE (Protection Limiter Drive) level duplicates the effect of the LESS-MORE control, and LO PASS sets the bandwidth of the processor.

> As with all of the other LO PASS parameters within presets, the *lowest* of the settings of this parameter, the MAX LPF filter parameter (in the Setup page), and the LPF parameter (as programmed into a remote interface input) prevails. In all cases, setting a 32kHz output sample frequency will also limit the filter cutoff frequency to 15kHz or lower, regardless of the setting of any of the other parameters above.

Protection Limiting: Orban's Approach

Traditionally, protection limiters have used peak-sensing automatic gain control (AGC) processors to control peak levels. Superficially, this approach seems reasonable. The purpose of a protection limiter is to control the peak levels in a transmission channel.

This traditional approach ignores one crucial requirement for protection limiter performance: The limiter must provide natural-sounding control that is undetectable to the ear except by an A/B comparison to the original source material. Because the human ear is basically average-sensing, not peak-sensing, the simplistic peak-sensing AGC technique causes highly unnatural variations in subjectively-perceived loudness. Audio material with a high peak-to-average ratio emerges from such a limiter much quieter than audio material with a low peak-to-average ratio. The ear perceives this as an unnatural, unpleasant pumping quality. Thus the traditional peak-sensing AGC limiter fails to provide natural sound quality and we must use more sophisticated techniques.

To achieve natural sound quality, the gain control section of the limiter must respond like the ear. This means that the gain control must respond approximately to the power (not the peak level) in the signal. Further, because the sensitivity of the ear decreases dramatically below 150Hz, the control must be frequency-weighted to compensate. Otherwise, heavy bass would audibly modulate the loudness of midrange program material, a problem called spectral gain intermodulation.

The dual-band limiter controls the level driving the following look-ahead limiter. Prior to the HF limiter, a phase-coherent crossover divides the signal into frequency bands above and below 150Hz. The above-150Hz material is connected to the master band, which determines the overall limiting. This prevents limiter-induced spectral gain intermodulation — audible modulation of the loudness of midrange and high frequency program material by bass-generated limiting.

The below-150Hz material is connected to the bass band. The gain-control signal produced by the master band is cross-coupled into the bass band, so that the gain of the bass band ordinarily tracks the gain of the master band exactly, preserving frequency balances. When the bass band encounters exceptionally heavy bass, it momentarily provides extra limiting to preclude excessive level at the dual-band limiter's output.

The dual-band limiter has an attack time of approximately 2 milliseconds. This moderate attack time prevents it from producing limiting on every transient spike. Such limiting could otherwise create audible holes in the program.

The dual-band limiter is gated: When its input level drops below the factory-set threshold of gating, the release rate is radically slowed to avoid audible noise breathing.

> This compressor gate is not the same as a conventional noise gate because it is not intended to reduce noise or other low-level undesired sounds to a lower level than that occurring in the original program. Its only purpose is to prevent the unnatural exaggeration of such material.

The dual-band limiter can produce a maximum of 25dB of limiting. This is more than adequate for protection limiting. It is important not to overdrive the limiter past 25dB gain reduction; the sound will rapidly become highly distorted.

Because the gain control section of the Protection Limiter structure is not peak-sensing, its output contains peak overshoots that must be eliminated by further processing. The look-ahead peak limiter does this. Its attack time is essentially instantaneous and its release time is very fast, so it acts more like a clipper than a traditional peak limiter. However, compared to a simple clipper the bandwidth of the modulation distortion that it produces is much lower. Hence the modulation distortion is more likely to be psychoacoustically masked by the desired program material.

Modulation of Sine Waves

There is an important and sometimes confusing consequence of this system design: The system will not permit sine waves to reach 100% peak modulation. It will restrain sine wave modulation to a lower level — typically 7dB below 100% (45% modulation). Therefore, in its normal OPERATE mode the Protection Limiter structure will not pass an externally-generated line-up tone at 100% modulation; it will produce limiting that constrains the tone to approximately 45% modulation.

This is a direct consequence of the level detection's being power-sensing. For a given peak level, sine waves have very high average power by comparison to program material. To preserve natural sound, the processing must reduce their peak level below the peak level of program material to preserve consistent average power at the limiter's output. This is a characteristic of any limiter that achieves natural-sounding dynamic performance and that does not modulate program loudness according to the peak-to-average ratio of the input signal.

Almost all program material will produce frequent peaks at 100% modulation at the 6200's output. Program material that does not produce such peaks has an unusually low peak-to-average ratio and will sound naturally balanced when applied to the transmission system below 100% peak modulation.

Equalization Controls

MULTI-BAND EQUALIZATION				
Parameter Labels	**Units**	**Default**	**Range (CCW to CW)**	**Step**
LOW BASS	dB	0 (2P)	2P 0 … +12, 3P 0 … +12	1
DJ BASS	---	off	off, on	---
MID BASS	dB	0	0 … +12	1
MF GAIN	dB	0.0	–10.0 … +10.0	0.5
MF FREQ	Hz	3.0	250 … 4000	1/10 oct LOG
MF WIDTH	oct	1.0	0.30 … 2.00	LOG
HF GAIN	dB	0.0	–10.0 … +10.0	0.5
HF FREQ	kHz	6.0	2.00 … 20.00	1/10 oct LOG
HF WIDTH	oct	1.0	0.30 … 2.00	LOG
LO PASS	kHz	20	4.00 … 12.0, 13.0, 14.0, 15.0, 20.0	0.5 ---
B1 OUT	dB	+0.0	–3.0 … +3.0	0.1
B2 OUT	dB	+0.0	–3.0 … +3.0	0.1
B3 OUT	dB	+0.0	–3.0 … +3.0	0.1
B4 OUT	dB	+0.0	–3.0 … +3.0	0.1
B5 OUT	dB	+0.0	–3.0 … +3.0	0.1

Table 3-3: Multi-Band Equalization Controls

Most equalization controls are common to both the Two-Band and Multi-Band structures. The equalizer is located between the AGC and multi-band compressor sections of both structures.

The Protection Limiter structure does not have equalization controls.

Any equalization that you set will be automatically saved in any User Preset that you create and save. For example, you can use a User Preset to combine an unmodified Factory Programming Preset with your custom equalization. Of course, you can also modify the Factory Preset (with LESS-MORE or FULL CONTROL) before you create your User Preset.

In general, there is no good reason to have to use large amounts of EQ (or even any EQ at all) with modern, well-recorded program material. The 6200 multi-band compressor was "tuned" with reference to modern well-recorded CDs, and should produce a highly "commercial" spectral balance with no extra EQ at all. Only if you want to create unconventional spectral balances for your transmission should you use the equalizers.

Table 3-3 shows a summary of the equalization controls available for the Multi-Band structure.

LOW BASS boost control, the Multi-Band structure's low bass equalization control, is designed to add punch and slam to rock and urban music. It provides a shelving boost in two ranges with two switchable characteristics. 2P provides a range of 0dB to +12dB (in

1dB steps) with a 12dB/octave slope. 3P provides a range of 0dB to +12dB (in 1dB steps) at 110Hz and below with an 18dB/octave slope.

Because the Multi-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 (12dB/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 (18dB/octave) shelving boost creates a solid, punchy bass from the better consumer radios with decent bass response. 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 +4dB boost for most formats.

> The LOW BASS boost control steps first through the 2P curves and then through the 3P curves. For example, a 4dB boost with a 12dB/octave slope would appear as "2P +4," and a 10dB boost with an 18dB/octave slope would appear as "3P +10."

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 on. When the control is on, the processing simply freezes the gain of the lowest band under gated conditions. Accordingly, there can be a large average gain difference between the two low frequency bands and the system can produce considerable dynamic bass boost on voice.

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

> This control is only available in the Multi-Band structure.
>
> If the GATE THR (Gate Threshold) control is turned off, the DJ BASS boost setting is disabled (and set to off).

MID BASS control provides a 12dB/octave shelving boost at 200Hz. Use it in conjunction with the LOW BASS boost control to tailor your on-air bass to your exact requirements. A mid bass boost is mainly useful to stations that program to an audience likely to be listening on smaller radios. It can force a thin-sounding radio with a small speaker to seem to have more bass. However, bass boost in this frequency range can make larger radios sound very muddy and boomy, so adjust the MID BASS boost control with great care, listening to both small radios and radios with good bass response.

Midrange 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 ±0.15dB worst-case). This provides warm, smooth, "analog-sounding" equalization.

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

MF FREQ determines the center frequency of the equalization, in Hertz. Range is 250-4000Hz.

MF WIDTH determines the bandwidth of the equalization, in octaves. The range is 0.3-2.0 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, the effect is an actual boost in the amount of power in the presence region, which can increase the loudness of such material substantially. As you increase the gain reduction in the midrange bands (by turning the MB DRIVE control up), the MF GAIN control will have progressively less audible effect. The compressor for the midrange bands will tend to reduce the effect of the MF 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 3.7kHz band compressor is partially coupled to the gain reduction in the 6.2kHz band in most presets, tuning MF FREQ to 2-4kHz and turning up the MF GAIN control will decrease energy in the 6.2kHz band — you will be increasing the gain reduction in both the 3.7kHz and 6.2kHz bands. You may wish to compensate for this effect by turning up the B4>B5 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 2-3dB boost, although 10dB can be achieved.

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.

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

<u>HF FREQ</u> determines the center frequency of the equalization, in Hertz. The range is 2-20kHz

<u>HF WIDTH</u> determines the bandwidth of the equalization, in octaves. The range is 0.3-2.0 octaves. If you are unfamiliar with using a parametric equalizer, 1 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 4dB boost as a practical maximum, unless source material is primarily from compact discs of recently recorded material.

<u>LO PASS</u> control sets the bandwidth (and therefore the amount of high frequency signal the 6200 passes) from 4kHz to 20kHz. The lowpass filter can replace any anti-aliasing filters in downstream equipment. Set the filter to 20kHz (full bandwidth) for downstream equipment with sample rates of 44.1 or 48kHz. Set the filter to 15kHz for 32kHz sample rate. For other sample rates, set the filter so that it is as close as possible to 45% of the sample rate without exceeding 45%.

<u>Multi-Band OUT Mix</u> controls determine the relative balance of the bands in the multi-band 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 ±3dB range.

> Their range has been purposely limited because the only gain control element after these controls is the look-ahead limiter. Like a clipper, this circuit can produce considerable intermodulation distortion if overdriven. The thresholds of the individual compressors have been carefully tuned to prevent audible IM distortion with almost any program material. Large changes in the frequency balance of the compressor outputs will change this tuning, leaving the 6200 more vulnerable to unexpected IM distortion with certain program material. Therefore you should make large changes in EQ with the bass and parametric equalizers, because these are located *before* the compressors and the compressors will therefore protect the system from unusual overloads caused by the chosen equalization. Use the multi-band OUT mix controls only for fine-tuning.

The Two-Band Structure

The Two-Band structure consists of a slow two-band gated AGC (Automatic Gain Control) for gain riding, followed by a gated two-band compressor and a look-ahead limiter. Like the "2-Band Purist" structure in Orban's OPTIMOD-FM 8200, it is phase-linear throughout to maximize sonic transparency.

This structure is equipped with a CBS Loudness Controller, intended primarily for television applications.

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 B2>B1 CPL 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. Its two-band compressor and look-ahead limiter are identical to the Protection Limiter structure, although more controls are made available to the user.

Setting Up the Two-Band Structure

To set up the Two-Band structure for radio, recall preset 2B CLASSICAL for a "smooth," unprocessed-sounding quality, or 2B PROCESSED for a louder, more processed sound. Of course, you can modify any preset using the LESS-MORE control or using the Full Control screen, and you can then store the resulting modified preset as a User Preset for future recall at any time.

Note that choosing a conservative Multi-Band preset (like one of the POP presets or INSTRUMENTAL) will usually create an even more attractive, smooth, and consistent sound on pop music than will the Two-Band structure. Additionally, because each band in the Multi-Band structure handles a smaller part of the audio spectrum than each band in the Two-Band structure, the Multi-Band structure will create less spectral gain intermodulation when driven heavily. We therefore recommend using the Two-Band structure for purist jazz, classical or light classical music, or if your goal is to broadcast a sound that is more faithful to the frequency balances of the original program material than the sound produced by the Multi-Band structure.

> Unlike the Two-Band structures in Orban's FM processors, the 6200's Two-Band structure does not reduce high frequency content because it has no high frequency limiter. (It is not required because digital broadcasting does not use pre-emphasis.) Therefore, the Two-Band structure can be very useful if the broadcaster wants to preserve the spectral balance of the original program material. An example might be a "heritage rock" format where the broadcaster wants to transmit the original frequency balance of a mix that is very familiar to the audience.

For television, recall a TV Two-Band preset that matches the program material. (See page 3-11).

Gain Reduction Metering

Unlike the metering on some processors, when any OPTIMOD-DAB 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 compressor in the Two-Band structure has 25dB of gain reduction range, the meter should never come close to 25dB gain reduction if OPTIMOD-DAB has been set up for a sane amount of gain reduction under ordinary program conditions.

But 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 10dB more peak gain reduction than does music! (A PPM will indicate relative peak levels much more accurately.)

Using the Two-Band Structure for Classical Music

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, making the listening rather 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 compressed 10-15dB so that quiet passages in the music never fade into inaudibility under these less-favorable listening conditions. OPTIMOD-DAB 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 10dB, while the dynamics of crescendos are maintained.

The 2B CLASSICAL 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 Limiter structure.

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 2B CLASSICAL preset) that performs less gain reduction. You can create such a preset by modifying the 2B CLASSICAL preset with the LESS-MORE control — turn it down to taste.

> Internally, the Protection Limiter structure is the same as the Two-Band structure, but the AGC is turned off and its presets have been tuned to pro-

vide less compression and a frequency balance that is maximally faithful to the source material.

The Two-Band Structure's Full Setup Controls

The table shows a summary of the Two-Band controls:

2B FULL CONTROL				
Parameter Labels	**Units**	**Default**	**Range (CCW to CW)**	**Step**
AGC	---	on	off, on	---
AGC DRIVE	dB	(per L/M tables)	–10 … 0 … 25	1
AGC REL	dB/S	(per L/M tables)	0.5, 1.0, 1.5, 2 … 20	1
AGC B CPL	%	(per L/M tabels)	0 … 100	5
GATE THR	dB	(per L/M tables)	off, –44 … –15	1
2B DRIVE	dB	(per L/M tables)	0 … 25	1
2B REL	dB/S	(per L/M tables)	0.5, 1.0, 1.5, 2 … 20	1
LC THRESH	dB	(per L/M tables)	-3.0 … +3.0, off	0.5
BASS CPL	%	(per L/M tables)	0 … 100	5%
BASS CLIP	dB	(per L/M tables)	–6.00 … 0.00	0.25
FINAL LIMT	dB	(per L/M tables)	0.00 … +6.00	0.25
LESS-MORE	---	(per preset)	***(read-only)***	---
PARENT PRESET	---	(per preset)	***(read-only)***	---

Table 3-4: Two-Band Controls

These controls are explained in detail below.

Each Two-Band Factory Preset has a LESS-MORE control 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 FULL CONTROL parameters to your own taste. Always start with LESS-MORE to get as close to your desired sound as possible. Then edit the FULL CONTROL parameters using the Full Control screen, and save those edits to a User Preset.

AGC on/off control activates or defeats the slow AGC prior to the two-band compressor.

AGC DRIVE control adjusts signal level going into the slow 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 Two-Band structure is the sum of the gain reduction in the AGC and the gain reduction in the two-band compressor. The total 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 2B DRIVE control, discussed below.

AGC REL (AGC Release) control determines how fast the AGC compressor releases. It is ordinarily operated in the slow end of its range to allow the AGC to do gentle gain riding. The two-band compressor does the hard work to increase program density (if desired). See 2B REL (Two-Band Release), below, for a further discussion of release time.

AGC B CPL (AGC Bass Coupling) determines the amount of bass coupling in the two-band AGC. Because the AGC is generally operated slowly, one usually sets this control at 80% or higher to prevent the AGC from significantly changing the frequency balance of the program material. "Automatic equalization" is ordinarily done in the two-band compressor section following the AGC. For a further discussion see **BASS CPL** on page 3-24.

GATE THR (Gate Threshold) control determines the lowest input level that will be recognized as program by OPTIMOD-DAB; lower levels are considered to be noise or background sounds, and cause the compressor to gate, effectively freezing its gain.

The two-band gain reduction will eventually recover to 0dB and the AGC gain reduction will eventually recover to –10dB even when the compressor gate is gated. However, recovery is slow enough to be imperceptible. This avoids OPTIMOD-DAB'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 GATE THR control to –40. Higher settings are primarily useful for radio drama, outside sports broadcasts, and other non-musical programming which 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 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 0-25dB.

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 10-15dB. If less gain reduction is used, loudness can be lost. For classical formats, operating with 0-10dB of gain reduction (with the gain riding AGC set to off) maintains a sense of dynamic range while still controlling levels effectively. Because OPTIMOD-DAB's density gently increases between 0 and 10dB of compression, 10dB of compression sounds very natural, even on classical music.

2B REL (Two-Band Release) control determines how fast the two-band compressor releases (and therefore how quickly loudness increases) when the level of the program material decreases. It can be adjusted from 0.5dB/second (slow) to 20dB/second (fast). Settings toward 20dB/second result in a more consistently loud output, while settings toward 0.5dB/second allow a wider variation of dynamic range. Both the setting of the 2B REL 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 REL 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 REL control) will no longer yield more loudness, and will simply degrade the punch and definition of the sound.

When the 2B REL control is set between 8 and 1dB/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 REL control between 8 and 1dB/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 REL control settings (above 8dB/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 10dB). 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 10dB 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-DAB remarkably resistant to operator gain-riding errors.

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

Settings toward 100% (wideband) make the output sound most like the input. Because setting the BASS CPL 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. Adjust the BASS CPL control until the band 1 and band 2 Gain Reduction meters track as closely as possible.

With the 2B REL (Two-Band Release) control set to 2dB/second, setting the BASS CPL 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 BASS CPL toward 100% (wideband) do not sound good. Instead, set the BASS CPL 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 Multi-Band preset instead.

BASS CLIP threshold controls Orban's patented embedded bass clipper. It is embedded in the multi-band crossover so that harmonics created by clipping are rolled off by part of the crossover filters. The threshold of this clipper is ordinarily set between 4dB and 6dB below the threshold of the final limiter in the processing chain, depending on the setting of the LESS-MORE control in the parent preset upon which you are basing your FULL CONTROL adjustments. This provides headroom for contributions from the other four bands, so that bass transients don't smash against the look-ahead limiter, causing overt intermodulation distortion between the bass and higher frequency program material.

Some 6200 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 look-ahead limiter threshold) is 0 to –6dB. 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) and listen for IM distortion induced by the bass' pushing the midrange into the look-ahead limiter. 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 as a result of extensive listening tests with many types of critical program material.

FINAL LIMT (Final Limit) adjusts the level of the audio driving the look-ahead limiter that OPTIMOD-DAB uses to control fast peaks, and then adjusts the peak-to-average ratio. The loudness/distortion trade-off is primarily determined by the FINAL LIMT control.

Turning up the FINAL LIMT control drives the look-ahead limiter harder, reducing the peak-to-average ratio, and increasing the loudness on the air. When the amount of limiting is increased, the audible distortion caused by limiting is increased. Lower settings reduce loudness, of course, but result in a cleaner sound.

> You may find it illuminating to recall several Factory Presets, adjust LESS-MORE to several points in its range, and then open the Full Control screen to examine the trade-offs between the release time and FINAL LIMT drive made by the factory programmers.

LC THRESH (Loudness Controller Threshold) sets the maximum subjective loudness produced by the processing. "0" is a good setting for most DTV programming to ensure that even highly processed commercials will not be objectionably louder than surrounding program material.

Because this is a threshold control, turning it to higher numbers raises the threshold, which means it does less work.

According to CBS Laboratory's research, no common meter, whether peak-reading, average-reading, or even r.m.s.-reading with frequency weighting, can indicate subjective loudness well enough to be relied upon as an objective measurement tool to prevent viewer complaints about loud commercials. True loudness meters incorporate elaborate multiband psychoacoustic models, which agree with average listeners with an accuracy of typically ±2dB. Because listeners will disagree amongst themselves by as much as 4dB when asked to assess the loudness of a piece of program material by comparison to a standardized tone or filtered noise, ±2dB is about as much accuracy as is usable anyway. This also means that a peak-reading or average-reading meter connected to the 6200's output (with Loudness Controller activated) may vary quite a bit in its readings with different program sources. This just shows the Loudness Controller at work and the inability of a conventional meter to accurately assess loudness.

Note that the Loudness Controller determines the *highest* subjective loudness permitted by the 6200 processing. If a commercial follows a piece of quiet program material whose loudness never gets to the threshold of the Loudness Controller (for example, a quiet outdoor scene), then the following commercial will still be louder. Esthetically, this is unavoidable. The best one can do by automatic control is to ensure that commercials are no louder than normally modulated sections of programming that surround the commercials.

To defeat the loudness controller entirely, set the LC THRESH to OFF. For radio programming, you will usually want to defeat the loudness controller.

The Multi-Band Structure

The Multi-Band structure consists of a slow gain-riding two-band AGC, a three-band parametric equalizer, a five-band compressor, a dynamic single-ended noise reduction system, an output mixer (for the five bands), and a look-ahead limiter.

Unlike the Two-Band structure, whose two-band compressor has a continuously variable release time, the release time of the multi-band compressor is switchable to four settings: slow, mslow (medium-slow), mfast (medium-fast), and fast. Each setting makes a very 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.

The Multi-Band structure does not have a Loudness Controller because its “automatic re-equalization” function tends to greatly smooth out loudness differences between one piece of program material and the next. Therefore, television users will find that the Multi-Band structure usually controls loudness well enough to prevent commercials from becoming obtrusively loud.

Putting the Multi-Band Structure on the Air

The Multi-Band structure is very flexible, enabling you to fine-tune your on-air sound to complement your programming. There are 21 Factory Programming Presets whose names are the same as common radio 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 FULL CONTROL to fine-tune them.

The controls in the Multi-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 Multi-Band structure controls to suit your format, taste, and competitive situation.

The Multi-Band Structure's Full Setup Controls

The table shows a summary of the Multi-Band controls:

MULTI-BAND FULL CONTROL				
Parameter Labels	**Units**	**Default**	**Range (CCW to CW)**	**Step**
AGC	---	on	off, on	---
AGC DRIVE	dB	(per L/M tables)	–10 … +25	1
AGC REL	dB/S	(per L/M tables)	0.5, 1.0, 1.5, 2 … 20	1
AGC B CPL	%	(per L/M tables)	0 … 100	5
BASS CLIP	dB	(per L/M tables)	–6.00 … 0.00	0.25
GATE THR	dB	(per L/M tables)	off, –44 … –15	1
MB DRIVE	dB	(per L/M tables)	0 … 25	1
MB REL	---	(per L/M tables)	slow, mslow, mfast, fast	---
B2>B1 CPL	%	(per L/M tables)	0 … 100	5
B3>B4 CPL	%	(per L/M tables)	0 … 100	5
B4>B5 CPL	%	(per L/M tables)	0 … 100	5
B5 THR	dB	(per L/M tables)	–16 … 0	0.3
DWNEXP THR	dB	(per preset)	off, –6 … +12	0.5
FINAL LIMT	dB	(per L/M tables)	0.00 … 6.00	0.25
LESS-MORE	---	(per preset)	***(read-only)***	---
PARENT PRESET	---	(per preset)	***(read-only)***	---

Table 3-5: Multi-Band Controls

These controls are explained in detail below.

Each Multi-Band 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 FULL CONTROL parameters to your own taste. Always start with LESS-MORE to get as close to your desired sound as possible. Then edit the FULL CONTROL parameters using the Full Control screen, and save those edits to a User Preset.

AGC ON/OFF control activates or defeats the slow AGC prior to the multi-band compressor.

AGC DRIVE control adjusts signal level going into the slow 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 Multi-Band structure is the sum of the gain reduction in the AGC and the gain reduction in the multi-band compressor. The total system gain reduction determines how much the loudness of quiet passages will be increased (and, therefore, how consistent overall loudness will be). It is determined by the setting of the AGC DRIVE control, by the level at which the console VU meter or PPM is peaked, and by the setting of the MB DRIVE (compressor) control.

AGC REL (AGC Release) control provides an adjustable range from 0.5dB/second (slow) to 20dB/second (fast). The increase in density caused by setting the AGC REL control to fast settings sounds different than the increase in density caused by setting the MB REL control to fast, and you can trade the two off to produce different effects.

Unless it is purposely speeded-up (with the AGC MB REL control), the automatic gain control (AGC) that occurs in the AGC prior to the multi-band compressor makes audio levels more consistent without significantly altering texture. Then the multi-band compression and associated multi-band 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 multi-band compressor is sped up.
- Moderate AGC + heavy compression (particularly with a fast multi-band 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 Multi-Band structure’s controls.

AGC B CPL (AGC Bass Coupling) determines the amount of bass coupling in the two-band AGC. Because the AGC is generally operated slowly, one usually sets this control at 80% or higher to prevent the AGC from significantly changing the frequency balance of the program material. “Automatic equalization” is ordinarily done in the two-band compressor section following the AGC. For a further discussion see **BASS CPL** on page 3-24.

BASS CLIP threshold controls Orban’s patented embedded bass clipper. It is embedded in the multi-band crossover so that harmonics created by clipping are rolled off by part of the crossover filters. The threshold of this clipper is ordinarily set between 4dB and 6dB below the threshold of the final limiter in the processing chain, depending on the setting of the LESS-MORE control in the parent preset upon which you are basing your FULL CONTROL adjustments. This provides headroom for contributions from the other four bands, so that bass transients don’t smash against the look-ahead limiter, causing overt intermodulation distortion between the bass and higher frequency program material.

Some 6200 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 look-ahead limiter threshold) is 0 to –6dB. 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 look-ahead limiter. 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 as a result of extensive listening tests with many types of critical program material.

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

The gate causes the gain reduction in bands 2 and 3 of the multi-band 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 blatant coloration, the gating smoothly preserves the average overall frequency response "tilt" of the multi-band compressor, broadly maintaining the "automatic equalization" curve it generates for a given piece of program material.

> **Note:** If the GATE THR (Gate Threshold) control is turned off, the DJ BASS control (in the Equalization screen) is disabled.

MB DRIVE (Multi-Band Drive) control adjusts the signal level going into the multi-band compressor, and therefore determines the average amount of gain reduction in the multi-band compressor. Range is 25dB.

Adjust the MB DRIVE control to your taste and format requirements. Used lightly with a slow or mslow (medium-slow) multi-band release time, the multi-band compressor produces an open, re-equalized sound. The multi-band compressor can increase audio density when operated at fast or mfast (medium-fast) release times because it acts more and more like a fast limiter (not a compressor) as the release time is shortened. With fast and mfast (medium-fast) release times, density also increases when you increase the drive level into the multi-band 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 REL (Multi-Band 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 GATE THR (Gate Threshold) control to "freeze" the gain when the input gets quiet and/or by activating the single-ended noise reduction.

When the release time of the multi-band compressor is set to fast, or mfast, 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 multi-band compressor drive will no longer yield more loudness, and will simply degrade the punch and definition of the sound.

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

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

MB REL (Multi-Band Release) control can be switched to fast, mfast (medium-fast), mslow (medium-slow), and slow.

Slow produces 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 setting produces an unprocessed sound with a nice sense of dynamic range. With these settings, the Multi-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.

Medium Slow is 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.

Medium Fast is 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.

Fast 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 multi-band 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 multi-band 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, and will be less dense and less fatiguing when the radio is played loudly.

B2>B1 CPL (Band 2 to Band 1 Coupling) control determines the extent to which the gain of band 1 (below 100Hz) is determined by and follows the gain of band 2 (centered at 400Hz). Set towards 100% (fully coupled) it reduces the amount of dynamic bass boost, preventing unnatural bass boost in light pop and talk formats. Set towards 0% (independent), it permits frequencies below 100Hz (the "slam" region) to have maximum impact in modern rock, urban, dance, rap, and other music where bass punch is crucial. The default setting is 30%.

B3>B4 CPL (Band 3 to Band 4 Coupling) control determines the extent to which the gain of band 4 (centered at 3.7kHz) is determined by and follows the gain of band 3 (centered at 1kHz). 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.

B4>B5 CPL (Band 4 to Band 5 Coupling) control determines the extent to which the gain of band 5 (above 6.2kHz) is determined by and follows the gain of band 4 (centered at 3.7kHz). Set towards 100% (fully coupled) it reduces the amount of dynamic HF boost, preventing unnatural HF boost in light pop and instrumental formats.

> When combined with the B3>B4 CPL control, this control can adjust the multi-band processing to be anything from full five-band to quasi-three-band processing. The two upper coupling controls are useful when you want to control the amount of overall “automatic equalization” in the midrange and high frequency region. This can prevent the sound from becoming strident or overly bright and is particularly useful with formats designed to attract a female audience. The INSTRUMENTAL and POP presets use large amounts of coupling for this reason.

B5 THR (Band 5 Compression Threshold) control determines the threshold of compression in band 5 (6.2kHz and above).

The Factory Presets were tuned to prevent sibilance on live voice from becoming too harsh-sounding. However, this requires setting the threshold of the band 5 compressor quite low so it can act as a de-esser. A side effect is that some brightly-mixed music can be audibly rolled off by the action of band 5.

If you are willing put some effort into tuning your microphone channel to tame sibilance at the source, then you can achieve a brighter sound on-air by increasing the setting of the band 5 threshold (B5 THR) control. A number of manufacturers make all-in-one microphone processors that include de-essers. If you use one of these with the de-esser set aggressively, you should be able to advance the B5 THR control by several dB without causing any problems. Listen to sibilance on live voice when making your final decision; make sure that it does not sound unpleasantly harsh or “spitty.”

> dbx makes an inexpensive, good sounding mic processor called the 286A. It contains a mic preamp, a compressor, a de-esser, a dynamic high frequency enhancer, and a low frequency equalizer. Its only potential drawbacks are that it has an unbalanced output and no special RFI suppression, so it is best suited for studio sites that are not co-located with high-powered transmitters.

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

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

Ordinarily, the gating on the AGC and multi-band 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 DWNEXP THR control of the noise reduction system to complement the program material you are processing. The DWNEXP THR should be set higher when the input is noisy and lower when the input is relatively quiet. The best way to adjust the DWNEXP THR control is to start with the control set very high. Reduce the control setting while watching the gain reduction meters. Eventually, you will see the gain increase in sync with the program. Go further until you begin to hear noise modulation — a puffing or breathing sound (the input noise) in sync with the input program material. Set the DWNEXP THR control higher until you can no longer hear the noise modulation. This is the best setting.

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

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

FINAL LIMT (Final Limit) control adjusts the level of the audio driving the look-ahead limiter OPTIMOD-DAB uses to control fast peaks, and then adjusts the peak-to-average ratio. The loudness/distortion trade-off is primarily determined by the FINAL LIMT control.

Turning up the FINAL LIMT control drives the look-ahead limiter harder, reducing the peak-to-average ratio, and increasing the loudness on the air. When the amount of limiting is increased, the audible intermodulation distortion caused by limiting is increased. Lower settings reduce loudness, of course, but result in a cleaner sound.

If the MB REL control is set to fast or mfast (medium-fast), intermodulation distortion in the look-ahead limiter will increase as the MB DRIVE control is advanced, and the FINAL LIMT it control may have to be turned down to compensate. To best understand how to make loudness/distortion trade-offs, perhaps the wisest thing to do is to recall a factory multi-band preset, and then to adjust the LESS-MORE control to several settings throughout its range. At each setting of the LESS-MORE control, examine the settings of the MB DRIVE control, the FINAL LIMT control, and the FINAL CLIPPING DRIVE control. (You can see them by calling up the Full Control screen by pressing the FULL CONTROL soft key.) 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.

Test Modes

The 6200 has a built-in test tone generator to allow system testing. It also has a bypass preset with adjustable gain. To access these modes, first press the *Setup* button, and then press the TEST soft key. The table below shows the test modes available in detail:

SETUP: TEST				
Parameter Labels	**Units**	**Default**	**Range (CCW to CW)**	**Step**
MODE	---	operate	operate, bypass, tone	---
BYPASS GAIN	dB	0.0	–18 … +15	1
TONE FREQ	Hz	400	**For DO RATE = 44.1 or 48kHz:** 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, 10000, 12500, 15000, 16000, 20000 **For DO RATE = 32kHz:** 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, 10000, 12500, 15000	LOG
TONE LVL	%	100	0 … 100	1
TONE CHAN	---	l+r	l+r, l–r, l; r	---

Table 3-6: Test Modes

Using the 6200 for Mastering

6200 can be a useful tool for mastering applications in the professional audio industry, such as preparation of equalized, level-controlled, peak limited CD masters. We have frequently used the 6200 in this context, achieving excellent results.

Because of their broadcast origins, most of the 6200's presets provide more processing than would ordinarily be required for mastering. In addition, we would expect that the mastering engineer would want to carefully tweak a preset to complement the program material being mastered.

You can't create a preset "from scratch"; it must always be modified from an existing preset. Each preset has an "easy adjustment" facility called LESS-MORE, which is a one-knob provision for turning the amount of processing up or down.

Systematically, the following is a good method for creating mastering presets. It assumes that you have already set the INPUT REF level and INPUT CLIP level controls to complement your operating levels. (See Chapter 2.)

A) Decide whether you are going to use two-band or five-band processing.

> Two-band processing retains any fixed equalization originally applied to the program (except for a mild amount of dynamic adjustment to bass below 150Hz); five-band processing performs an "automatic re-equalization" function. Both flavors of processing can be extremely smooth and unobtrusive.

B) If you are going to use two-band processing, recall the 2B CLASSICAL preset. If you are going to use five-band processing, recall the POP MEDIUM preset.

C) Press the MODIFY button, and adjust LESS-MORE to 1.0 (the lowest setting).

> To change a setting, you must hold down the soft key while turning the rotary knob.

D) Press the FULL CONTROL button.

> This allows you to access the individual processing settings. The NEXT and PREV buttons flash whenever you can push them to scroll the display horizontally, accessing more settings.

E) If you are using five-band processing, set the AGC to OFF (unless you need a very large amount of compression). If you have chosen the CLASSICAL preset for two-band processing, the AGC will already be off.

F) Unless you will be using a large amount of compression for special applications, set the GATE THR to OFF.

G) Adjust the 2B DRIVE control (two-band) or MB DRIVE control (five-band) to achieve the desired amount of multiband gain reduction.

H) Adjust the release time control (2B RELEASE or MB RELEASE) to achieve the desired compression density.

> The release characteristic is always "automatic" (i.e., multiple time constant), and the RELEASE control simply scales this process. This, combined with multiband operation, makes the compression remarkably resistant to the usual compressor pumping and squashing.

I) Adjust equalization as necessary. (Access it by pressing MODIFY, then the EQ softkey.)

> As discussed above, there is fixed equalization available between the AGC and multiband compressor. In five-band mode, there is also a five-band graphic equalizer after the multiband compressor, which consists of mix controls driven from the output of each band compressor. In five-band mode, any fixed equalization will be partially "undone" by the dynamic re-equalization effect of the multiband compression, so two-band mode is most useful when you are relying on the 6200's fixed EQ, or on external EQ earlier in the signal path. Note also that you can use the BASS COUPLING and HF COUPLING controls to affect the amount of automatic re-equalization performed by the multiband compression. As you set these controls closer to

100%, they permit less and less dynamic LF and HF program-adaptive boost. If you feel that the dynamic re-equalization is not producing enough brightness when the program material lacks high frequencies, you should turn the BAND 3>4 and BAND 4>5 COUPLING closer to 0%. Similarly, if weak bass is not sufficiently boosted, turn the BAND 2>1 COUPLING closer to 0%.

J) Set the amount of peak limiting with the FINAL LIMT control.

In general, the less peak limiting you use, the better sounding the result will be. However, if your client demands a "loud" CD, the 6200's look-ahead peak limiter is a *very* powerful tool for achieving this with minimum distortion or other side effects. Nevertheless, be aware that this function is not like some familiar "look-ahead" limiters. The release time is in the order of a few milliseconds and is not user adjustable. The purpose of the limiter is *only* to limit peaks that pass through the earlier compressors because of their finite attack times. Functionally it is used like a peak clipper, but it has vastly reduced modulation distortion by comparison to a clipper, whether "soft knee" or "hard knee."

K) Adjust the BASS CLIP control to complement the amount of final limiting.

For most mastering applications, you can set it at "0dB," which essentially defeats it. However, if you hear pumping or distortion in the look-ahead limiter caused by heavy bass transients, you can reduce this effect by setting the BASS CLIP to a lower level. (The BASS CLIP control is calibrated in "dB below the final limiter threshold.")

L) Save your preset in one of 32 user locations.

Press *ESCAPE* until you see the "home screen." It will say "on air: XXXXX." The soft key to the far right will be labeled SAVE PRESET. Press it and follow the on-screen instructions. (Once you have created one "mastering" preset, you can edit it to create others and save them in different locations.)

M) For a 44.1kHz output sample rate, set the digital output level to –0.5dBfs; this will prevent overshoots caused by sample rate conversion. For a 48kHz output sample rate, set the digital output level to –0.1dBfs.

At 44.1kHz, the output samples are not exactly the same ones that the look-ahead limiter controlled at the internal 48kHz sample rate, so slight overshoot can occur. At 48kHz output sample rate, overshoot will be less than 0.1dB.

Limitations in Mastering Applications

The 6200 was designed for the digital broadcast market (DAB, DTV, and webcasting), and therefore has some limitations in mastering applications. The main one is lack of individual compressor threshold controls (except on band #5) and ratio controls. The compressors always operate with very high ratios. While this doesn't give the usual "squashed" sound associated with high ratios in many compressors, it nevertheless removes long-term dynamic contrasts. For this reason, we suspect that most program material will work best with 5dB or less gain reduction so that dynamic contrasts will not be severely changed.

The thresholds of the bands in the five-band compressor have been designed to produce a sane frequency balance with material commonly broadcast, including both speech and music. Broadcast compressors must work well with any material thrown at them, without adjustment. Therefore, the hard-wired threshold settings are likely to be acceptable. Further, the band 5 threshold *is* adjustable, so you can use it to make the brightness/sibilance tradeoff, as discussed above. You can also adjust the band OUT controls (under the EQ menu) to make incremental adjustments, although this will also affect the below-threshold EQ.

The fixed equalization was designed to complement material that was already produced to commercial standards. The midrange and HF parametric equalizers are quite general, permitting boosts and cuts with a variety of bandwidths and center frequencies. However, the bass equalizers can only boost, not cut, and bass cuts are sometimes needed in mastering. Small amounts of cut (up to 3dB) are available from the BAND 1 OUTPUT MIX control, and the BASS COUPLING controls can limit potential bass boost. Further, the multiband compression will usually control excessive bass automatically. If you need more control than this, we suggest using your favorite external equalizer before the 6200.

A final limitation is lack of a convenient bypass facility for comparing “processed” and “unprocessed” sounds. There are two reasonable workarounds.

The first uses the 6200’s TEST BYPASS facility. Access this by pressing SETUP, then TEST. The MODE soft key allows you to toggle between OPERATE and BYPASS. You can set the bypass gain with the BYPASS GAIN soft key, immediately to the right of the MODE soft key. The disadvantage of this is that TEST BYPASS loads new code into the 6200’s DSPs, so there is about a one second mute each time you toggle. (If you expect input levels to be close to full scale, you will clip internally if you specify a BYPASS GAIN greater than 0dB.)

The second involves creating a preset with no processing. (This does not mute when toggled.) Recall 2B CLASSICAL and modify it to a LESS-MORE of 1. Turn off the gate, and back off the 2B DRIVE control to achieve no gain reduction. Adjust the FINAL LIMT control to make up the gain. Be sure that there is no gain reduction in the final limiter (as indicated on its gain reduction meters), even if this means that this preset is significantly quieter than your working preset. This will accurately reflect the amount of loudness you are achieving by using the working preset.

When you are satisfied, save this preset, naming it “BYPASS.” You can then use the RECALL NEXT softkey (under the RECALL menu) to toggle between the BYPASS preset and your working preset.

Section 4 Maintenance

Routine Maintenance

The 6200 OPTIMOD-DAB 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 6200's output.

A good ear will pick up many faults. Familiarize yourself with the "sound" of the 6200 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 6200 is at fault. The troubleshooting information in Section 5 will help you determine if the problem is with OPTIMOD-DAB 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 6200 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 silk-screened lettering (99% isopropyl alcohol can be safely used).

Getting Inside the Chassis

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 6200 and remove it from the rack.

Be sure power is disconnected before removing the cover.

Warning: Hazardous voltage is exposed with the unit open and the power 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 Display Assembly. (Model 6200 OPTIMOD-DAB)

[Refer to step 3 to remove 6200S Display board.]

A) Detach the four ribbon cables that connect the display board assembly to the main board and power supply board.

a) First, identify the four cables:

A larger white flat cable connecting to jumper J102 on the main board;

A gray ribbon cable connecting to a DIP header at jumper J103 on the main board.

A smaller white flat cable connecting the display assembly to the power supply at jumper J202.

A two-wire cable connecting the display assembly to the power supply at jumper J201.

b) Gently lift each cable up from where it connects to its jumper, so that the jumper pins unseat without bending or breaking.

If present, remove the white fastener that ties the two-wire cable next to the large black cables of J4.

B) Detach the front panel from the unit.

a) Disconnect the three-wire cable at the back of the encoder.

b) Detach the ground lug that connects the panel's ground wire to the chassis.

Use a 1/4-inch nut driver or needle-nose pliers.

c) Remove the front panel.

The front panel is held in place by four ball studs at each corner. It should snap off, with a little force.

C) Remove the ten gold-colored screws and washers that connect the display board to the front of the chassis.

Use a screwdriver.

D) Remove the display board assembly by removing the tape from the top front edge of the chassis, so that the white ribbon cables are no longer attached to the chassis and the display panel is free.

Do not remove the tape from the white ribbon cables.

3. Removing the Display Assembly. (Model 6200S)

[Refer to step 2 to remove 6200 OPTIMOD-DAB Display board.]

A) Detach the four ribbon cables that connect the display board assembly to the main board and power supply board.

a) First, identify the four cables:

A larger white flat cable connecting to jumper J102 on the main board;

A gray ribbon cable connecting to a DIP header at jumper J103 on the main board.

A smaller white flat cable connecting the display assembly to the power supply at jumper J202.

A two-wire cable connecting the display assembly to the power supply at jumper J201.

b) Gently lift each cable up from where it connects to its jumper, so that the jumper pins unseat without bending or breaking.

If present, remove the white fastener that ties the two-wire cable next to the large black cables of J4.

B) Detach the front panel display assembly from the unit.

a) Remove the six rack ear screws on the sides of the unit.

C) Remove the display boards from the front panel assembly.

a) Remove the four nuts that secure the rear cover of the display assembly to the front panel and then remove the rear cover.

b) Remove the eight screws that mount the rear display board to the front panel.

c) Disconnect the rear display board from the front display board.

d) Remove the two screws that hold the front display board to the front panel.

4. Removing the Main Board.

A) If you have not done so yet, remove the top cover (Step 1, above).

B) If you have not done so yet, remove the ribbon cables that connect the display assembly to the main board (at jumpers J102 and J103).

C) Disconnect the Remote Interface assembly from the J101 connector. (Model 6200 OPTIMOD-DAB only)

D) Remove the four hex nuts holding the two RS-232 connectors to the chassis, using a 3/16-inch hex nut driver.

E) Unlock all 7 XLR connectors, using a jeweler's screwdriver; engage the locking mechanism (in the center of the triangle formed by the three contact pins) and turn counterclockwise until the XLR is no longer attached.

F) Remove the two ribbon cables that connect the power supply to the main board at jumpers J900 and J901. (If present, you must first remove the black retainer clips.)

G) Remove the ten #1 Phillips screws (and their washers) that connect the main board to the chassis.

H) Carefully pull the main board forward to clear XLRs from their housing and then out of the chassis.

5. Removing the Power Supply Board.

A) If you have not done so yet, remove the top cover (Step 1, above).

B) If you have not done so yet, remove the ribbon cables that connect the display assembly to the power supply board (at jumpers J201 and J202), as well as the two cables that connect the main board to the power supply (at jumpers J900 and J901).

C) Remove the seven #1 Phillips screws (and their washers) holding the heat sink to the side of the chassis.

D) Remove the nut and star washer from the ground wire with a ¼-inch nut driver.

E) Remove the two Phillips screws (and matching washers) that hold the IEC connector to the chassis.

F) Remove the two plugs (J4 and J6) that connect the power supply board to the transformer.

> If present, remove the white fasteners that tie the two cables to the power supply board.

G) Remove the three Phillips screws holding the power supply board to the main chassis.

H) Carefully lift the power supply board up.

6. Reattaching the Power Supply Board.

A) Set power supply board into main chassis, so that it aligns with its mounting holes.

B) Replace the two Phillips screws that hold the IEC connector.

C) Replace the seven #1 Phillips screws that hold the heat sink to the side of the chassis.

D) Replace the ground wire nut.

E) Replace the three Phillips screws that hold the power supply board to the main chassis.

F) Reattach the two plugs (J4 and J6) that connect the power supply board to the transformer.

G) If the display board is installed, reattach the two ribbon cables that connect the display board to the power supply board (at jumpers J202 and J201).

H) If the main board is installed, reattach the two ribbon cables that connect the main board to the power supply (J900 and J901).

7. Replacing the Main Board.

A) Set the main board into the main chassis, so that it aligns with its mounting holes.

B) Reattach all 7 XLR analog connectors, using a jeweler's screwdriver.

C) Reattach the two RS-232 connectors to the chassis with the four hex nuts.

Use a 3/16-inch hex nut driver.

D) Reconnect the Remote Interface assembly to the J101 connector. (Model 6200 OPTIMOD-DAB only)

E) Replace the ten #1 Phillips screws that connect the main board to the chassis.

F) If the power supply board is installed, reattach the two ribbon cables that connect the main board to the power supply (J900 and J901).

G) If the display board is installed, reattach the two ribbon cables that connect the main board to the display board (J102 and J103).

8. Replacing the Display Board. (Model 6200 OPTIMOD-DAB)

[Refer to step 9 to replace 6200S Display board.]

A) Set the display assembly in place so that it aligns with its mounting holes.

B) Replace the ten gold-colored screws that connect the display board to the front of the chassis.

C) Reattach the four cables that connect the display board to the main board and power supply board.

The larger white flat cable connects to jumper J102 on the main board.

The gray ribbon cable connects to a DIP header at jumper J103 on the main board.

The smaller white flat cable connects the display assembly to the power supply at jumper J202.

The two-wire cable connects the display assembly to the power supply at jumper J201.

D) Attach the front panel assembly to the unit.

a) Line up the plastic front panel and snap it back on, making sure each key pad button feeds through its respective hole properly.

b) Reattach the ground lug that connects the panel's ground wire to the chassis.

Use a 1/4-inch nut driver or needle-nose pliers.

c) Reconnect the three-wire cable at the back of the encoder. (6200 OPTIMOD-DAB only)

9. Replacing the Display Board. (Model 6200S)

[Refer to step 8 to replace 6200 OPTIMOD-DAB Display board.]

A) Reattach the display boards to the front panel assembly.

a) Line up the front display board to the front panel, making sure each pad button feeds through its respective hole properly.

b) Reattach the two screws that hold the front display board to the front panel.

c) Reconnect the rear display board to the front display board.

d) Reattach the eight screws that mount the rear display board to the front panel.

e) Reconnect the assembly rear cover and rack ears to the front panel by attaching the four nuts that secure the rear cover of the display assembly to the front panel.

f) Reattach the six rack ear screws that mount the front panel to the main chassis.

B) Reattach the four ribbon cables that connect the display board assembly to the main board and power supply board.

A larger white flat cable connecting to jumper J102 on the main board;

A gray ribbon cable connecting to a DIP header at jumper J103 on the main board.

A smaller white flat cable connecting the display assembly to the power supply at jumper J202.

A two-wire cable connecting the display assembly to the power supply at jumper J201.

10. Replacing the Top Cover.

A) Place top on unit and reattach the twenty Phillips screws. (Be careful not to pinch the ribbon cables or the two-wire backlight power connector.)

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

  (Tektronix 5L4N plug-in with 5111 bistable storage mainframe, or similar. Alternatively, a sweep generator with 50-15,000Hz 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 5MHz or greater vertical bandwidth.

  It is assumed that the technician is thoroughly familiar with the operation of this equipment.

- Two 620Ω ±5% resistors.

- Optional: Audio Precision System 1 (without digital option) or System 2 (for digital tests).

This procedure is useful for detecting and diagnosing problems with the 6200'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 6200 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 which could develop in the 6200 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: All levels are the same regardless of whether the output is balanced or unbalanced. To unbalance an output, connect either pin 2 or pin 3 of the XLR to pin 1 (ground). The remaining pin is "hot."

All levels refer to balanced outputs. These readings are 6dB higher than unbalanced readings. For unbalanced output, measure between pin 1 (ground) and pin 2 (hot).

Note: All analog output measurements must be taken with a 600Ω resistor tied between pin 2 and 3.

1. Prepare the unit.

A) Use the front panel controls to set the 6200's software controls to their default settings, as follows:

> **Note:** You can automatically set default settings by holding down both the *Escape* and *Setup* buttons simultaneously, then applying power and selecting RESTORE.

Preset

GENERAL-OPEN

I/O CALIB (ANLG IN CALIB)

INPUT	analog
AI REF VU	+4.0dBu
AI REF PPM	+12.0dBu
AI CLIP	+20.0dBu
R CH BAL	0.0 dB

I/O CALIB (DIG IN CALIB)

INPUT	analog
DI REF VU	–16.0dBFS
DI REF PPM	–8.0dBFS
DI MODE	normal
R CH BAL	0.0 dB
USER BITS	block

IO CALIB (ANLG OUT CALIB, for both analog outputs)

MON OUT	post lim

IO CALIB (DIG OUT CALIB)

DO 100%	0.0dBFS
DO RATE	48kHz
DO SYNC	internal
SYNC DELAY	off

WORD LEN	20bits
DITHER	off
STAT BITS	aes/ebu

TEST

MODE	operate
BYPASS GAIN	0 dB
TONE FREQ	400Hz
TONE LVL	100%
TONE CHAN	l+r

Additional Setup Settings

MAX LPF	20.0 kHz
ST CHASSIS	no
MONO/ST	stereo

B) Set the GND LIFT switch to the earth ground symbol setting (down position), so that ground is connected.

C) Press *Setup* button to access Setup menu.

D) Press TEST soft key to access Test menu.

E) Recall bypass preset: Press MODE soft key, use the control knob to scroll to bypass, then release the MODE soft key.

Bypass defeats all compression, limiting, and program equalization, but retains MAX LPF frequency setting.

2. Test Digital +5 volt supply (Power Supply Board).

A) Measure the +5 volt supply with the DVM. Verify the presence of +5 volts (±0.25V).

The +5 volt digital supply appears between TP6 and ground test point TP9 on the Power Supply Board.

3. Test Analog ±15 volt supply (Power Supply Board).

A) Measure the +15 volt supply with the DVM. Verify the presence of +15 volts (±0.75V).

The +15 volt supply appears between TP1 and ground test point TP3 on the Power Supply Board.

B) Using the oscilloscope, measure the total ripple and noise on the +15 volt supply.

The ripple and noise should not exceed 50mVp-p.

C) Measure the –15 volt supply with the DVM. Verify the presence of –15 volts (±0.75V).

The –15 volt supply appears between TP4 and ground test point TP3 on the Power Supply Board.

D) Using the oscilloscope, measure the total ripple and noise on the –15 volt supply.

The ripple and noise should not exceed 50mVp-p.

4. Test Analog ±5 volt supply (Power Supply Board).

A) Measure the analog +5 volt supply with the DVM. Verify the presence of +5 volts (±0.25V).

The analog +5 volt supply appears between TP2 and ground test point TP3 on the Power Supply Board.

B) Using the oscilloscope, measure the total ripple and noise on the +5 volt supply.

The ripple and noise should not exceed 50mV.

C) Measure the analog –5 volt supply with the DVM. Verify the presence of –5 volts (±0.25V).

The analog –5 volt supply appears between TP5 and ground test point TP3 on the Power Supply Board.

D) Using the oscilloscope, measure the total ripple and noise on the –5 volt supply.

The ripple and noise should not exceed 50mV.

5. Check Analog Input Clip Level.

A) Verify 6200 software controls are set to their default settings. (Refer to page 4-9.)

B) Press *Setup* button to re-access Setup menu.

C) Press TEST soft key to access Test menu.

D) Recall bypass preset: Press MODE soft key, use the control knob to scroll to bypass, then release the MODE soft key.

Bypass defeats all compression, limiting, and program equalization, but retains MAX LPF frequency setting.

E) Connect the oscillator to the Left Input XLR connector.

F) Inject the Left Input XLR connector with a level of +20dBu (matching the AI CLIP level setting) at 1kHz.

G) Connect the audio analyzer to the 6200's Left Monitor (output) XLR connector.

H) Verify that the output is +12.6dBu ±1.5dB.

I) Apply 0dBu to the input of the unit.

J) Vary the AI CLIP level control from +27.0dBu to +5.0dBu and verify the following Left Monitor (output) level readings:

AI CLIP Setting	Left Monitor Output Level
+20dBu	~–7.4dBu
+27dBu	~–14.3dBu
+10dBu	~+2.6dBu
+5dBu	~+7.6dBu

Readings can vary as much as ±1.5dB. Verify the output increments smoothly through its range. If this is not the case (i.e., the output level jumps erratically to some large or small level), the gain stages of the SSM2017 IC are not functioning correctly.

K) Repeat steps 5-A through 5-J for the Right Input XLR connector.

6. Check frequency response of Analog I/O.

A) Verify 6200 software controls are set to their default settings. (Refer to page 4-9.)

B) Press *Setup* button to re-access Setup menu.

C) Press TEST soft key to access Test menu.

D) Recall bypass preset: Press MODE soft key, use the control knob to scroll to bypass, then release the MODE soft key.

Bypass defeats all compression, limiting, and program equalization, but retains MAX LPF frequency setting.

E) Connect the oscillator to the Left Input XLR connector.

F) Inject the Left Input XLR connector with a level of +7.3dBu with the oscillator set to 1kHz.

G) Connect the audio analyzer to the 6200's Left Monitor (output) XLR connector.

H) Verify a level of 0dBu ±1dB. Use this level as the reference level.

I) Measure and verify output signal level from 30Hz to 20kHz is within ±0.4dB of the reference level.

The maximum deviation between the maximum and minimum output should be within ±0.8dB.

J) Repeat steps 6-A through 6-I for the right channel.

7. Check distortion performance of Analog I/O.

A) Verify 6200 software controls are set to their default settings. (Refer to page 4-9.)

B) Press *Setup* button to re-access Setup menu.

C) Press TEST soft key to access Test menu.

D) Recall bypass preset: Press MODE soft key, use the control knob to scroll to bypass, then release the MODE soft key.

Bypass defeats all compression, limiting, and program equalization, but retains MAX LPF frequency setting.

E) Connect a THD analyzer to the Left Input XLR connector. Set the THD analyzer's bandwidth to 22kHz.

F) Connect the oscillator to the Left Input XLR connector.

G) Inject the Left Input with a level of +20dBu at 1kHz.

H) Measure and verify THD+N is below 0.01% from 30Hz to 20kHz (0.005% typical).

I) Disconnect the THD analyzer from the Left Monitor (output) XLR connector, and connect to the Right Monitor (output) XLR connector.

J) Repeat the above measurements for Right Monitor (output) XLR connector.

K) Disconnect the oscillator and THD analyzer from the 6200.

8. Verify Digital Receiver and Transmitter Sync.

A) Verify 6200 software controls are set to their default settings. (Refer to page 4-9.)

B) Press *Setup* button to re-access Setup menu.

C) Press IO CALIB soft key to access I/O CALIB menu.

D) Press DIG IN CALIB soft key.

E) Set INPUT to digital: Press INPUT soft key, use the control knob to select digital, then release the INPUT soft key.

F) Press *Escape*, then press the DIG OUT CALIB soft key.

G) Set DO SYNC to external: Press DO SYNC soft key, use the control knob to select external, then release the DO SYNC soft key.

H) Connect the digital source generator to the AES/EBU Digital Input XLR connector of the 6200.

I) Inject the Digital Input with a sample rate of 32kHz, 44.1kHz and 48kHz. As the input sample rate changes, also change the DO RATE to match the input sample rate: Press DO RATE soft key, use the control knob to select a sample rate, then release the DO RATE soft key.

J) View TP606 (FSYNC) on Channel 1 of an oscilloscope, then view TP610 (SYNC INPUT) on Channel 2 of an oscilloscope. Verify that both signals are synced together at all sample rates.

K) Set DO SYNC to internal: Press DO SYNC soft key, use the control knob to select internal, then release the DO SYNC soft key.

L) Verify that both signals are not synced together at any combination of sample rates and DO RATE settings: Press DO RATE soft key, use the control knob to select a sample rate, then release the DO RATE soft key.

M) Disconnect the digital source generator from the 6200.

9. Test Digital Receiver Sample Rate Lock.

A) Verify 6200 software controls are set to their default settings. (Refer to page 4-9.)

B) Press *Setup* button to re-access Setup menu.

C) Press I/O CALIB soft key to access I/O CALIB menu.

D) Press DIG IN CALIB soft key.

E) Set INPUT to digital: Press INPUT soft key, use the control knob to select digital, then release the INPUT soft key.

F) Connect the digital source generator to the AES/EBU Digital Input XLR connector of the 6200.

G) Inject the Digital Input with a sample rate of 32kHz, 44.1kHz and 48kHz. Use 24-bit words.

H) Measure the frequency of the AES/EBU receiver at TP614. Verify that the measured frequency corresponds with the input sample rate.

I) Connect AES/EBU digital source to Sync XLR Input connector and repeat steps 9-C through 9-H using TP618 to verify sample rate lock.

J) Disconnect the digital source generator from the 6200.

10. Test Digital Transmitter Sample Rate.

A) Verify 6200 software controls are set to their default settings. (Refer to page 4-9.)

B) Press *Setup* button to re-access Setup menu.

C) Press IO CALIB soft key to access I/O CALIB menu.

D) Press DIG IN CALIB soft key.

E) Set INPUT to digital: Press INPUT soft key, use the control knob to select digital, then release the INPUT soft key.

F) Change the DO RATE to 32kHz, 44.1kHz and 48kHz, and verify that the frequencies measured at TP606 (FSYNC) follow the chart below within given tolerances: Press DO RATE soft key, use the control knob to select a sample rate, then release the DO RATE soft key.

Sample Rate	Tolerance (PPM)	Tolerance (Hz)
32kHz	50 PPM	±1.60 Hz
44.1kHz	100 PPM	±4.41 Hz
48kHz	50 PPM	±2.40 Hz

11. Check frequency response of Digital I/O

A) Verify 6200 software controls are set to their default settings. (Refer to page 4-9.)

B) Press *Setup* button to re-access Setup menu.

C) Press IO CALIB soft key to access I/O CALIB menu.

D) Press DIG IN CALIB soft key.

E) Set INPUT to digital: Press INPUT soft key, use the control knob to select digital, then release the INPUT soft key.

F) Press *Setup* button to re-access Setup menu.

G) Press TEST soft key to access Test menu.

H) Recall bypass preset: Press MODE soft key, use the control knob to scroll to bypass, then release the MODE soft key.

> Bypass defeats all compression, limiting, and program equalization, but retains MAX LPF frequency setting.

I) Connect the digital source generator to the AES/EBU Digital Input XLR connector of the 6200.

J) Inject the digital input with a level of −4dBFS at 1kHz (48kHz sample rate).

> Measure the levels present at the AES/EBU Digital Output XLR connector and verify a level of −4.1dBFS ±0.2dB. Use the level measured as the reference level.

K) Verify that frequency response between 30Hz and 20kHz is within ±0.03dB of the reference level.

L) Disconnect the digital source generator from the 6200.

12. Check noise and distortion performance of Digital I/O.

A) Verify 6200 software controls are set to their default settings. (Refer to page 4-9.)

B) Press *Setup* button to re-access Setup menu.

C) Press IO CALIB soft key to access I/O CALIB menu.

D) Press DIG IN CALIB soft key.

E) Set INPUT to digital: Press INPUT soft key, use the control knob to select digital, then release the INPUT soft key.

F) Press *Setup* button to re-access Setup menu.

G) Press TEST soft key to access Test menu.

H) Recall bypass preset: Press MODE soft key, use the control knob to scroll to bypass, then release the MODE soft key.

> Bypass defeats all compression, limiting, and program equalization, but retains MAX LPF frequency setting.

I) Connect the digital input of the THD analyzer to the AES/EBU Digital Output XLR connector of the 6200.

J) Connect the digital source generator to the AES/EBU Digital Input XLR connector of the 6200.

K) Inject the Digital Input XLR connector with a level of −1dBFS at 1kHz, 48kHz sample rate, word length of at least 18 bits (noise floor values will be limited by smaller word lengths).

L) Measure and verify THD+N is below 0.01% from 20Hz to 20kHz (0.001% is typical for both audio channels).

M) Disconnect the digital source generator and THD analyzer from the 6200.

13. Return OPTIMOD-DAB to service.

A) Remove the 600Ω resistors connected across the outputs.

B) Recall your normal operating preset.

Section 5
Troubleshooting

Problems and Possible Causes

Always verify that the problem is not the source material being fed to the 6200, or in other parts of the system.

RFI, Hum, Clicks, Or Buzzes

A grounding problem is likely. Review the information on grounding on page 2-10. The 6200 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.

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 6200 ordinarily controls peak modulation to an accuracy of ±1% when operated with 48kHz output sample rate. As explained in Section 1, output sample rate conversion will slightly compromise this control because the peak control occurs with reference to individual sample values at 48kHz. The converted samples no longer have the same peak values as the 48kHz samples, and some values can be slightly higher. However, the overshoot of the converted signal almost never exceeds 0.5dB and is therefore not a significant problem.

Using the analog output will cause similar amounts of overshoot because the samples in the transmitter are not synchronous with the peak-controlled samples in the 6200. Further, analog connections can cause analog-domain overshoot if the connection is not phase linear and has a low-frequency cutoff of greater than 0.15Hz (at –3dB).

Audible Distortion On-Air

Make sure that the problem can be observed on more than one receiver and at several locations.

Verify that the source material at the 6200'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. Advancing the FINAL LIMT control too far will inevitably cause distortion. (Distortion is very probable if gain reduction in the final limiter frequently exceeds 6dB.) Setting the LESS-MORE control beyond “9” will cause audible distortion of some 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 6200's A/D converter will clip and distort. (See page 2-24).

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 6200 reduces this problem with its *compressor gate*, which freezes the gain of the AGC and compressor systems whenever the input noise drops below a level set by the GATE THR (Gate Threshold) control, preventing noise below this level from being further increased.

If you are using the 6200'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 6200 to digitize the input. (This ratio is better than 100dB.)

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 6200 under normal operation. If, in an attempt to build in a “safety factor” or increase headroom, you specify a higher level than this, every 1dB of extra headroom that you gain will be accompanied by a 1dB increase in the 6200's noise floor.

The 6200's AES/EBU input is capable of receiving words of up to 20 bits. A 20-bit word has a dynamic range of approximately 120dB. The 6200'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 6200, the STL's noise level can severely limit the overall noise performance of the system because compression in the 6200 can exaggerate the STL noise. For example, the overload-to-noise ratio of a typical analog microwave STL may only be 70-75dB. 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.

Shrill, Harsh Sound

This problem can be caused by excessively high settings of the HF GAIN control. It can also be caused by excessively high settings of the B5 THR (Band 5 Compression Threshold) control. In this case, you are first likely to notice the problem as harsh sibilance on voice.

System Receiving 6200's Output Will Not Lock

Be sure that the 6200's output sample rate is set to match the sample rate that the driven system expects. Be sure that the 6200's output Stat Bit (Status Bits) control is set to match the standard expected by the driven system (either aes/ebu or spdif).

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

This is normal in operate mode. 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 close to that of program material, promoting a more consistent and well-balanced sound quality.

To pass line-up tones transparently, initiate 6200's built-in bypass from the front panel: Open the Setup Test screen and change the Mode setting to bypass. The 6200 can also generate its own test tones: Change the Mode setting to tone.

Both bypass and tone modes can also be initiated using the 6200's optically-isolated remote interface (model 6200 OPTIMOD-DAB only), or via PC Control. Refer to Section 2, Installation for details.

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 6200, or the 6200 should be temporarily switched to bypass to pass the tones.

General Dissatisfaction with Subjective Sound Quality

The 6200 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.

Excessively Loud Commercials (DTV Applications)

The Two-Band factory presets that use the Loudness Controller set their threshold at an "average" level. If, in any preset, you feel that commercials are too loud by comparison to surrounding program material, set the Loudness Controller to a lower threshold by editing your on-air preset's LC THRESH control and saving it as a user preset. (Also, make sure that the preset you are using has the Loudness Controller activated!)

The five-band structure does not have a Loudness Controller because its “automatic re-equalization” function makes spectral balances (and, therefore, loudness) more consistent without the need for a special loudness controller function.

Inconsistent Levels (DTV Applications)

The release time setting in the two-band or five-band compressor determines the speed at which the 6200 corrects program material with excessively low levels. If the 6200 seems to take too long to correct low levels, speed up the release time of the compressor.

Also, the GATE THR control is set quite high in the DTV-oriented presets to prevent noise, low-level FX, and underscoring from pumping up unnaturally. This means that it is important to set the reference level control (in the appropriate Setup I/O Calib screen) correctly, so that the presets operate in the range for which they were designed. (See Section 2 for instructions on setting these controls — there is one for the analog input and one for the digital input.) If the reference level is set too high, the gate will frequently be on even when program material has normal levels. It will therefore inhibit the ability of the processor to quickly correct material that is slightly too low in level.

Incorrect Lip Sync (DTV Applications)

The 6200 has the ability to add delay so that the total delay through the unit is exactly one frame of various standards, including 24, 25, and 30 frames per second. It is usually wise to set the 6200’s throughput delay so that it is exactly one frame, and then add one frame of video delay to compensate. (The minimum delay through the 6200 is about one-half frame.)

Multi-frame lip sync problems are not caused by the 6200 and cannot be corrected by it.

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 6200'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 6200 — this is printed on the rear panel of the unit.

Telephone:	(1) 510/351-3500	or Write:	Customer Service
			Orban
or Fax:	(1) 510/351-0500		1525 Alvarado Street
			San Leandro, CA 94577 USA

E-Mail: custserv@orban.com

Web: www.orban.com

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, operation procedures and any appropriate troubleshooting suggestions. 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 strongly 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, transportation charges to the factory (which are usually quite nominal) are paid by the customer.

Shipping Instructions

Use the original packing material if it is available. If it is not, use a sturdy, double-walled carton. For Model 6200 OPTIMOD DAB, it should be no smaller than 4″ (H) x 15.5″ (D) x 22″ (W) — 10 cm (H) x 40 cm (D) x 56 cm (W), with a minimum bursting test rating of 200 pounds (91 kg). For Model 6200S, it should be no smaller than 1.75″ (H) x 15.5″ (D) x 22″ (W) — 4.5 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.

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Section 6
Technical Data

Specifications

It is impossible to characterize the listening quality of even the simplest limiter or compressor on the basis of the usual specifications, because such specifications cannot adequately describe the crucial dynamic processes that occur under program conditions. Therefore, the only way to meaningfully evaluate the sound of an audio processor is by subjective listening tests.

Certain specifications are presented here to assure the engineer that they are reasonable, to help plan the installation, and to help make certain comparisons with other processing equipment. Some specifications are for features that are only available on the 6200.

Frequency Response: ±0.1dB, 2-20,000Hz (Bypass mode).

Input/Output Delay: approximately 15 milliseconds. Can be padded to full frame of: 24 milliseconds, 30 fps (33.33ms), 29.97 fps (NTSC color video; 33.37ms), 25 fps (most PAL video; 40ms), 24 fps (film; 41.67ms), and 96 milliseconds.

Internal Filters: 4.0, 4.5-11.5, 12, 13, 14, 15, 20kHz provide anti-aliasing for low sample rate services, such as Eureka-147 (24kHz mode), and the proposed AM IBOC (32kHz) and shortwave IBOC systems.

Internal Sample Rate: 48kHz.

Internal Resolution: 24-bit fixed point (Motorola DSP56009).

Input/Output Resolution: 20 bit, limited by Analog Devices sample rate converters.

Peak Control Accuracy: If output samples are synchronous with internal samples, maximum overshoot of any output sample is 0.1dB. This is true at 48kHz output sample rate. If sample rate conversion after internal processing makes output samples asynchronous with internal samples, output samples can overshoot as much as 1 dB (0.3B typical).

Phase Response: All processing is linear-phase (constant group delay).

Internal Processing: Input→Two-Band defeatable AGC→Four-Band Equalizer →Multiband Compressor→Look-ahead Limiter→Output.

Multiband Compressors: Three available: Five-Band, Two-Band, and Protect, selectable by mute-free crossfade.

Equalizers: Shelving Low Bass Eq, selectable 12dB or 18dB/octave; Shelving Mid Bass Eq; Parametric Midrange Eq with analog-style bell-shaped curves; Parametric High Frequency Eq with analog-style bell-shaped curves

Number of Factory Presets: 26, each with 19-step LESS-MORE control. Presets are fully customizable with FULL CONTROL.

Number of User Presets: 32 memory locations to save customized presets.

Analog Audio Input

Configuration: Stereo.

Impedance: Electronically balanced 600Ω or >10kΩ load impedance, selected using a configuration jumper.

Nominal Input Level: Software adjustable (via AI REF control) –9dBu to +13dBu VU (–1dbu to +21dBu PPM).

Maximum Input Level: +27 dBu, peak.

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

AD Conversion: 24-bit 128X oversampled delta sigma A/D converter with linear phase anti-aliasing filter.

RF Filtering: Effective 0.5-1000MHz.

High Pass Filter: 0.15 Hz.

Analog Audio Output (Monitor)

Note that this output is not intended to be used as a transmitter output, rather it is for monitoring. Nevertheless it has been designed for low noise, distortion, and overshoot and can drive an STL or transmitter in an analog plant. Peak control will not be as good as at the digital output because on-air samples will be asynchronous with peak-controlled internal samples.

The source of this output is switchable between the peak limiter output and the multi-band compressor output. Because most of the delay occurs in the peak limiter, this can make headphone monitoring much more comfortable for on-air talent.

Configuration: Stereo output.

Source Impedance: 30Ω, electronically balanced and floating.

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

Output Level Control: Fixed at +14 dBu peak, with MON OUT source set to “POST LIMiter.” Pre-limiter peaks up to +26 dBu possible. No output level adjustment control is provided.

Signal-to-Noise: 90 dB unweighted signal-to-noise, 20Hz-20kHz (bypass mode).

Distortion: ≤0.01% THD (Bypass mode) 20Hz-20kHz.

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

DA Conversion: 20-bit 128X oversampled D/A converter.

DC Offset: Less than 10mV.

RF Filtering: Effective 0.5-1000MHz.

Digital Audio Input

Configuration: Stereo per AES/EBU-standard (AES3-1992), 20 bits resolution, software selection of stereo, mono from left, mono from right or mono from sum (as source to use as a mono processor). Labeled “AES/EBU Input.”

Sample rate: 32, 44.1 or 48kHz, automatically-selected.

Connector: XLR-type, female, EMI-suppressed. Pin 1 chassis ground, Pins 2 (+) and 3 transformer balanced and floating, 110 ohm impedance.

Status Bits: Input channel status is ignored. Input User bits are optionally passed through to the output.

Input Reference Level: Variable within the range of –10 to –30dBFS (VU) in 0.5dB steps.

RF Filtering: Effective 0.5-1000MHz.

Digital Audio Output

Configuration: Two-channel AES/EBU-standard. (AES3-1992).

Output Level Control: Peak level is adjustable from 0 to –20dBFS in 0.1dB steps. Controlled by software. (See System Settings).

Sample rate: Internal free running at 32 kHz ±10 PPM, 44.1 kHz ±100 PPM, or 48 kHz ±10 PPM, selected in software. Can also be synced to the AES/EBU SYNC input or the AES/EBU DIGITAL input at 32 kHz ± 4%, 44.1 kHz ± 4%, or 48 kHz ± 4%.

Word Length: Selectable 20, 18, or 16 bit. Optional dither can be added, with level adjusted appropriate to word length. This is first-order noise-shaped dither (that is, white TPDF dither of peak amplitude equal to the quantizer step size with noise shaping spectral density of 6dB/octave). It sounds substantially quieter than white triangular PDF dither but, in contrast to more extreme noise-shaped dither, it adds only 3dB unweighted noise by comparison to white PDF dither.

Sync: Internal free running or external per AES11. The "AES/EBU SYNC Input" is the primary source for external sync. The "AES/EBU Input" is an alternate source.

Connector: XLR-type, male, EMI-suppressed. Pin 1 chassis ground, Pins 2 and 3 transformer balanced and floating.

Status/User Bits: The channel status optionally supports the SPDIF or AES/EBU formats. The user bits received from the "AES/EBU Input" are optionally passed through to the output. The 192-bit block structure is used where preamble "Z" indicates the start of block.

Jitter: Less than 10ns rms.

RF Filtering: Effective 0.5-1000MHz.

Digital Input Sync

Configuration: Used for synchronization of the Digital Output signal to an external reference provided at this input. Labeled "AES/EBU Sync Input."

Sampling Rate: 32, 44.1 and 48 kHz.

Connector: XLR-type, female, EMI-suppressed. Pin 1 chassis ground, Pins 2 (+) and 3 transformer balanced and floating, 110 ohm impedance.

Filtering: RFI filtered.

Remote Control Interface

Configuration: Eight (8) inputs, opto-isolated and floating, and two open-collector fault tally outputs.

Inputs Operate: Controlled by software (see System Settings).

Control: Momentary or continuous contact closure, minimum current required is 10mA; maximum permitted current is 30mA, which is achieved with a 9V DC input.

Note: More than one contact closure can be active at once. If one contact is held, and another is closed then the second requested function is activated.

Tally Outputs: Two tally outputs provided on the Remote Interface Connector, indicating Loss of Digital Signal Lock and Loss of Digital Sync Lock.

These tally outputs are provided as two open collectors. A tally condition is defined to be active (the collector conducts to supply common) when the fault condition exists. When the fault condition does not exist, the tally outputs will be high impedance with no pull-up resistor.

Power Supply: Current-limited 9 VDC provided for use with contact closure.

Connector: Mini DB, EMI-suppressed.

Filtering: RFI filtered.

Power

Voltage: U: 100-132 VAC; E: 200-264VAC; J: 89-120VAC, 178-240 VAC; 50VA.

Line Frequency: 50 to 60 Hz.

AC Connector: IEC, detachable 3-wire power cord supplied. AC is EMI-suppressed.

Grounding: Circuit ground is independent of chassis ground; can be isolated or connected with a rear panel switch. There is no ground lug. However, analog power common is available on pin 13 of the Remote Interface DB-25 connector.

Surge and Transient: According to regulatory testing requirements.

Memory Backup: All system parameters and user presets are stored in an E2PROM. No battery is required.

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

Environmental

Operating Temperature Range: Model 6200 OPTIMOD-DAB, 32° to 122°F/0° to 50°C at nominal operating voltages; Model 6200S, 32° to 113°F/0° to 45°C at nominal operating voltages.

Humidity: 0-95% RH, non-condensing.

RFI / EMI: Tested according to Cenelec procedures.

Dimensions (W x H x D): Model 6200 OPTIMOD-DAB, 19″ x 3.5″ x 14.25″/48.3cm x 8.8 cm x 36.2cm, two rack units high; Model 6200S, 19″ x 1.75″ x 13.25″/48.3cm x 4.4cm x 33.7cm, one rack unit high.

Weight: 6200 OPTIMOD-DAB, 14 lbs/6.4kg; 6200S, 13 lbs/6kg.

Shipping Weight: 6200 OPTIMOD-DAB, 19 lbs/8.7kg; 6200S, 18 lbs/8.2kg.

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 circuits used in the 6200. It starts with an overview of the 6200 system, identifying circuit sections and describing their purpose. Then each section is treated in detail by first giving an overview of the circuits followed by a component-by-component description. Keywords are highlighted throughout the circuit descriptions to help you quickly locate the information you need.

Overview

The block diagram on page 6-32 illustrates the following overview of 6200 circuit sections.

The 12.288MHz Oscillator and System Clocking section provides the various clocks needed by the control, I/O and DSP circuits to carry out their functions.

The Control Circuits administrate control of the 6200 system.

The User Control Interface and LED Display Circuits section includes the connector, RF-filtering, and circuitry for the remote control inputs and RS-232 interface. It also includes circuitry for the front panel pushbutton switches, LED control status indicators, and LED Meters. The LED Meters measure various 6200 signal levels and display the results on ten front panel 10-segment LED meters.

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 Output Circuits include the connectors and RF-filtering for the analog and digital audio outputs, and the circuitry to interface the digital processing to these outputs.

The DSP Circuits implement the bypass, test tone, and audio processing using digital signal processing.

The Power Supply provides power for all 6200 circuit sections.

12.288MHz Oscillator and System Clocking

A synchronous clocking scheme is used on the 6200 to eliminate any asynchronous clocks operating in the sensitive regions of the input A/D converter. A single 12.288MHz crystal oscillator provides the timing reference for all system digital clock signals. The only clocks that run asynchronous to this clock are the AES/EBU digital audio input related clocks, the 11.2896MHz free running crystal clock oscillator providing the 44.1kHz AES/EBU output sample rate, and the 17.000MHz free running crystal clock oscillator used as the master clock for the sample rate converters. These do not fall within a sensitive region of the A/D. Synchronous counters are used to divide the 12.288MHz clock to produce the various clock signals for the system. A PLL circuit is used to synthesize an 18.432MHz clock for operating the host microprocessor at an internal 9.216MHz rate, which the serial ports utilize to support RS-232 communications.

Component-Level Description:

The 12.288MHz digital output from crystal oscillator Y602 is buffered by IC606-C, which feeds digital multiplexer chip IC609. This in turn routes the 12.288MHz to AES/EBU digital audio transmitter chip IC603 when an internally generated 32kHz or 48kHz output sample rate is selected. The 12.288MHz clock is also sent to an 8-bit synchronous counter implemented in programmable logic array (PLA) IC613. This counter divides down to obtain the lower frequency system clocks. All outputs of the PLA have their transitions coincident with the rising edge of the 12.288MHz clock. The 12.288MHz clock is inverted by buffers IC605-A, -B to provide clocks $\overline{\text{12.288MHZA}}$ and $\overline{\text{12.288MHZB}}$ that have falling edges coincident with the transitions of the lower frequency clocks. $\overline{\text{12.288MHZA}}$ feeds the bit clock of the inter-DSP communication links following buffers IC710-B, -D. $\overline{\text{12.288MHZB}}$ feeds the A/D and D/A master clocks. 17.000MHz crystal oscillator Y601 feeds the master clock (MCLK) inputs of both the input and output SRC chips IC601 and IC603.

The 6.144MHz clock output from IC613 feeds the PLL circuit made up of PLA IC618, 74HC4046 phase detector/VCO IC619 and associated components. The PLA first buffers the 6.144MHz signal, providing a clean 6.144MHz output at pin 12 used as the reference input to the PLL phase detector (IC619 pin 14). Of the three detectors included in the 74HC4046, the phase frequency detector (PFD) is used by the 6200. The output of the phase detector (pin 13) feeds the loop filter made up of resistors R607, R608 and capacitor C605 that provide a single pole low-pass filter forming a second order loop. Pin 9 of IC619 is the input control voltage to the VCO. Resistor R614 eliminates subharmonic frequency modulation of the VCO caused by parasitic capacitance. Resistors R605 and R606 set the PLL's lock-in frequency range. A divide-by-three counter is placed between the VCO output and the phase detector comparator input. This places the VCO output at 18.432MHz. The divide-by-three is implemented by the PLA IC618 between pins 2 and 16. A 4.096MHz clock is provided at pin 17 of the PLA. The PLA provides an 18.432MHz output at pin 14 which feeds Z-180 microprocessor IC100 via buffer IC201-A.

Inverter IC605-C provides the $\overline{\text{6.144MHZ}}$ bit clock for the output D/A and, via buffer IC606-B, the bit clock for the input SRC, IC615.

IC614-A, -D provide buffered clocks 6.144MHZA and 6.144MHZB for driving the EXTAL inputs (pin 27) of the DSP chips. Each buffer drives four DSP chips.

The 192kHz clock output of IC613 (pin 14) is used for the inter-DSP word clock. The 192kHz, 96kHz and 48kHz clocks are all used in the LCD backlight drive circuit. The 48kHz clock also provides, via buffers IC606-A and IC607-C, the word clock interfacing the DSP to the input and output SRC chips and the A/D and D/A converters. The 48kHz clock is used to generate DSP interrupt request signals (IRQBA, IRQBB) required for process timing and interchip synchronization. The circuit consisting of flip-flop IC612 and IC614-B, -C is required to ensure that the first falling edges of all IRQB signals are coincident. This synchronization occurs every time the unit is powered up and when there is a processing algorithm change. It is controlled by the Z-180 via pin 2 of latch IC611. The 48kHz clock is also used, along with IC313, in the A/D clock synchronizing circuit. This circuit makes the IRQB and the L/R clocks, both operating at 48kHz, phase synchronous. This ensures that the process-to-output buffer transfer internal to the DSP doesn't overlap the output buffer-to-peripheral transfer.

AC terminations are used on various clocks throughout the board to improve signal integrity for sensitive devices.

Control Circuits

The control circuits process and execute user-initiated requests to the system. The source of these requests is the front panel buttons, the rear panel RS-232 port, and the remote contact closures (6200 OPTIMOD-DAB only). These changes affect hardware function and/or DSP processing. The control circuits also send information to the LCD display, LED status, and LED meter circuits. A RAM chip stores code segments. For quick access, an EEPROM chip stores dynamic system state information. A ROM chip contains the executable form of 6200 DSP and Control software.

1 Microprocessor and Power Monitoring Circuit

A Z-180 microprocessor executes software code required to control the functionality of the 6200. The EXTAL pin of the Z-180 receives an 18.432MHz clock signal from the clock divider/PLL circuit and is internally divided down to 9.216MHz to provide the Z-180 system clock frequency. ROM contains control software for the Z-180. User system setup and other dynamic system state information that must survive power down is stored in non-volatile EEPROM. Power monitoring circuitry prevents data corruption by placing and holding the Z-180 in reset if AC mains power is insufficient.

The Z-180 communicates to the DSP through the synchronous serial data host port. When the DSP requires executable code, the Z-180 reads it from the ROM and sends it to the DSP. The Z-180 sends parameter control data to the DSP and receives status data from the DSP. If status from DSP is irregular, the Z-180 will place the 6200 hardware and DSP in a reset state and execute initialization procedures.

Component-Level Description:

The Z-180 is IC100. Watchdog timer/voltage monitor IC122 provides the system reset function. IC122 pin 7 monitors pulses generated every 1 second by the Z-180. If the Z-180 is not operating correctly to provide the pulses, IC122 will reset the Z-180. IC122 also monitors the voltage on the +5V source that supplies power to the 6200 digital electronics. When the +5V line is above the minimum operating voltage of +4.75V, R103 will pull $\overline{\text{RESET}}$ high which allows the Z-180 to exit the reset condition. When the +5V line is below the minimum operating voltage, the open-collector output of IC122 pulls Z-180's $\overline{\text{RESET}}$ low which puts the Z-180 into the reset condition, thereby preventing the Z-180 and the 6200 electronics from executing incorrectly due to low +5V line voltage.

Z-180 IC100 pins 55, 56, and 57 comprise the host serial data communication port. The Z-180 uses this port to communicate with the DSP IC700-IC707 via host port interface pins 26, 35, and 41; and with EEPROM IC107 via pins 2, 5, and 6. Communication is SPI type with Z-180 as master and DSP as slave.

2 RAM, ROM and EEPROM

A RAM chip provides temporary storage for Z-180 data and program code segments. A ROM chip provides permanent storage of the executable control software and the executable DSP software. System state information that must be maintained while the 6200 is powered down is stored in an EEPROM. The EEPROM does not lose data when the 6200 is powered down.

Component-Level Description:

IC104 decodes Z-180 memory addresses to access instructions to execute from ROM IC105 and to read or write data from 32KB RAM IC106. EEPROM IC107 is selected by latch IC611 pin 6.

3 Data Latches, Tri-State Data Buffers and Address Decoders

Digital logic decodes Z-180 I/O addresses, allowing the Z-180 to access RAM, ROM and EEPROM. The logic provides Z-180 data bus allocation by using latches and tri-state data buffers to allow other 6200 hardware to communicate to the Z-180. To control other hardware, the Z-180's data bus state is latched at the appropriate time, and the latched control signals are provided to other hardware. For the Z-180 to read information from other hardware, the Z-180's data bus is connected at appropriate times to other hardware's source signals through tri-state data buffers (e.g., IC120).

Component-Level Description:

Decoder IC104 allows the Z-180 to access ROM IC105 and RAM IC106. Decoders IC101, IC102, and IC103 allow the Z-180 to access all other 6200 hardware. The decoded outputs from IC101, IC102, and IC103 are used to latch the state of the Z-180 data bus at appropriate times with data latches IC1, IC2, IC3 IC4, IC5, IC6, IC9, IC303, IC604, IC611, IC708, and IC709, and to allocate the Z-180 data bus at appropriate times to various peripherals via tri-state data buffers IC120 and IC8. IC120 buffers or tri-states status information from the remote contact closure circuitry onto the Z-180 data bus. IC8 buffers or tri-states information from the user control interface onto the Z-180 data bus.

User Control Interface and LED Display Circuits

The user control interface enables the user to control the functionality of the 6200 unit. A rear panel remote interface connector enables remote control of certain functions. Front panel pushbutton switches select between various operational modes and functions. Data latches detect and store the commands entered with these switches. Front panel status LEDs indicate the control status of the unit, and meter LEDs indicate signal levels and processing activity within the unit.

1. Remote Interface (Model 6200 OPTIMOD-DAB only) and RS-232 Interfaces

A remote interface connector and circuitry enables remote control of certain operating modes; Model 6200 OPTIMOD-DAB has eight remote contact closure inputs.

A valid remote signal is a momentary pulse of current flowing through the particular remote signal pins. Current must flow consistently for 50msec for the signal to be interpreted as valid. Generally, the 6200 will respond to the most recent control operation whether it came from the front panel, remote interface, or RS-232.

Component-Level Description:

26-pin header J101 attaches via ribbon cable to the 25-pin D-connector, which connects the remote control input signals. The ribbon cable incorporates a ferrite block to filter out RFI from the signals. The associated opto-isolators (e.g., IC110) isolate the inputs from the detector circuitry on the 6200. The associated diodes (e.g., CR102) prevent the opto-isolators from breaking down under a reverse bias. The outputs of the opto-isolators are inverted and buffered (e.g., by IC118-A) and latched by tri-state data buffer IC120. When $\overline{\text{REMOTE}}$ signal provided to IC120 pin 19 is brought low, IC120 places remote signals on the Z-180 data bus.

The dual RS-232 interface is comprised of 9-pin D-connectors J100 and J104, and ICs 121 and 123. IC121 and IC123 interface the RS-232 signals with the Z-180 microprocessor.

2. Switch Matrix and LED Indicators

Eleven front panel pushbutton switches are arranged in a matrix, configured as three columns and four rows. These switches are the primary element of the physical user interface to the 6200 control software. The host microprocessor controls the system setup and function of the DSP according to the switch/rotary encoder entered commands, the AES Status bits from the Digital Input signal, the RS-232, and the remote control interface status (Model 6200 OPTIMOD-DAB only); and updates the LED control status indicators accordingly.

Component-Level Description:

S1-S11 are the front panel pushbutton switches. CR11-CR15 are the front panel LED control status indicators. Via decoder IC102, the host microprocessor Z-180 periodically selects data latch IC3 (on the display board) to drive one of the three columns in the switch matrix low, then commands tri-state data buffer IC8 (also on the display board) to read its inputs to determine if any new information is being received from one or more of the switches in that column. If no switches are closed, pull-up resistors R25-R28 pull the buffer inputs to +5V. The buffer, in turn, de-bounces the signals and places the appropriate word on the data bus for the Z-180 to read. The Z-180 transmits the updated information to data latch IC3 which directly drives the LED Control Status Indicators.

3. LED Meter Circuits

The meter LEDs are arranged in an 8x16 matrix, in rows and columns.

Each row of LEDs in the matrix has a 1/8 duty cycle ON time. The rows are multiplexed at a fast rate so that the meters appear continuously illuminated. Via the serial port, the DSP sends meter data values to the Z-180, which sends the appropriate LED control words (8 bits at a time) to the data latches that drive the LEDs directly.

Component-Level Description:

The meter LED matrix consists of ten 10-segment LED bargraph assemblies (CR1-CR9, CR16) and one discrete LED (CR10). Row selector latches IC4, IC5, IC6, and IC9 are controlled by the Z-180, and alternately sink current through the LEDs selected by column selector latches IC1 and IC2, which are also controlled by the Z-180. IC1 and IC2 drive the selected row of LEDs through current limiting resistor packs RP1 and RP2.

Input Circuits

This circuitry interfaces the analog and digital audio 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 transmits them to the input sample rate converter (SRC). The digital audio from the A/D and SRC is transmitted to the DSP.

1. Analog Input Stages

The RF-filtered left and right analog input signals are each applied to a resistor load and a resistor pad. The load is enabled or disabled by a jumper that is positioned by hand. The loaded and padded signal is applied to a floating-balanced amplifier that has an adjustable (digitally-controlled) gain. The gain is set by FET transistors and analog switches. The state of the FETs and switches is set by the outputs of a latch. The control circuits control the gain according to what the user specifies from the front panel controls by writing data to the latch. The gain amplifier output feeds a circuit that scales, balances, and DC-biases the signal. This circuit feeds an RC low-pass filter which applies the balanced signal to the analog-to-digital (A/D) converter.

Component-Level Description:

The left channel balanced audio input signal is applied to the filter/load/pad network made up of L300, L301, L302, L303, R300-R305, R316, R317, C323 and C326. J301 is a jumper that removes or inserts the optional 600Ω termination load (R300) on the input signal. CR200-CR203 are protection diodes applied to the padded input signal before it is applied to IC300, a differential amplifier. R306, R307, R310-R313, FETs Q300-Q301, and quad analog switch IC307 make up the circuit that sets the gain of IC300. The FETs, along with IC307, are used as switches to change the resistive paths in the circuit. The state of the FET switches is set by the outputs of digital latch IC303. The latch outputs feed IC306, a quad comparator, which outputs 0V to turn on a FET and −15V to turn off a FET. The control circuit writes directly to IC307 to control the state of the switches on

IC307. IC300 feeds IC304 and associated components. This stage balances, DC-biases, and scales the signal to the proper level for the analog-to-digital (A/D) converter. IC301-B and associated components comprise a servo amp to correctly DC-bias the signal feeding the A/D converter. R352, R353, R357, and C332 make an attenuator/RC filter necessary to filter high frequency energy that would otherwise cause aliasing distortion in the A/D converter. The corresponding right channel circuitry is functionally identical to that just described.

2. Stereo Analog-to-Digital (A/D) Converter

The A/D is a stereo 24-bit sigma-delta converter, implemented on a dual-chip integrated circuit. The A/D oversamples the audio at 6.144MHz. It applies noise shaping, then it filters and decimates to a 48kHz sample rate. The samples are output in two's complement, 32-bit word, two-word frame serial format, MS bit first, and transmitted to the DSP. The A/D is configured in “Slave Mode” — all of its clocks originate from the 6200 system clocking, to interface to the input of the DSP. For more information on 6200 input clocking, please refer to “12.288MHz Oscillator and System Clocking.”

Component-Level Description:

The balanced left analog input is applied to pins 4(+) and 5(–), and the balanced right analog input is applied to pins 25(+) and 24(–) of the A/D (IC309). The maximum differential signal that the A/D can accept is typically ±2.45Vpeak. The A/D samples the left and right inputs simultaneously at 128 times the 6200 sample rate of 48kHz. MCLK, the master clock input of the A/D (pin 17), is fed a 12.288MHz clock providing the 6.144MHz input sample rate required. The A/D sends the digitized stereo audio to the first DSP chip (IC700) via its synchronous serial port formed by the data SDATA (pin 15), the bit clock SCLK (pin 14) and the word clock LRCK (pin 13).

3. Digital Input Receiver and Sample Rate Converter (SRC)

The digital input receiver accepts digital audio signals using the AES/EBU interface format (AES3-1992). The receiver and input sample rate converter (SRC) together will accept and sample-rate convert any of the “standard” 32kHz, 44.1kHz, 48kHz rates in addition to any digital audio sample rate within the range of 25kHz and 55kHz. The audio signal received is decoded by the AES receiver and sent to the SRC. The SRC converts the input sample rate to the 48kHz 6200 system sample rate. Via a synchronous serial interface, the SRC sends the 48kHz sample rate audio to the DSP for processing.

Component-Level Description:

The differential digital input signal is received through a shielded 1:1 pulse transformer (T600). T600 has very low inter-winding capacitance, providing a high level of isolation for high frequency common mode interference. IC600 is a dedicated AES/EBU digital audio receiver integrated circuit. It contains a phase locked loop that recovers the clock and the synchronization information present in the AES/EBU signal. A Schmitt trigger at the input provides 50mV of hysteresis for added noise immunity. R604 provides a 110Ω input impedance per the AES/EBU specification.

AES receiver chip IC600 communicates with the Z-180 via control registers and data memory accessed through the parallel port made up of the 4-bit address bus (pins 15-18), the 8-bit data bus (pins 1-6, 27-28) and the $\overline{\text{CS}}$ and RD/$\overline{\text{WR}}$ control pins (pins 23 and 24).

The data consists of input sample rate, signal validity/error information, and user and status bits from the AES stream. IC600 pin 14 interrupts the Z-180 when user data is to be read from its internal registers.

Received AES audio is transmitted from the AES receiver to the input sample rate converter (SRC IC601). The AES receiver is master and the SRC is slave. The AES receiver outputs data on pin 26, the bit clock on pin 12, and the frame clock on pin 11. These signals are sent to the SRC serial input interface pins 3, 4, and 6 respectively.

The MCK clock output at pin 19 of the AES receiver chip has a frequency 256 times the input sample rate of the received signal. This is used to drive the output AES/EBU transmitter when an output sample rate that is synchronous to the input or sync input sample rate (external sync) is required.

Crystal oscillator Y601 provides the input SRC a master clock of 17.000 MHz on pin 2. This MCLK frequency allows the input SRC to operate with input sample rates in the range of 8.5kHz (MCLK/2000) to 59kHz (MCLK/286). SRC_RST is an active low reset signal tied to pin 13 of the SRC. This signal is controlled by the Z-180 via pin 2 of latch IC604.

The MSBDLY_I, BKPOL_I, and TRGLR_I pins of the SRC chip configure the chip to interface with the AES/EBU receiver chip. Pin 1 of the SRC (GPDLYS) is tied high to minimize the chip's group delay to approximately 700µs as opposed to approximately 3ms, giving up some tolerance to variations in sample rates. Pin 28 (SETLSLW) is tied high to cause the SRC to settle slowly to changes in sample rates, resulting in the best rejection of sample rate jitter.

The sample rate converted output of the input SRC feeds the first DSP chip (IC700). The SRC output port and the DSP input port are both slaves, with clocks supplied by the 6200 system clocking. The SRC generates DIG_IN (data) on pin 23, and receives the bit clock and the word clock on pins 26 and 24 respectively.

4. Digital Sync Input Receiver

The digital sync input receiver accepts digital signals using the AES/EBU interface format (AES3-1992). This receiver will accept any of the “standard” 32kHz, 44.1kHz, 48kHz rates. The signal received is decoded by the AES receiver and the frame, bit, and master clocks are fed to selector IC615 for use in syncing the 6200’s AES/EBU digital output signal when external sync is selected. (The selected output sample rate and the incoming sync input rate must match for this to happen.)

Component-Level Description:

The differential digital input signal is received through a shielded 1:1 pulse transformer (T602). T602 has very low inter-winding capacitance, providing a high level of isolation for high frequency common mode interference. IC610 is a dedicated AES/EBU digital audio receiver integrated circuit. It contains a phase locked loop that recovers the clock and the synchronization information present in the AES/EBU signal. A Schmitt trigger at the input provides 50mV of hysteresis for added noise immunity. R616 provides a 110Ω input impedance per the AES/EBU specification.

AES receiver chip IC610 communicates with the Z-180 via control registers and data memory accessed through the parallel port made up of the 4-bit address bus (pins 15-18), the 8-bit data bus (pins 1-6, 27-28) and the $\overline{\text{CS}}$ and RD/$\overline{\text{WR}}$ control pins (pins 23 and 24).

The data consists of input sample rate and signal validity/error information from the AES stream.

The AES receiver outputs the master clock on pin 19, the bit clock on pin 12, and the frame clock on pin 11. The MCK clock output at pin 19 of the AES receiver chip has a frequency 256 times the input sample rate of the received signal. These clocks are used to drive the output AES/EBU transmitter when an output sample rate that is synchronous to the sync input sample rate (external sync) is required.

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 output audio. 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

The D/A is a stereo, 20-bit delta-sigma converter. It receives the serial left and right audio data samples from the DSP circuits.

For information on 6200 system clocking, please refer to “12.288MHz Oscillator and System Clocking.”

Component-Level Description:

IC400 is the digital-to-analog (D/A) converter for the left and right analog monitor output signals. The synchronous serial input interface consists of the serial data input (SDATA), serial data clock (SCLK), and the left/right clock (LRCK). This interface is configured via DIF∅, DIF1, and DIF2 pins for 20-bit left justified audio data with MSB aligned with the leading edge of LRCK. Data is latched on the falling edge of SCLK. The processed digital output (ANLG_OUT) is provided by DSP IC707 on its SAI output port SDO2 (pin 45), and is received by the D/A on pin 10.

A 6.144MHz bit clock is provided from the system clock circuitry to both the final DSP and the D/A chips. The DSP output data format is 32 bits per word, two words per frame. DSP chip IC707 receives a 48kHz frame clock at its WST input (pin 50) that sets the word transfer rate to two words per 48kHz period. The D/A receives a 48kHz clock at its LRCK input (pin 7). LRCK delineates the left and right samples used by the D/A; therefore the D/A uses the first sample received for the left output and the second sample for the right output. The DSP output samples are formatted to ensure that the D/A uses samples that represent the simultaneously sampled analog input.

2. Analog Output Stages

The left and right analog signals emerging from the digital-to-analog (D/A) converter are each filtered, amplified, and applied to a floating-balanced line driver, having a 30Ω, ±5% output impedance. The line driver outputs are applied to the RF-filtered left and right analog output connectors. These analog signals provide a convenient means of monitoring the processed audio.

Component-Level Description:

The left channel signal emerging from the digital-to-analog (D/A) converter is filtered by IC404-A, IC404-B, IC406-A, and associated components. The purposes of these stages are to remove common-mode errors including noise, distortion, and DC offset, and to reduce the out-of-band noise energy resulting from the delta-sigma D/A’s noise-shaping filter.

IC404-A, IC404-B, and associated components implement a 3rd order Chebychev low-pass filter characteristic to the differential signal from the D/A. The 20kHz passband nominally has 0.05dB ripple and about 7 degrees of deviation from linear-phase. This low-sensitivity filter, utilizing tight tolerance components, does not induce significant overshoot of the processed audio, which would otherwise waste modulation.

IC406-A and associated components comprise a low-frequency servo amplifier to remove residual DC from the signal. The 0.15Hz –3dB frequency averts tilt-induced overshoot of the processed audio.

IC404-B feeds the stage consisting of IC403-A, IC403-B, IC402-A, and associated components, which is a floating-balanced line driver. The floating characteristic is achieved by complex cross-coupled positive and negative feedback between two 5532 opamps, and its operation is not readily explainable except by a detailed mathematical analysis. Opamps may be replaced; resistors are specially matched and should not be replaced. IC402-A, R408, R409, R412, and C410 comprise a servo amplifier, which centers around ground the average DC level at output connector J400.

The balanced audio output signal is applied to the RF filter network made up of L412, L413, L402, and L403, and then to XLR connector J400.

The corresponding right channel circuitry is functionally identical to that just described.

3. Digital Sample Rate Converter (SRC) and Output Transmitter

An output sample rate converter (SRC) chip is used to convert the 48kHz 6200 system sample rate to any of the standard 32kHz, 44.1kHz or 48kHz rates. A digital audio interface transmitter chip is used to encode digital audio signals using the AES/EBU interface format (AES3-1992). A synchronous serial interface is used for all inter-chip communication between the DSP, SRC, and AES transmitter chips.

Component-Level Description:

The processed digital output (DIG_OUT) provided at the SAI output port SDO0 (pin 47) of DSP IC707 is received by asynchronous sample rate converter (SRC) IC602 pin 3. A 6.144MHz bit clock is provided from the system clock circuitry to both the final DSP and the SRC chips. DSP chip IC707 receives a 48kHz frame clock at its WST input (pin 50) that sets the word transfer rate to two words per 48kHz period. The SRC receives a 48kHz clock at its $L/\overline{R}$_I input (pin 6). $L/\overline{R}$_I delineates the samples of the two channels used by the SRC. The DSP output samples are formatted to ensure that the SRC uses samples that represent the simultaneously sampled analog input.

Crystal oscillator Y601 provides the output SRC a master clock of 17.000MHz on pin 2. This MCLK frequency allows the output SRC to operate with an output sample rate in the range between 30kHz and 59kHz. OSRC_RST is an active low reset signal tied to pin 13 of the SRC. This signal comes from multiplexer chip IC609 and is controlled by the Z-180 via either pin 2 of latch IC604 or pin 8 of IC605-D.

The MSBDLY_I, BKPOL_I, and TRGLR_I pins of the SRC chip configure the chip to interface with the last DSP chip (IC707). Pin 1 of the SRC (GPDLYS) is tied high to minimize the chip's group delay to approximately 700µs as opposed to approximately 3ms, giving up some tolerance to variations in sample rates. Pin 28 (SETLSLW) is tied high to cause the SRC to settle slowly to changes in sample rates, resulting in the best rejection of sample rate jitter.

The output side of the sample rate converter is tied directly to IC603, an AES/EBU digital audio transmitter integrated circuit. This interface uses the AES transmitter chip as master unless external sync has been selected and a valid sync signal is present at the input or sync input AES receiver chips, in which case the chip with the valid sync signal becomes master (see external sync available clocks). Two free running clocks provide the standard sample rates of 32kHz, 44.1kHz and 48kHz when an internal sync is requested. These clocks run at a frequency that is 384 or 256 times the sample rate they represent. They have a frequency stability of ±100PPM. The third clock is the clock that is recovered from the appropriate AES/EBU receiver chip. This clock has a frequency of 256 times the input sample rate of the received signal. This is used to drive the output

AES/EBU transmitter when an output sample rate is required that is synchronous to the input or sync input sample rate (external sync).

The inter-chip serial data format, the input MCK multiplication factor, and the user and output channel status data are controlled by the Z-180 via internal control registers and data memory accessed through the parallel port made up of the 5-bit address bus (pins 9-13), the 8-bit data bus (pins 1-4, 21-24), and the $\overline{\text{CS}}$ and RD/$\overline{\text{WR}}$ control pins (pins 14 and 16) of IC603. IC603 pin 15 interrupts the Z-180 when its internal registers can be written with user data.

The on-chip RS422 line driver provided by IC603 is a low skew, low impedance, differential output capable of driving a 110Ω transmission line with a 4Vp-p signal. Shielded 1:1 pulse transformer T601 transmits the differential digital output signal to XLR connector J601.

DSP Circuits

The DSP circuits consist of eight general-purpose 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 37 million instructions per second (MIPS) for a total of 295MIPS, provide the necessary signal processing. A 48kHz sampling rate is used. Two of the on-board serial audio interface (SAI) peripherals on each DSP chip are used to transfer data chip-to-chip at a 24.576Mbit/s rate, maintaining a 24-bit word length. The DSP chips are cascaded, processing the audio serially. The first chip receives the analog input via the A/D chip and the digital input via the SRC chip. Input source selection is performed seamlessly, internal to the DSP chip.

During system initialization (which normally occurs when power is first applied to the 6200), and when processing algorithms are changed, the Z-180 downloads the DSP executable code stored in the ROM, via the serial host interface (SHI) port of each DSP chip. Once a DSP chip begins executing its program, execution is continuous. The Z-180 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 via this same SHI port.

The “analog” and digital outputs are sent to the D/A and the output SRC chips via the SAI port of the last DSP chip, IC707.

Component-Level Description:

IC700 through IC707 are the DSP chips.

CAUTION: Do not attempt to remove these chips from the PCB. These chips should be removed only by the Orban service department. A chip can be ruined by static discharge or by damage to its delicate pins.

The EXTAL pin of each DSP chip receives a 6.144MHz clock. All DSP chips use their internal PLL to multiply this by 12 to operate the chip's internal oscillator (Fosc) at 73.728MHz. Each DSP chip is reset by the Z-180 via latch IC709. DSP mode configuration is controlled by the state of the MODA, MODB and MODC (pins 37, 38, 39) on each chip as the chip is brought out of reset. All DSP chips are configured to bootstrap via the SHI port. The MODB pin, which also serves as the IRQB input after leaving the reset state, is forced low prior to bringing the DSP chips out of reset.

Pins 26, 35, 41 and 42 comprise the DSP host port. Host port communication conforms to the SPI format with the Z-180 set up as the master and the DSPs as slaves. The Z-180 generates the HOSTCK clock signal and provides it to SCK (pin 26) of each DSP. The Z-180 provides the data on the HOSTTX line tied to pin 41 of each DSP. The data output (pins 35) of each DSP have tri-state outputs that are wire-ORed to provide the data on the HOSTRX line sent to the Z-180. The Z-180 controls the slave select ($\overline{\text{SS}}$) (pin 42) of each DSP via latch IC708. The $\overline{\text{SS}}$ pin is used to enable each of the slaved DSP SPI ports for transfer.

DSP IC700 pins 56 and 57 receive serial audio from the digital and analog inputs. These are the two input ports of the synchronous serial audio interface (SAI) receiver internal to the DSP. The two serial audio streams are received simultaneously. Both inputs share the same frame clock, $\overline{\text{L}}$/R (48kHz) provided to DSP IC700 pin 55 and the same bit clock, SCK (6.144MHz) provided to DSP IC700 pin 51.

Communication between DSP chips IC700 (first) through IC707 (last) is one-way, in series from the first to the last. Two of the on-board SAI peripherals on each DSP are used to transfer 8 words each per frame chip-to-chip. The I2S communication protocol (two 32-bit words per cycle of the word clock) is used with the DSPs as slaves, and the 6200 system clocking as master. Data is sent from the two transmit data port pins 46 and 47 of one chip to the next chip's receive data port pins 56 and 57. A 192kHz word clock is provided to the transmit pin 50 and the receive pin 55. A 12.288MHz bit clock is provided to the transmit pin 49 and the receive pin 51. The SAI links between DSPs are synchronized to each other (to align the SAI time slots) by making the first occurrence of all IRQBs coincident, (controlled by Z180 and external hardware) and having all DSPs initialize their SAI ports on the first reception of IRQB.

The “analog” and digital outputs are transferred respectively to the D/A and the output SRC from the last DSP chip (IC707). (“Analog” refers to DSP signal that ultimately gets converted to analog.)

Power Supply

The power supply converts an AC line voltage input to various power sources used by the 6200. To ensure lowest possible noise, five linear regulators provide ±15VDC and

±5VDC for the analog circuits and +5VDC for the digital circuits. An unregulated voltage powers the LED backlight on the LCD display.

Component-Level Description:

L1 is a power line filter that filters out RFI. F1 is a 1/2-amp “Slo-Blo” fuse. T1 is a dual-primary dual-secondary power transformer used to step down the input voltage for the ±15VDC analog and +5VDC digital supply regulators. Each primary winding has a Metal-Oxide Varistor (V1, V2) connected in parallel to suppress high-voltage spikes across the AC line. Rear panel switch S1 configures the primary windings either in parallel (for 115V ±15% line voltages) or series (for 230V ±15% line voltages).

T1 has three secondary windings for stepping down the AC line voltage. The lower voltage winding feeds storage capacitors C15 and C19 through full-wave bridged rectifier diodes CR13, CR14, CR15, CR17 and CR18. C15 filters the rectified voltage for input to low-dropout linear voltage regulator IC5, which provides the +5VDC source used to power all of the digital circuits in the 6200. C19 filters the rectified voltage to power the LED backlight on the LCD display. Components Q1, Q2, R3-R7, and CR20 form a pulsed current source to illuminate the 25x2 LED array (the backlight on the LCD display). The signal LEDPULSE, a 48kHz pulse at 1/8 duty cycle, feeds the base of high-current Darlington transistor Q1. The feedback circuit consisting of Q2, CR20 and R3-R7 controls the magnitude of the signal LEDPULSE so as to limit Q1's current pulses to about 1.5A (1/8 duty cycle). These current pulses illuminate the 25 x 2 LED array via keyed header J201, which attaches the LED array between the collector of Q1 and supply cap C19. The signal LEDPULSE is gated on for approximately one hour after the 6200 has last been powered up or a front panel button has last been pressed; otherwise, it is gated off. This drastically increases the lifetime of the LCD display and saves power. The LED meter circuits are described in “User Control Interface and LED Display Circuits.”

The higher voltage pair of transformer secondary windings is configured in series to form a single center-tapped winding. This winding is connected to rectifier diodes CR1-CR4 in a full-wave center tap configuration. C1 and C2 filter the rectified voltage for input to the voltage regulators IC1 and IC2. These regulators provide the +15VDC and −15VDC sources used to power most of the analog circuits in the 6200. They also serve as the respective inputs to the voltage regulators IC3 and IC4. IC3 provides the +5VDC analog supply for the converter chips, which draw only a modest amount of current through power-dissipating resistors R8-R15.

Test points and supply bypass capacitors are placed throughout the PC board. S2 is the ground lift switch used to connect or lift 6200 circuit ground from chassis ground.

Parts List

Parts are listed by ASSEMBLY, then by TYPE, then by REFERENCE DESIGNATOR. Widely used common parts are not listed; such parts are described generally below (examine the part to determine exact value). See the following assembly drawings for locations of components.

SIGNAL DIODES, if not listed by reference designator in the following parts list, are:

> Orban part number 22101-000, Fairchild (FSC) part number 1N4148, also available from many other vendors. This is a silicon, small-signal diode with ultra-fast recovery and high conductance. It may be replaced with 1N914 (BAY-61 in Europe).
>
> (BV: 75V min. @ Ir = 5µA; Ir: 25nA max. @ Vr = 20V; Vf: 1.0V max. @ If = 100mA; trr: 4ns max.) See Miscellaneous list for ZENER DIODES (reference designator VRxx).

RESISTORS should only be replaced with the same style and with the exact value marked on the resistor body. If the value marking is not legible, consult the schematic or the factory. Performance and stability will be compromised if you do not use exact replacements.

> Unless listed by reference designator in the following parts list, you can verify resistors by their physical appearance:
>
> Metal film resistors have conformally-coated bodies, and are identified by five color bands or a printed value. They are rated at 1/8 watt @ 70°C, ±1%, with a temperature coefficient of 100 PPM/°C. Orban part numbers 20038-xxx through 20045-xxx, USA Military Specification MIL-R-10509 Style RN55D. Manufactured by R-Ohm (CRB-1/4FX), TRW/IRC, Dale, Corning, and Matsushita.
>
> Carbon film resistors have conformally-coated bodies, and are identified by four color bands. They are rated at 1/4 watt @ 70°, ±5%. Orban part numbers 20001-xxx, Manufactured by R-Ohm (R-25), Dale, Phillips, Spectrol, and Matsushita.
>
> Carbon composition resistors have molded phenolic bodies, and are identified by four color bands. The 0.090 x 0.250 inch (2.3 x 6.4 mm) size is rated at 1/4 watt, and the 0.140 x 0.375 inch (3.6 x 9.5 mm) size is rated at 1/2 watt, both ±5% Orban part numbers 2001x-xxx, USA Military Specification MIL-R-11 Style RC-07 (1/4 watt) or RC-20 (1/2 watt). Manufactured by Allen-Bradley, TRW/IRC, and Matsushita.
>
> Cermet trimmer resistors have 3/8-inch (9 mm) square bodies, and are identified by printing on their sides. They are rated at 1/2 watt @ 70°C, = ±10%, with a temperature coefficient of 100 PPM/°C. Orban part numbers 20510-xxx and 20511-xxx. Manufactured by Beckman (72P, 68W- series), Spectrol, and Matsushita.

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 local offices. Addresses for each manufacturer's USA headquarters are given on page 6-29.

Ref Des	Description	Orban P/N	Ven	Vendor P/N	Alternate Vendors	Notes
MAIN BOARD						
	Capacitors					
C11-43	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C45-51	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C53	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C54	Ceramic, Axial, 100V, 5%; 1000pF	21127-210	KEM	C410C102J1G5CA		
C55-60	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C67-71	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C74-78	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C82-84	Alum., Radial, 25V, 10%; 10uF	21263-610	NIC	UKLIE101KPAANA		
C85-90	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C91-92	Alum., Radial, 25V, 10%; 10uF	21263-610	NIC	UKLIE101KPAANA		
C100	Ceramic, Axial, 100V, 5%; 1000pF	21127-210	KEM	C410C102J1G5CA		
C102	Alum, Radial, 63V, -20% +100%; 2.2Uf	21209-522	SPR	502D 225G063BB1C	PAN	
C103	Ceramic Disc, 100V, 5%; 33pF	21127-033	KEM	C410C330JIG5CA		
C104-5	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C200	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C201	Met. Polyester, 50V, 5%; 0.1uF	21445-410	PAN	ECQ-V1H104JZ		
C300	Met. Polyester, 50V, 5%; 0.1uF	21445-410	PAN	ECQ-V1H104JZ		
C302-3	Met. Polyester, 50V, 5%; .0047uF	21445-247	PAN	ECQ-B1H472 F1		
C306-7	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C308	Tantalum, 10V, 10%; 100uF	21303-710	SPR	196D 107X9010PE4	MANY	
C309-10	Met. Polyester, 50V, 5%; 0.22uF	21445-422	PAN	ECQ-V1H224JZ		
C311	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C312	Tantalum, 35V, 10%; 1uF	21307-510	SPR	196D 105X9035HA1	MANY	
C313-14	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C315	Tantalum, 35V, 10%; 1uF	21307-510	SPR	196D 105X9035HA1	MANY	
C316-19	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C320	Tantalum, 20V, 10%; 10uF	21305-610	SPR	196D 106X9020JA1	MANY	
C321-22	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C323-24	CAP,0.0082uF,1KV,10%,CER DISC	21112.282.01				
C325-26	CAP,0.0030uF,1KV,10%,CER DISC	21112.230.01				
C400	Tantalum, 35V, 10%; 1uF	21307-510	SPR	196D 105X9035HA1	MANY	
C401	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C402	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C403	Tantalum, 35V, 10%; 1uF	21307-510	SPR	196D 105X9035HA1	MANY	
C404-5	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C406	Tantalum, 20V, 10%; 10uF	21305-610	SPR	196D 106X9020JA1	MANY	
C407-8	Met. Polyester, 50V, 5%; 0.01uF	21445-310	PAN	ECQ-V1H103JZ		
C411	Ceramic, Axial, 100V, 5%; 1000pF	21127-210	KEM	C410C102J1G5CA		
C412	Ceramic Disc, 1kV, 10%; 0.0015uF	21112-215	CEN	DD-152 MUR		
C413	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C414	Tantalum, 35V, 10%; 1uF	21307-510	SPR	196D 105X9035HA1	MANY	
C415	Ceramic Disc, 100V, 5%; 33pF	21127-033	KEM	C410C330JIG5CA		
C416	Ceramic, Axial, 100V, 5%; 1000pF	21127-210	KEM	C410C102J1G5CA		
C417-18	Ceramic Disc, 1kV, 10%; 0.0015uF	21112-215	CEN	DD-152 MUR		
C419	Ceramic, Axial, 100V, 5%; 1000pF	21127-210	KEM	C410C102J1G5CA		
C420	Ceramic Disc, 1kV, 10%; 0.0015uF	21112-215	CEN	DD-152 MUR		
C421	Ceramic, Axial, 100V, 5%; 1000pF	21127-210	KEM	C410C102J1G5CA		
C422	Ceramic Disc, 100V, 5%; 33pF	21127-033	KEM	C410C330JIG5CA		
C423	Met. Polyester, 50V, 5%; 0.1uF	21445-410	PAN	ECQ-V1H104JZ		
C424-25	Ceramic Disc, 100V, 5%; 33pF	21127-033	KEM	C410C330JIG5CA		
C426	Met. Polyester, 50V, 5%; 0.1uF	21445-410	PAN	ECQ-V1H104JZ		
C604	Ceramic Disc, 100V, 5%; 150pF	21127-115	KEM	C410C151JIG5CA		
C605	Met. Polyester, 50V, 5%; 1.0uF	21445-510	PAN	ECQ-V1H105JZ		
C606	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C607	Alum., Radial, 25V, 10%; 100uF	21263-710	NIC	UKLIE101KPAANA		
C800	CAP,M/P,50V,5%,.018uF	21445.318.01				
C801	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C802	CAP,M/P,50V,5%,.018uF	21445.318.01				
C803	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C804	CAP,M/P,50V,5%,.018uF	21445.318.01				
C805	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		

Ref Des	Description	Orban P/N	Ven	Vendor P/N	Alternate Vendors	Notes
C806	CAP,M/P,50V,5%,.018uF	21445.318.01				
Capacitors (continued)						
C807	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C808	CAP,M/P,50V,5%,.018uF	21445.318.01				
C809	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C810	CAP,M/P,50V,5%,.018uF	21445.318.01				
C811	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C812	CAP,M/P,50V,5%,.018uF	21445.318.01				
C813	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C814	CAP,M/P,50V,5%,.018uF	21445.318.01				
C815	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
Diodes						
CR100-1	DIO,ZNR,1W,5%,12V	22004.120.01				
CR102-9	Diode, Rectifier, 400V, 1A	22201-400	MOT	1N4004	MANY	
CR200-3	Diode, Zener, 1W; 9.1V	22003-091	MOT	1N4739	MANY	
CR300-3	Diode, Signal, Hot Carrier	22102-001	HP	HP5082-2800	MANY	
Inductors						
L100	Inductor, RF Choke; 7uH	29501-004	OHM	Z-50		
L300	Filter, EMI, W/BEAD, 50V,1000PF	29508-210	TAI	STB102KB		
L301	IND,8.2MH,73F823AF (MILLER)	29503.822.01				
L302	Filter, EMI, W/BEAD, 50V,1000PF	29508-210	TAI	STB102KB		
L303	IND,8.2MH,73F823AF (MILLER)	29503.822.01				
L310	INDUCTOR 2A 2.2UH	240-003				
L400-1	Inductor, RF Choke; 1.2mH	29503-000	MIL	73F123AF		
L402	Filter, EMI, W/BEAD, 50V,1000PF	29508-210	TAI	STB102KB		
L403	Filter, EMI, W/BEAD, 50V,1000PF	29508-210	TAI	STB102KB		
L404-5	Inductor, RF Choke; 1.2mH	29503-000	MIL	73F123AF		
L406	Filter, EMI, W/BEAD, 50V,1000PF	29508-210	TAI	STB102KB		
L407	Filter, EMI, W/BEAD, 50V,1000PF	29508-210	TAI	STB102KB		
Integrated Circuits						
IC100	Digital, Microprocessor	24822-000	ZI	Z8018010VSC		
IC101-4	Address Decoder	24899-000	MOT	MC74AC138D		
IC105	ASSY,EPROM,6200 MAIN BD	44063.000				
IC106	Digital, SRAM	24817-000	TOS	TC55257CPL-10		
IC107	EEPROM,2Kx8BIT,5V,8LEAD	24904.000.01				
IC109	Digital, Inverter	24900-000	TI	SN74HC14AD		
IC110-17	Optoisolator, NPN	25003-000	SIE	SFH-601-1		
IC118-19	Digital, Inverter	24900-000	TI	SN74HC14AD		
IC120	Digital, Transceiver	24851-000	SIG	74HC245D		
IC121	Digital, Quad Line Driver	24661-302	NAT	DS14C88N		
IC122	Power Monitor/Watchdog	24872-000	MAX	1232CPA		
IC123	Digital, Quad Line Receiver	24662-302	NAT	DS14C89A		
IC201	Digital, AND Gate	24850-000	MOT	MC74HC08AD		
IC300	Audio Preamp	24727-402	AD	SSM-2017P		
IC301	Linear, Dual Opamp	24209-202	NAT	LF412CN		
IC302	Linear, Dual Opamp	24207-202	SIG	NE5532N TI, EXR		
IC303	Digital, Latch	24857-000	MOT	MC74HC374ADW		
IC306	Quad Comparator	24710-302	NAT	LM339		
IC307	Quad SPST Switches	24728-302	AD	ADG222		
IC312	20-BIT A/D,DIP/28	24933.000.01	CSC	CS5390KP		
IC312	Digital, A/D Converter	24643-000	CSC	CS5389KP-EP		
IC313	Digital, Flip-Flop	24858-000	TI	SN74HC74D		
IC314	Digital, AND Gate	24850-000	MOT	MC74HC08AD		
IC400	Digital, Stereo D/A Converter	24821-000	CSC	CS4328KP		
IC402	Linear, Dual Opamp	24209-202	NAT	LF412CN		
IC403	Linear, Dual Opamp	24207-202	SIG	NE5532N TI,EXR		
IC406	Linear, Dual Opamp	24207-202	SIG	NE5532N TI,EXR		
IC604	Digital, Flip-Flop	24858-000	TI	SN74HC74D		
IC605	Digital, Inverter	24900-000	TI	SN74HC14AD		
IC606	Digital, AND Gate	24850-000	MOT	MC74HC08AD		
IC607	Digital, AND Gate	24850-000	MOT	MC74HC08AD		
IC611	Digital, Latch	24857-000	MOT	MC74HC374ADW		

Ref Des	Description	Orban P/N	Ven	Vendor P/N	Alternate Vendors	Notes
IC612	Digital, Flip-Flop	24858-000	TI	SN74HC74D		
IC613	PAL	44032-100	ORB			
	Integrated Circuits (continued)					
IC614	Digital, AND Gate	24850-000	MOT	MC74HC08AD		
IC618	PAL	44031-100	ORB			
IC619	Digital, PLL	24901-000	SIG	74HC4046AD		
IC700-7	Digital, DSP	24897-000	MOT	DSP56004FJ50		
IC708-9	Digital, Latch	24857-000	MOT	MC74HC374ADW		
IC710	Digital, AND Gate	24850-000	MOT	MC74HC08AD		
	Resistors					
R100	Resistor, 1/4W; 0 OHM (Jumper)	20020-025	ROH	JPW-02A		
R101-5	RES,10K,1/8W,1%,TF,SMD1206	20124.100.01				
R106	Resistor Network, SIP; 100K	20221-101	BEK	L10-1C104		
R107	RES,10K,1/8W,1%,TF,SMD1206	20124.100.01				
R124	Resistor Network, SIP; 100K	20221-101	BEK	L10-1C104		
R126	RES,100K,1/8W,1%,TF,SMD1206	20125.100.01				
R127	RES,MF,1/2W,1%,301Ω	20080.301.01				
R128	RES,10K,1/8W,1%,TF,SMD1206	20124.100.01				
R129	RES,110Ω,1/8W,1%,TF,SMD1206	20122.110.01				
R131-32	Resistor, 1/4W; 0 OHM (Jumper)	20020-025	ROH	JPW-02A		
R216	RES,2K,1/8W,1%,TF,SMD1206	20123.200.01				
R301-2	RES,4.99K,1/8W,1%,TF,SMD1206	20123.499.01				
R303	RES,845Ω,1/8W,1%,TF,SMD1206	20122.845.01				
R304-5	RES,100K,1/8W,1%,TF,SMD1206	20125.100.01				
R306	RES,249Ω,1/8W,1%,TF,SMD1206	20122.249.01				
R307	RES,511Ω,1/8W,1%,TF,SMD1206	20122.511.01				
R308	RES,47.5K,1/8W,1%,TF,SMD1206	20124.475.01				
R309	RES,47.5K,1/8W,1%,TF,SMD1206	20124.475.01				
R310	RES,1.05K,1/8W,1%,TF,SMD1206	20123.105.01				
R311	RES,1.82K,1/8W,1%,TF,SMD1206	20123.182.01				
R312	RES,3.92K,1/8W,1%,TF,SMD1206	20123.392.01				
R313	RES,6.34K,1/8W,1%,TF,SMD1206	20123.634.01				
R314-15	RES,10K,1/8W,1%,TF,SMD1206	20124.100.01				
R330	RES,13.3K,1/8W,1%,TF,SMD1206	20124.133.01				
R331	RES,6.65K,1/8W,1%,TF,SMD1206	20123.665.01				
R332	RES,69.8K,1/8W,1%,TF,SMD1206	20124.698.01				
R333	RES,1.00M,1/8W,1%,TF,SMD1206	20126.100.01				
R334	RES,39.2Ω,1/8W,1%,TF,SMD1206	20121.392.01				
R335	RES,4.99K,1/8W,1%,TF,SMD1206	20123.499.01				
R336	RES,4.99K,1/8W,1%,TF,SMD1206	20123.499.01				
R337	RES,39.2Ω,1/8W,1%,TF,SMD1206	20121.392.01				
R338	RES,2.49K,1/8W,1%,TF,SMD1206	20123.249.01				
R348	RES,51.1Ω,1/8W,1%,TF,SMD1206	20121.511.01				
R349	Resistor, 1/4W; 0 OHM (Jumper)	20020-025	ROH	JPW-02A		
R402-3	RES,51.1Ω,1/8W,1%,TF,SMD1206	20121.511.01				
R407	RES,845Ω,1/8W,1%,TF,SMD1206	20122.845.01				
R408	RES,20.0K,1/8W,1%,TF,SMD1206	20124.200.01				
R415-16	RES,10K,1/8W,1%,TF,SMD1206	20124.100.01				
R417	RES,845Ω,1/8W,1%,TF,SMD1206	20122.845.01				
R418	RES,20.0K,1/8W,1%,TF,SMD1206	20124.200.01				
R424	RES,324K,1/8W,1%,TF,SMD1206	20125.324.01				
R425	RES,20.0K,1/8W,1%,TF,SMD1206	20124.200.01				
R428-29	RES,10K,1/8W,1%,TF,SMD1206	20124.100.01				
R430	RES,1.00M,1/8W,1%,TF,SMD1206	20126.100.01				
R431	RES,324K,1/8W,1%,TF,SMD1206	20125.324.01				
R432	RES,20.0K,1/8W,1%,TF,SMD1206	20124.200.01				
R433	RES,1.00M,1/8W,1%,TF,SMD1206	20126.100.01				
R434	RES,4.99K,1/8W,1%,TF,SMD1206	20123.499.01				
R435	RES,1.00M,1/8W,1%,TF,SMD1206	20126.100.01				
R436	RES,4.99K,1/8W,1%,TF,SMD1206	20123.499.01				
R437	RES,1.00M,1/8W,1%,TF,SMD1206	20126.100.01				
R339	RES,1.00M,1/8W,1%,TF,SMD1206	20126.100.01				
R600	RES,3.01K,1/8W,1%,TF,SMD1206	20123.301.01				

Ref Des	Description	Orban P/N	Ven	Vendor P/N	Alternate Vendors	Notes
R601	RES,10K,1/8W,1%,TF,SMD1206	20124.100.01				
R602	RES,15K,1/8W,1%,TF,SMD1206	20124.150.01				
R603	RES,432Ω,1/8W,1%,TF,SMD1206	20122.432.01				
	Resistors (continued)					
R605	RES,13.3K,1/8W,1%,TF,SMD1206	20124.133.01				
R606	RES,3.01K,1/8W,1%,TF,SMD1206	20123.301.01				
R607	RES,3.16K,1/8W,1%,TF,SMD1206	20123.316.01				
R608	RES,2.15K,1/8W,1%,TF,SMD1206	20123.215.01				
R611	RES,10Ω,1/8W,1%,TF,SMD1206	20121.100.01				
R614	RES,75.0K,1/8W,1%,TF,SMD1206	20124.750.01				
R700-32	RES,100K,1/8W,1%,TF,SMD1206	20125.100.01				
VR500	Trimpot, Cermet, 20 Turn; 1K	20512-210	BEK	89PR1K	BRN	
VR501	Trimpot, Cermet, 20 Turn; 1K	20512-210	BEK	89PR1K	BRN	
	Switches					
Q300-1	Transistor, JFET/N	23402-101	NAT	J108		
	Miscellaneous					
J100	CONN,"D",R-ANG,PCMOUNT;9P	27017.009.01				
J101	Connector, D Type, 25-pin	27017-025	AD	JMDF-25S		
J300	Connector, XLR, PC Mount, Female	27054-003	NEU	NC 3 FD-H		
J301-3						
J308	CONN,JUMPER RECEPTACLE,MINI;BLK	27401.000.01				
J400-1	Connector, XLR, PC Mount, Male	27053-003	NEU	NC 3 MD-H		
Y602	Oscillator; 16.384MHz	28074-001	ORB			
	ASSY,EPROM,6200 MAIN BD,V1.01	44063.101				

DISPLAY BOARD

Ref Des	Description	Orban P/N	Ven	Vendor P/N	Alternate Vendors	Notes
	Capacitors					
C1	Cap,6.8uF,25V,10%, Tantalum Chip SMT	21313.568.01				
C2-10	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
	Diodes					
CR1-6	LED ARRAY, 9-YEL, 1-RED	25168.000.01				
CR7	LED ARRAY, 1-RED,1-YEL, 8-GRN	25167.000.01	LUM			
CR8	LED ARRAY, 1RED, 6YEL, 3GRN	25170.000.01				
CR9	LED ARRAY,10POS,1RED,9GREEN	25169.000.01				
CR10	LED, Red	25106-003	HP	HLMP-1300	GI	
CR11-13	LED, Yellow	25106-001	HP	HLMP-1400	GI	
CR14-15	LED, Yellow	25106-001	HP	HLMP-1400	GI	
	Integrated Circuits					
IC1-2	DATA LATCH,SMT	24908.000.01	SIG	74AC574		
IC3	Digital, Latch	24857-000	MOT	MC74HC374ADW		
IC4-6	DATA LATCH	24905.000.01	SIG	74FCT574SO		
IC7	Digital, Inverter	24900-000	TI	SN74HC14AD		
IC8	Digital, Transceiver	24851-000	SIG	74HC245D		
	Resistors					
	RES,274Ω,1/8W,1%,TF,SMD1206	20122.274.01				
	RES,NET,DIL,2%,100Ω,SMT,Isolated Res	20226.000.01				
R17-24	RES,110Ω,1/8W,1%,TF,SMD1206	20122.110.01				
R17-24	RES,274Ω,1/8W,1%,TF,SMD1206	20122.274.01				
R25-28	RES,100K,1/8W,1%,TF,SMD1206	20125.100.01				
R29-30	RES,10K,1/8W,1%,TF,SMD1206	20124.100.01				
RP1-2	RES,NET,DIL,2%,100Ω,SMT,Isolated Res	20226.000.01				

Ref Des	Description	Orban P/N	Ven	Vendor P/N	Alternate Vendors	Notes
POWER SUPPLY						
Capacitors						
C1-2	Alum, Radial, 35V; 1000uF	21256-000	PAN	ECEA1VGE102		
C3-4	Ceramic, Axial, 50V, 20%; 0.1uF	21129-410	KEM	C410C104M5UCA		
C5-6	Alum., Radial, 25V, 10%; 100uF	21263-710	NIC	UKLIE101KPAANA		
C7-10	Ceramic, Axial, 50V, 20%; 0.1uF	21129-410	KEM	C410C104M5UCA		
C11	Alum., Radial, 25V, 10%; 100uF	21263-710	NIC	UKLIE101KPAANA		
C12-13	Ceramic, Axial, 50V, 20%; 0.1uF	21129-410	KEM	C410C104M5UCA		
C14	Alum., Radial, 25V, 10%; 100uF	21263-710	NIC	UKLIE101KPAANA		
C15	Alum, Radial, 16V; 6800uF	21255-000	PAN	ECOS1CA682AA		
C16-17	Ceramic, Axial, 50V, 20%; 0.1uF	21129-410	KEM	C410C104M5UCA		
C18	Alum, Radial, 25V; 47uF	21206-747	PAN	ECEAIEU471		
C19	Alum, Radial, 35V; 1000uF	21256-000	PAN	ECEA1VGE102		
C20	Ceramic, Axial, 50V, 20%; 0.1uF	21129-410	KEM	C410C104M5UCA		
Diodes						
CR1-CR9	Diode, Rectifier, 400V, 1A	22201-400	MOT	1N4004	MANY	
CR10	Diode, Zener, 1W, 5%; 5.6V	22004-056	MOT	1N4734A		
CR11	Diode, Rectifier, 400V, 1A	22201-400	MOT	1N4004	MANY	
CR12	Diode, Zener, 1W, 5%; 5.6V	22004-056	MOT	1N4734A		
CR13-15	Diode, Rectifier	22015-000	TAT	SBL-1630CT		
CR16	Diode, Zener, 1W, 5%; 5.6V	22004-056	MOT	1N4734A		
CR17-19	Diode, Rectifier, 400V, 1A	22201-400	MOT	1N4004	MANY	
CR20	Diode, Signal	22101-000	FSC	1N4148	MANY	
Integrated Circuits						
IC1-2	TRANSISTOR MTG. KIT;TO-220	15025.000.01				
IC1	D.C. Regulator, 15V Positive	24304-901	NAT	LM78M15UC	TI,MOT	
IC2	D.C. Regulator, 15V Negative	24303-901	NAT	LM79M15AUC	TI,MOT	
IC3-4	HEATSINK,TO-220	16013.000.01				
IC4	D.C. Regulator, 5V Negative	24308-901	NAT	LM79M05C	TI,MOT	
IC5	Regulator, 5V	24321-000	LT	LT1086CK-5		
IC13	D.C. Regulator, 5V Positive	24307-901	NAT	LM78M05C	TI,MOT	
Q1	HEATSINK,TO-220	16013.000.01				
Resistors						
R3-6	Resistor, CF, 1/2W, 5%; 2.0Ω	20021-920	ORB			
	RES,MO,1W,5%,120Ω	20140.120.01				
V1,V2	Varistor	22500-271	PAN	ERZ-C10DK271U		
Switches						
S1	Switch, Slide, Mains voltage selector	26143-000	SW	EPS2-PC3		
S2	Switch, Slide, SPDT	26146-000	ECG	SSP1-S1-M7-Q-E-A		
Transistors						
Q1	Transistor, Power, NPN	23606-201	TI	TIP120		
Q2	Transistor, Signal, NPN	23202-101	MOT	2N4400	FSC	
Miscellaneous						
F1	Fuse, 3AG, Slo-Blo, 1/2A	28004-150	LFE	313.500	BUS	
F1	Fuseholder, PC Mount	28112-001	LFE	345-101-01		
L1	Filter, Line	28012-000	DEL	03ME1		

Ref Des	Description	Orban P/N	Ven	Vendor P/N	Alternate Vendors	Notes
FINAL ASSEMBLY						
Miscellaneous						
	ASSY,CABLE,ROTARY,SIGNAL,6200	43026.006				
	LCD,GRB,BCKLGHT,GRY FLUID	25404.001.01	DEN	HC4402FGSNG0161		
	PCA NRSC MON.ROLL.FILTR; 9100B	31020.000				
	RECEPTACLE,2.8MMX0.5MM TAB	27746.001.01				
	SW,ROTARY ENCODER,VERT MNT,2-BIT,NON-DE1	26085.000.01	NOB	SDB161PVB20F 1 2 36		
	Switch, Slide; DPDT (Gold)	26106-000	CW	GF326-0149		
	XFMR, TOROID, 100V	55036.000.03				JPN
	XFMR,TOROID,115/230V	55035.000.03				USA/ EU

Vendor Codes

AB Rockwell Allen-Bradley
625 Liberty Ave
Pittsburgh, PA 15222-3123

AD Analog Devices, Inc.
2105 S Bascom Ave
Suite 325
Campbell, CA
95008

BEK Beckman Industrial Corporation
4141 Palm Street
Fullerton, CA 92635-1025

BRN Bourns, Inc
2533 N 1500 W
Ogden UT 84404

BUS Bussmann Division
Cooper Industries
PO Box 14460
St. Louis, MO 63178

CD Corning Incorporated
334 County
Route 16
Canton, NY 13617

CEN Mepcopal/Centralab
See Mepcopal

CSC Crystal Semiconductor Corp.
50 Airport Parkway
San Jose, CA 95110

CW CW Industries
130 James Way
Southampton, PA 18966

DAL Dale
1122 23rd St
Columbus, NE 68601-3647

DEL Delta Products Corp
3225 Laurel View Ct.
Fremont, CA 94538

DEN Densitron Corporation
P.O. BOX 11189
Torrance, CA 90510-1189

EXR Exar Corporation
2222 Qume Dr.
PO Box 49007
San Jose, CA 95161-9007

FR Fair-Rite Products Corp
PO Box J
1 Commerical Row
Willkill, NY 12589

FSC Fairchild Camera & Instr. Corp.
See National Semiconductor

GI General Instruments
Optoelectronics Division
See Quality Technologies

GS General Silicones Co. USA Inc.
650 W Duarte Rd, Ste 401
Arcadia, CA 91007

HP Hewlett-Packard Co.
321 E Evelyn Ave
Mountain View, CA 94039

KEM KEMET Electronics Corporation
Post Office Box 5928
Greenville, South Carolina 29606

LFE Littlefuse
A Subsidiary of Tracor, Inc.
800 E. Northwest Hwy
Des Plaines, IL 60016

LT Linear Technology Corp
1630 McCarthy Blvd.
Milpitas, CA 95035

LUM Lumex Opto/Components Inc.
292 E. Hellen Road
Palatine, IL 60067

MAT Matsushita Electric Corp
of America
One Panasonic Way
Secaucus, NJ 07094

MIL J.W. Miller Division
Bell Industries
306 E. Alondra
Gardena, CA 90247

MOT Motorola Semiconductor
5005 E McDowell Rd
Phoenix, AZ 85008

MUR Murate Erie North America
2200 Lake Park Drive
Smyrna, GA 30080

NAT National Semiconductor Corp.
2900 Semiconductor Drive
PO Box 61659
Santa Clara, CA 95051

NEU Neutrik USA Inc.
195 Lehigh Ave
Lakewood, NJ 08701-4527

NIC Nichicon
927 East State Parkway
Schaumburg, IL 60713

NOB Noble USA Inc.
5450 Meadowbrook Industri
Rolling Meadows, IL
60008-3800

OHM Ohmite Manufacturing Company
PO Box 49150
Chicago, IL 60678

ORB Orban, Inc.
1525 Alvarado Street
San Leandro, CA 94577

PHI Phillips Components – Signetics
See Signetics

PAN Panasonic Industrial Company
Two Panasonic Way
7E-2T
Secaucus, NJ 07094

QT Quality Technologies, Inc.
610 North Mary Ave.
Sunnyvale, CA 94086

ROH Rohm Electronics
3034 Owens Dr.
Antioch, TENN 37013

SIE Siemens Components Inc.
Heimann Systems Div.
186 Wood Avenue South
Iselin, NJ 08830

SIG Signetics - Philips Components
North American Phillips Corp.
811 E. Arques
Sunnyvale, CA 94088

SPE Spectrol
4051 Greystone Drive
Ontario, CA 91761

SPR Sprague Magnetics, Inc
15720 Stagg Street
Van Nuys, CA 91406

SW Switchcraft
A Raytheon Company
5555 N. Elation Avenue
Chicago, IL 60630

TAI Taiga America, Inc.
700 Frontier Way
Bensenville, IL 60106

TAT Taitron Components, Inc.
25202 Anza Drive
Santa Clarita, CA 91355

TI Texas Instruments, Inc.
PO Box 655012
Dallas, TX 75265

TOS Toshiba America, Inc.
9740 Irvine Blvd.
Irvine, CA 92718

TRW TRW Electronics Components
Connector Division
1501 Morse Avenue
Elk Grove Village, IL 60007

XIC Xicor, Inc.
1511 Buckeye Drive
Milpitas, CA 95035

WIM Wima Division
2269 Saw Mill Rd
Building 4C
PO Box 217
Elmsford, NY 10533

ZI ZILOG Inc.
210 Hacienda Ave.
Campbell, CA 95008

Schematics, Assembly Drawings

The following drawings are included in this manual:

Page	Function	Circuit Board	Drawing
6-32	Block Diagram		Assembly Drawing
6-33	Audio Processing	Main	Assembly Drawing
6-34	CPU/Remote/RS232		Schematic 1 of 7
6-35	Analog Input		2 of 7
6-36	Analog Output		3 of 7
6-37	Clock & Digital I/O		4 of 7
6-38	DSP 1		5 of 7
6-39	DSP 2		6 of 7
6-40	Power Distribution		7 of 7
6-41	Display, Controls	Display, 6200	Assembly Drawing
6-42	Display, Controls		Schematic 1 of 1
6-43	Display, Controls	Display Front, 6200S	Assembly Drawing
6-44	Display, Controls		Schematic 1 of 1
6-45	Display, Controls	Display Back, 6200S	Assembly Drawing
6-46	Display, Controls		Schematic 1 of 1
6-47	Power Supply	Power Supply	Assembly Drawing
6-48	Power Supply		Schematic 1 of 1

These drawings reflect the actual construction of your unit as accurately as possible. Any differences between the drawings and your unit are almost undoubtedly due to product improvements or production changes since the publication of this manual.

If you intend to replace parts, please read page 6-20.

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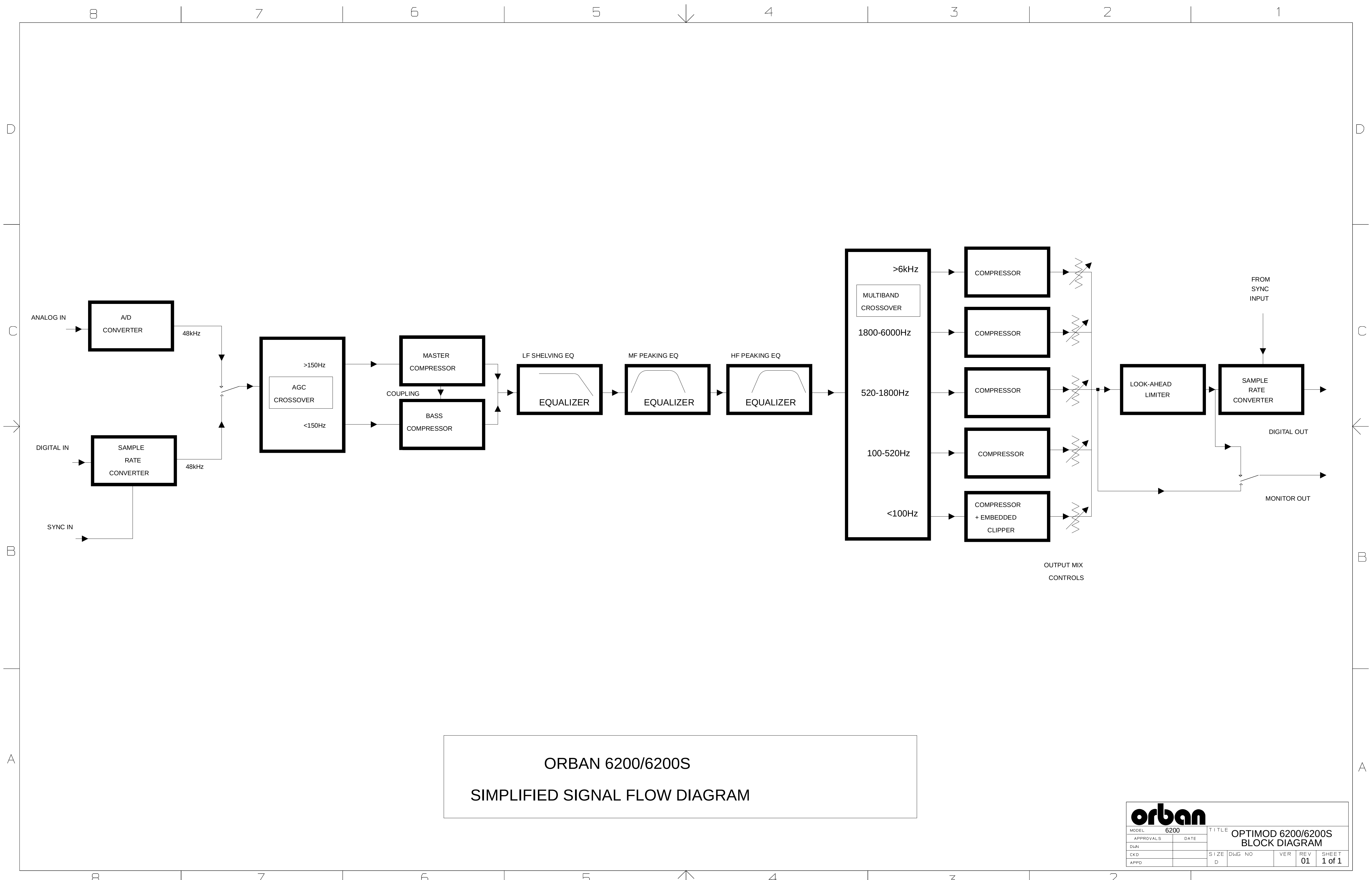
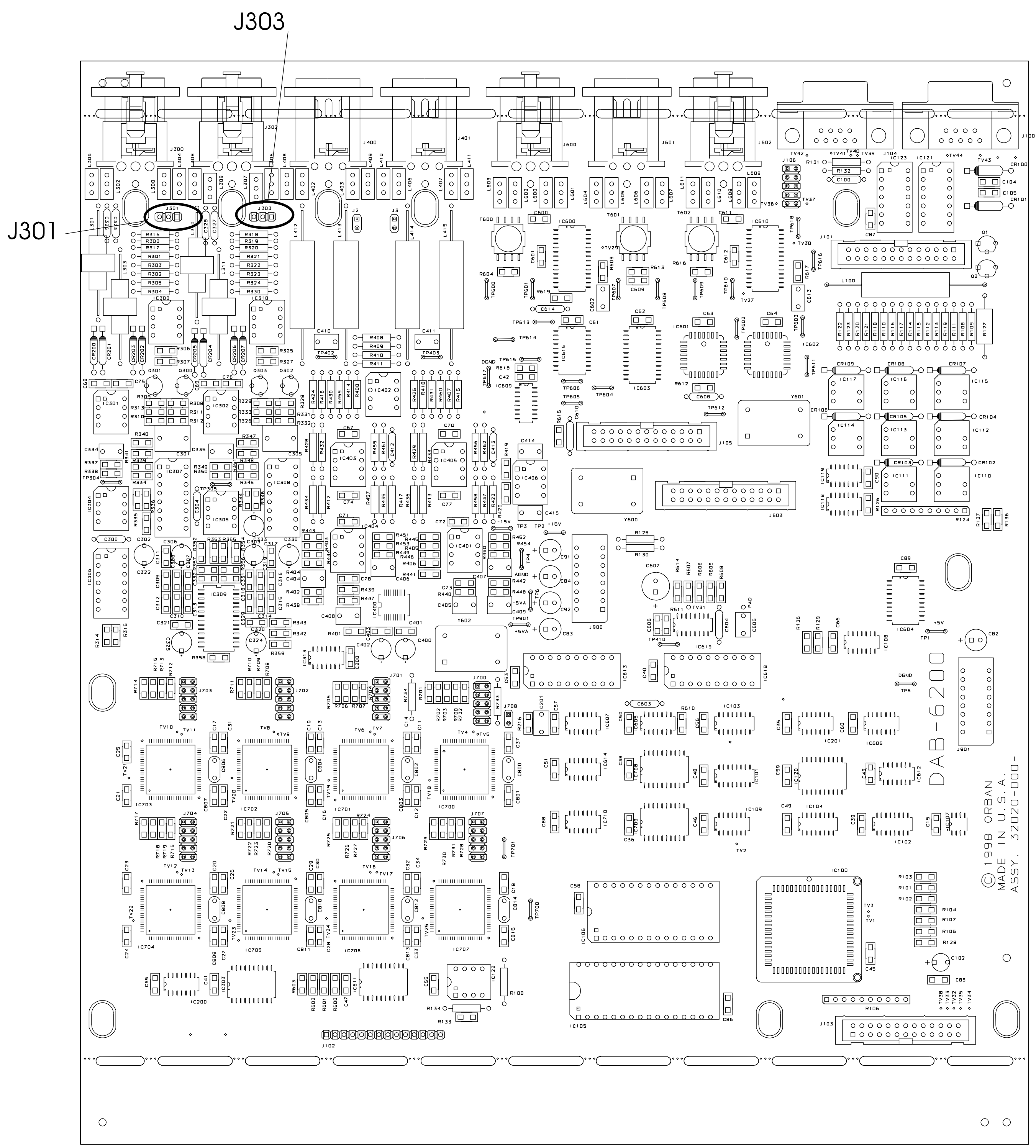

ANALOG IN
A/D CONVERTER
48kHz
DIGITAL IN
SAMPLE RATE CONVERTER
48kHz
SYNC IN
>150Hz
AGC CROSSOVER
<150Hz
MASTER COMPRESSOR
COUPLING
BASS COMPRESSOR
LF SHELVING EQ
EQUALIZER
MF PEAKING EQ
EQUALIZER
HF PEAKING EQ
EQUALIZER
>6kHz
MULTIBAND CROSSOVER
1800-6000Hz
520-1800Hz
100-520Hz
<100Hz
COMPRESSOR
COMPRESSOR
COMPRESSOR
COMPRESSOR
COMPRESSOR + EMBEDDED CLIPPER
OUTPUT MIX CONTROLS
LOOK-AHEAD LIMITER
FROM SYNC INPUT
SAMPLE RATE CONVERTER
DIGITAL OUT
MONITOR OUT
ORBAN 6200/6200S
SIMPLIFIED SIGNAL FLOW DIAGRAM
orban
MODEL 6200
APPROVALS
DATE
DWN
CKD
APPD
TITLE
OPTIMOD 6200/6200S
BLOCK DIAGRAM
SIZE D
DWG NO
VER
REV 01
SHEET 1 of 1



JUMPER SETTINGS
>10K BRIDGING (INPUT IMPEDANCE)

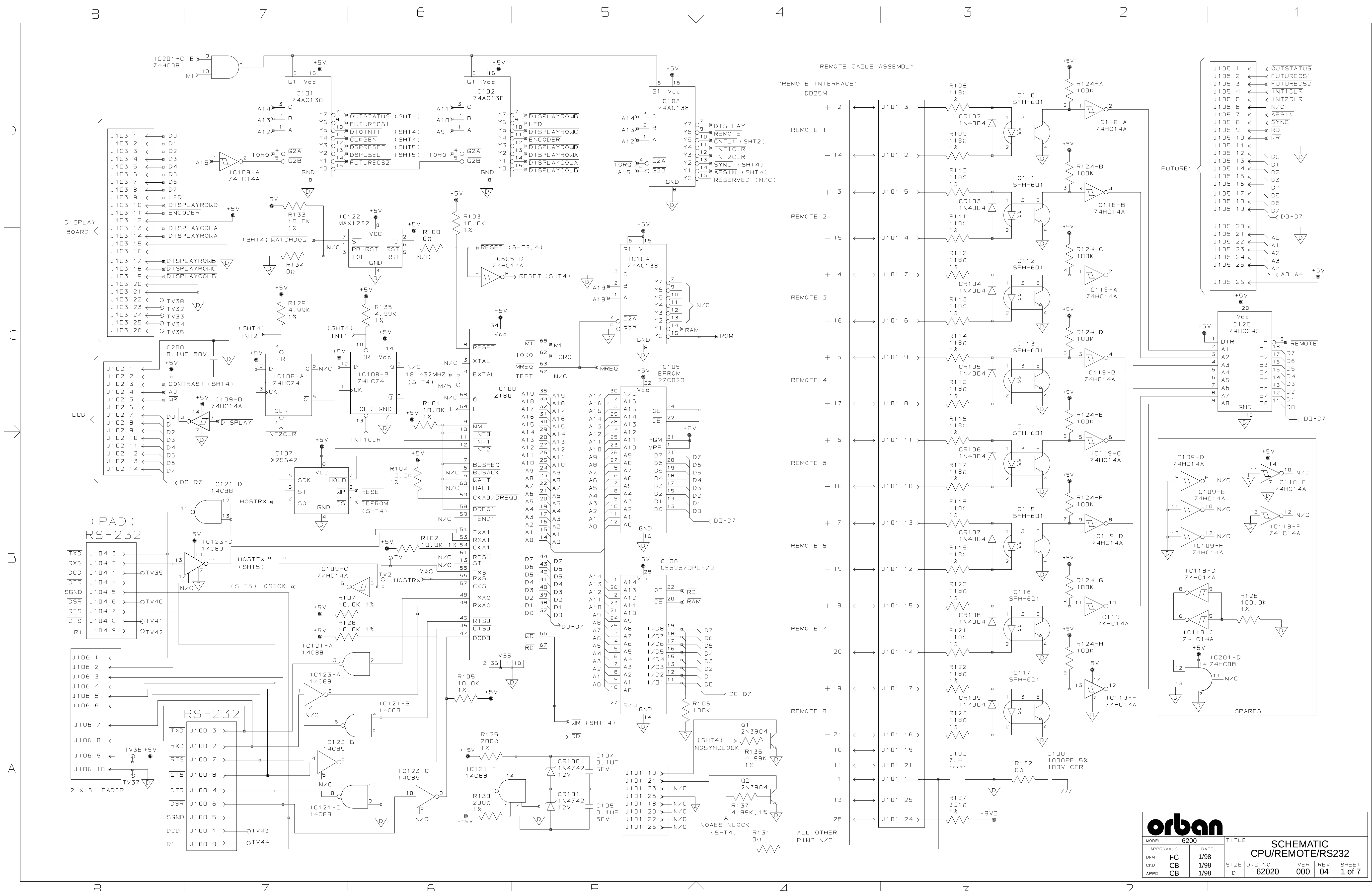
Left Channel	Right Channel
J301	J303

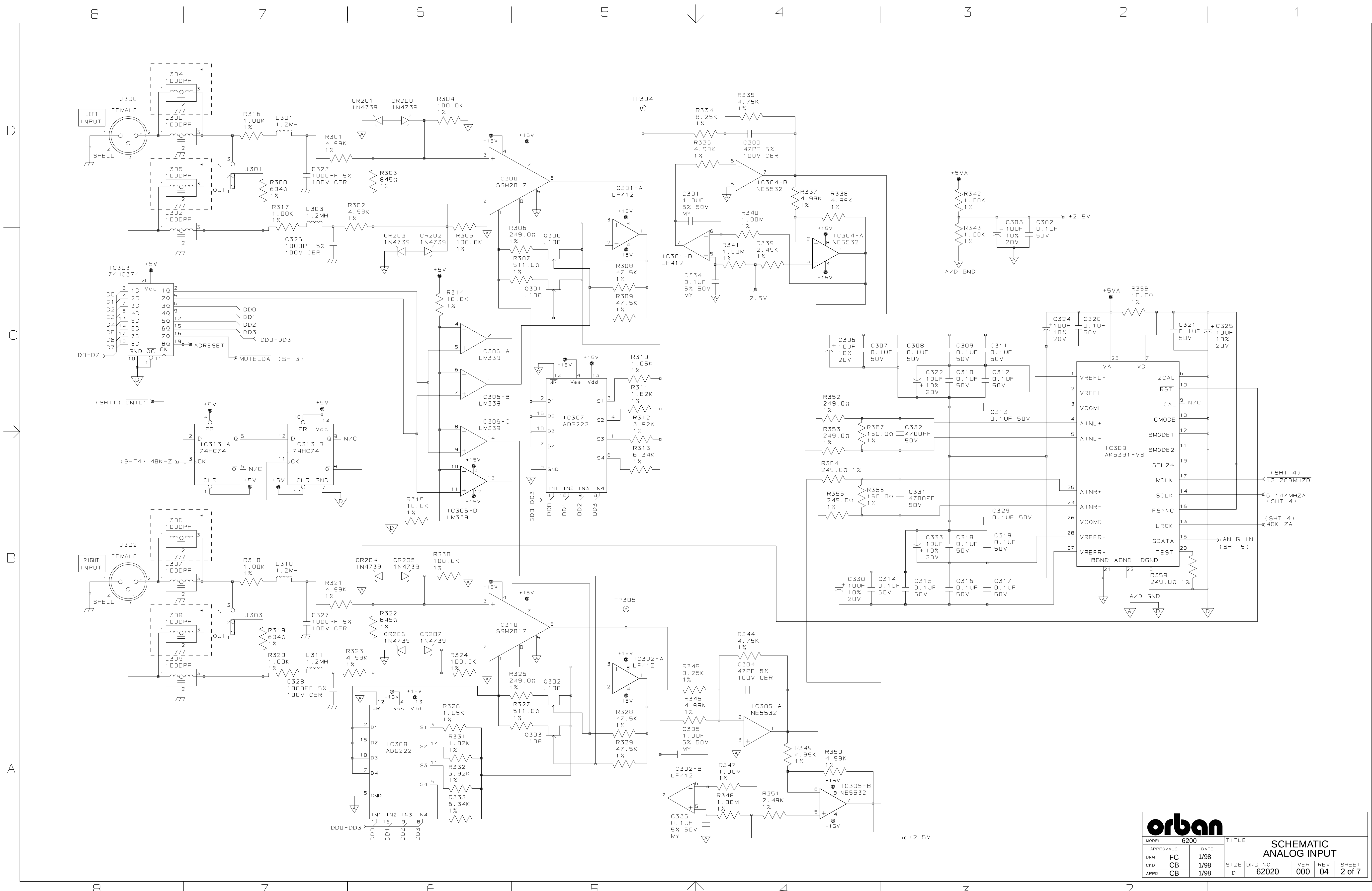
JUMPER SETTINGS
600Ω TERMINATION (INPUT IMPEDANCE)

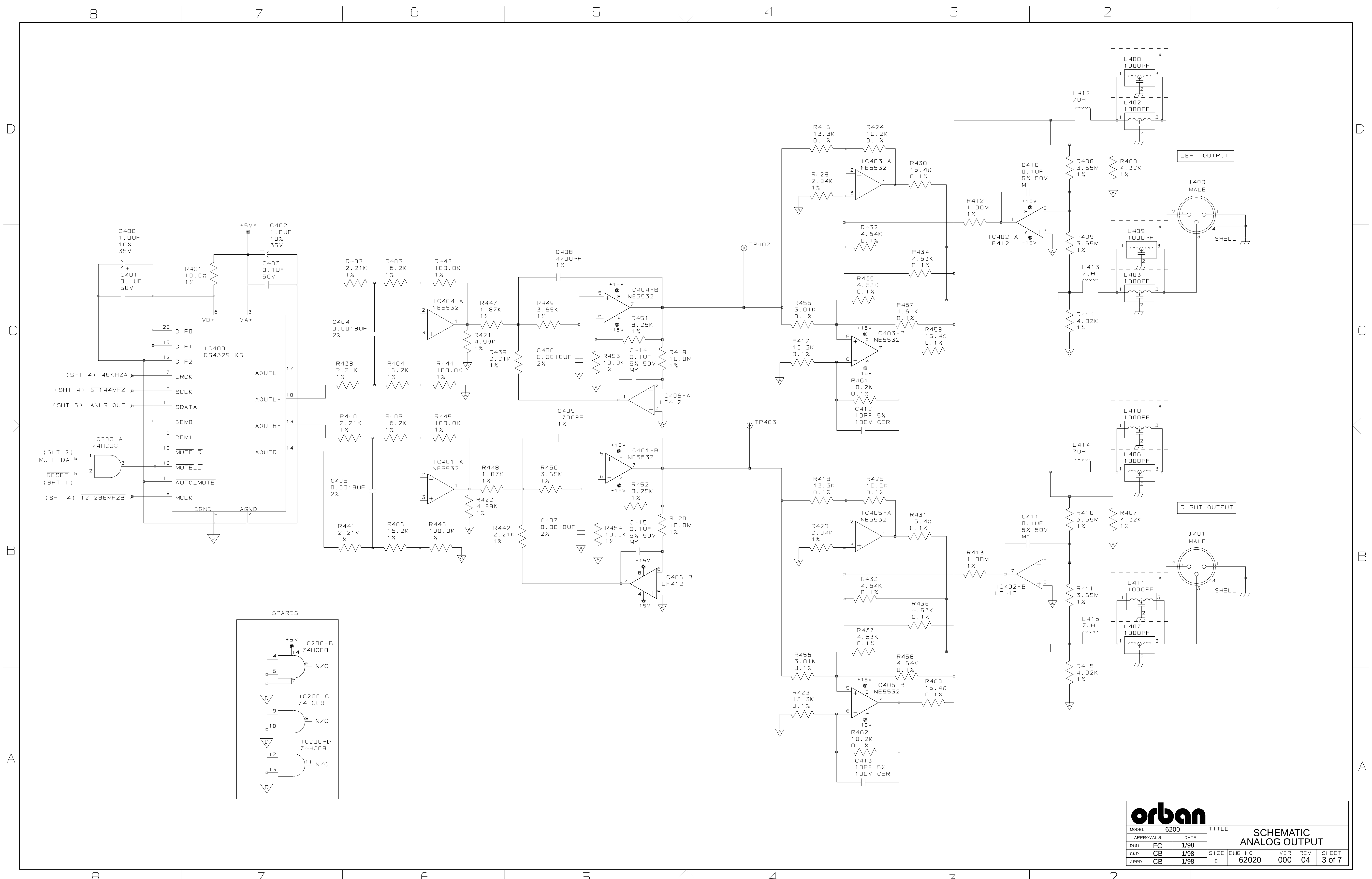
Left Channel	Right Channel
J301	J303

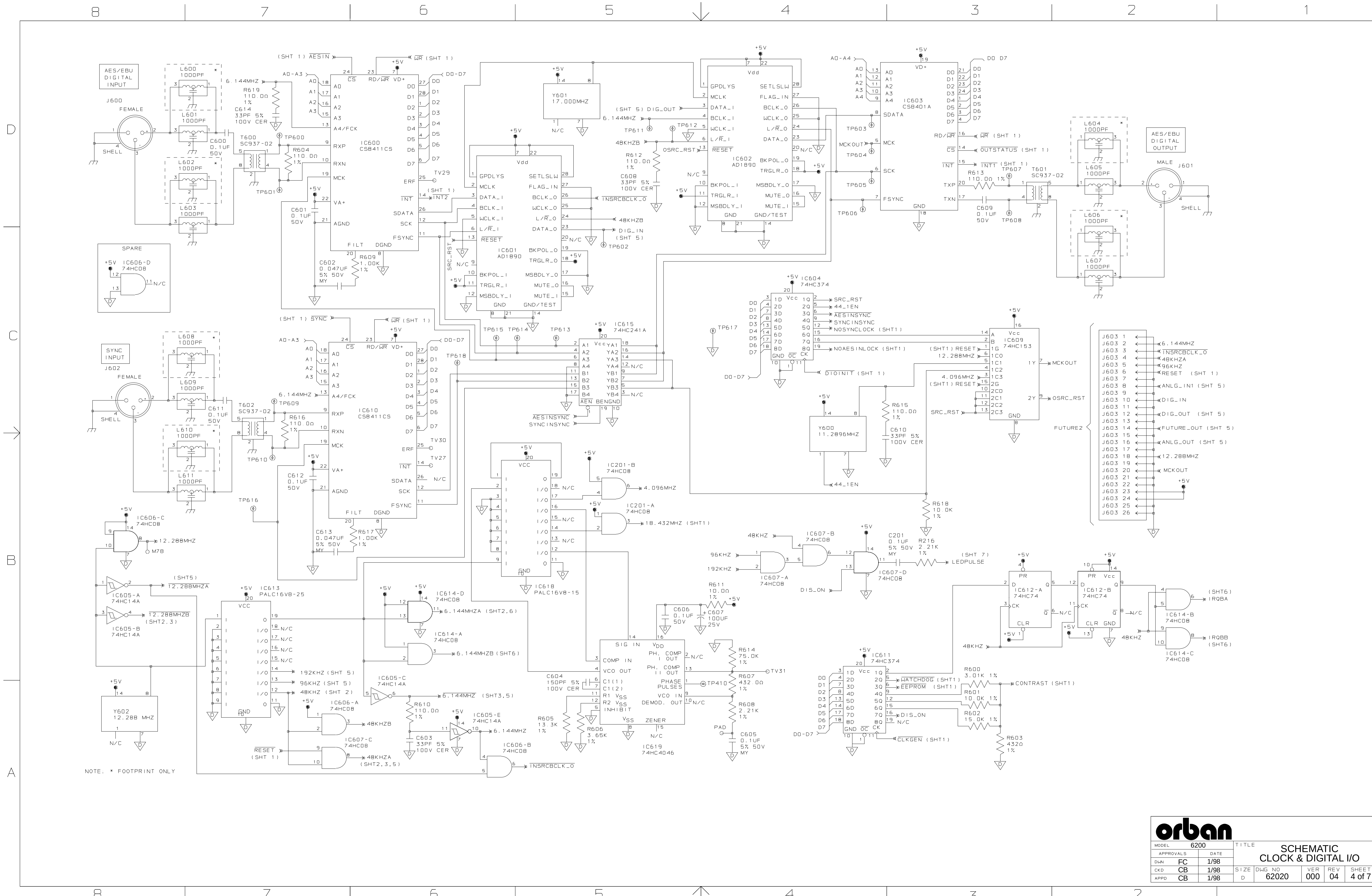
orban

MODEL	6200	TITLE	PCB ASSEMBLY MAIN			
APPROVALS		DATE				
DWN	FC	1/98				
CKD	CB	1/98	SIZE	DWG NO	VER	REV	SHEET
APPD	CB	1/98	D	32020	000	03	1 of 1

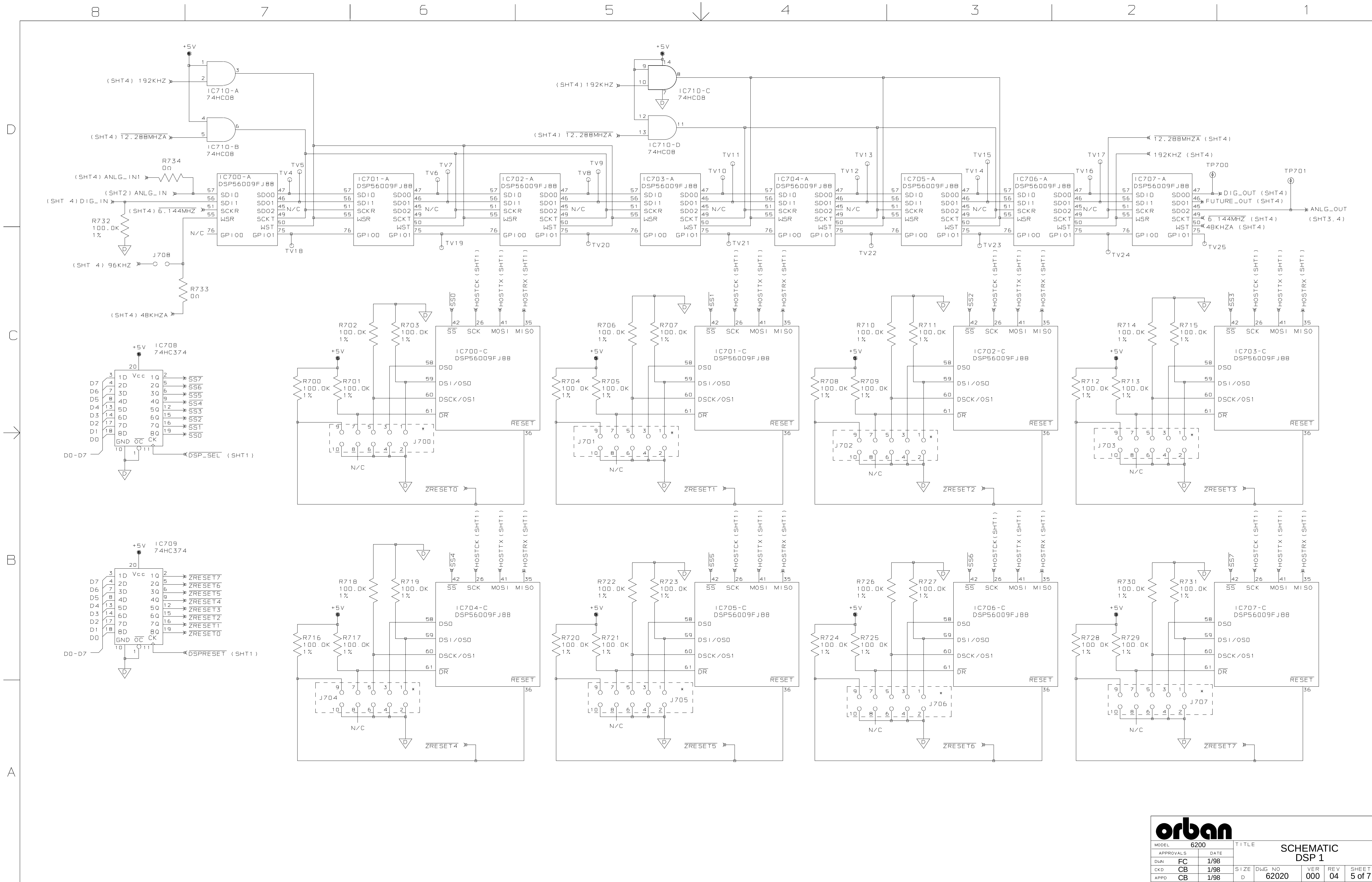

REMOTE CABLE ASSEMBLY
"REMOTE INTERFACE"
DB25M
FUTURE1
DISPLAY BOARD
LCD
SPARES
(PAD) RS-232
RS-232
2 X 5 HEADER
IC100 Z180
IC105 EPROM 27C020
IC106 TC55257DPL-70
IC120 74HC245
IC107 X25642
IC122 MAX1232
orban
SCHEMATIC
CPU/REMOTE/RS232
MODEL 6200
DWN FC 1/98
CKD CB 1/98
APPD CB 1/98
SIZE D
DWG NO 62020
VER 000
REV 04
SHEET 1 of 7

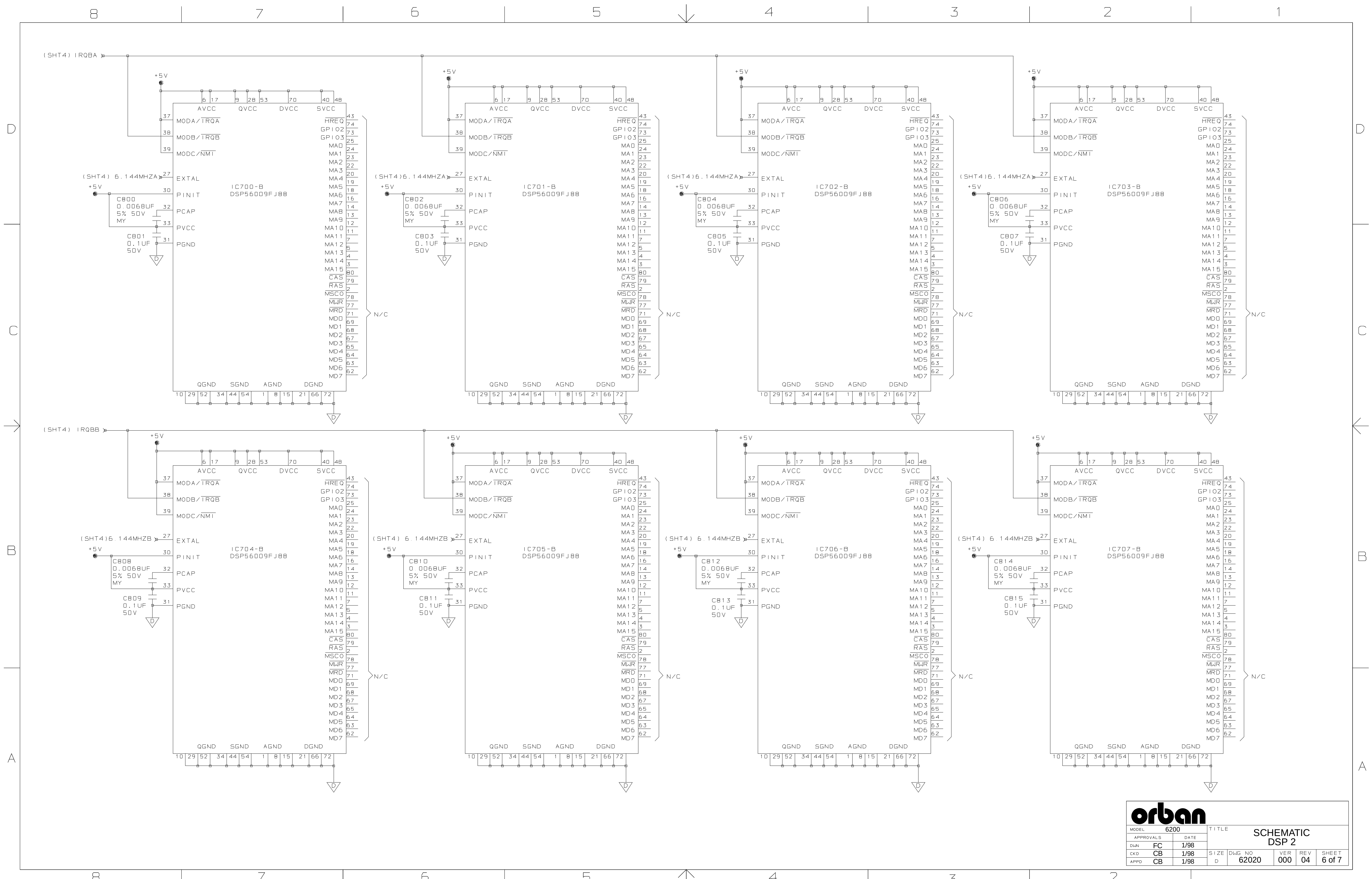

orban
MODEL 6200
TITLE
SCHEMATIC
ANALOG INPUT
APPROVALS DATE
DWN FC 1/98
CKD CB 1/98
APPD CB 1/98
SIZE D
DWG NO 62020
VER 000
REV 04
SHEET 2 of 7

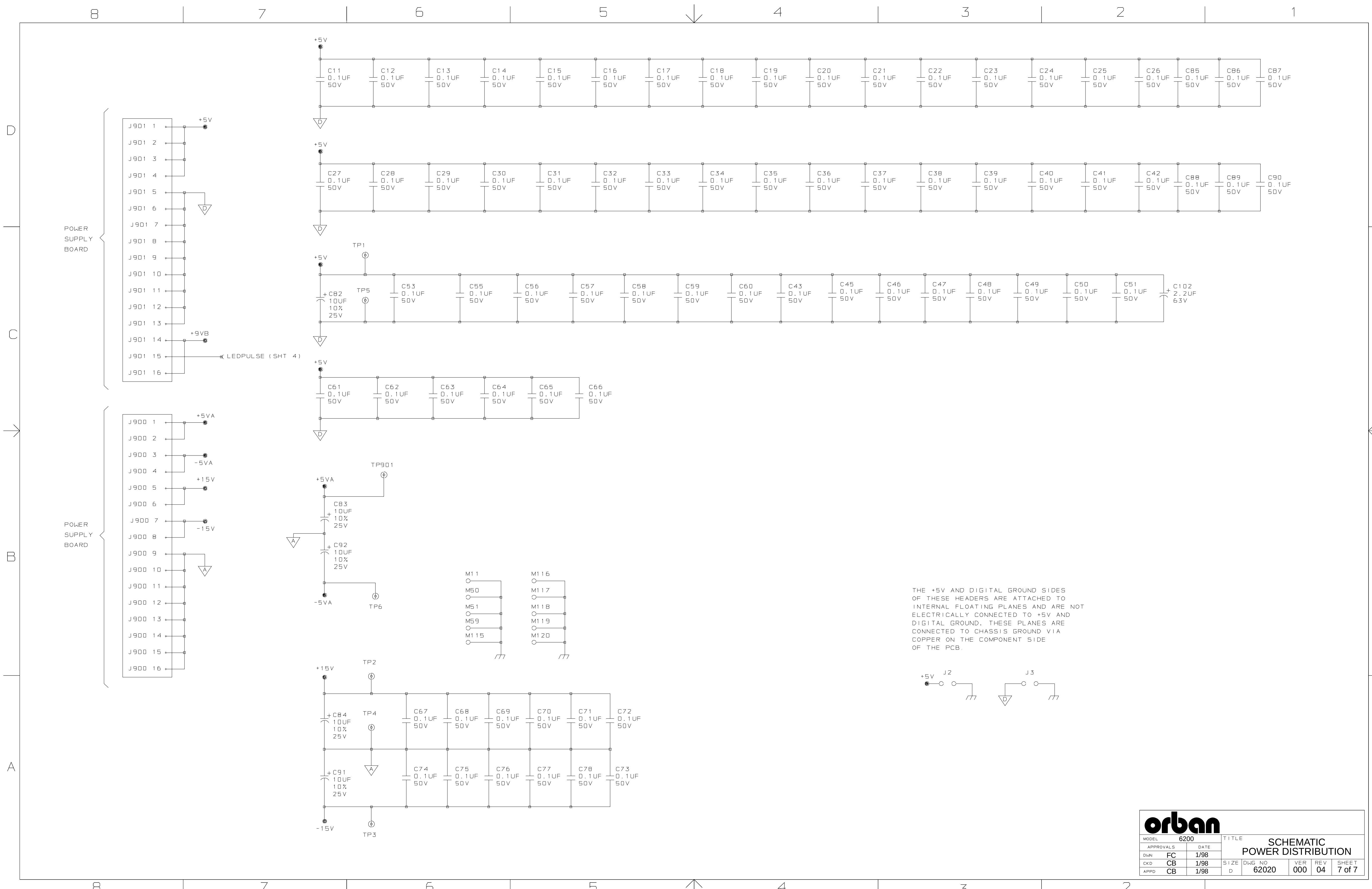

LEFT OUTPUT
RIGHT OUTPUT
J400 MALE
J401 MALE
SHELL
TP402
TP403
IC400 CS4329-KS
IC403-A NE5532
IC403-B NE5532
IC404-A NE5532
IC404-B NE5532
IC401-A NE5532
IC401-B NE5532
IC405-A NE5532
IC405-B NE5532
IC402-A LF412
IC402-B LF412
IC406-A LF412
IC406-B LF412
IC200-A 74HC08
(SHT 4) 48KHZA
(SHT 4) 6.144MHZ
(SHT 5) ANLG_OUT
(SHT 2) MUTE_DA
RESET (SHT 1)
(SHT 4) 12.288MHZB
SPARES
IC200-B 74HC08
IC200-C 74HC08
IC200-D 74HC08
N/C
orban
MODEL 6200
APPROVALS DATE
DWN FC 1/98
CKD CB 1/98
APPD CB 1/98
TITLE
SCHEMATIC
ANALOG OUTPUT
SIZE D
DWG NO 62020
VER 000
REV 04
SHEET 3 of 7

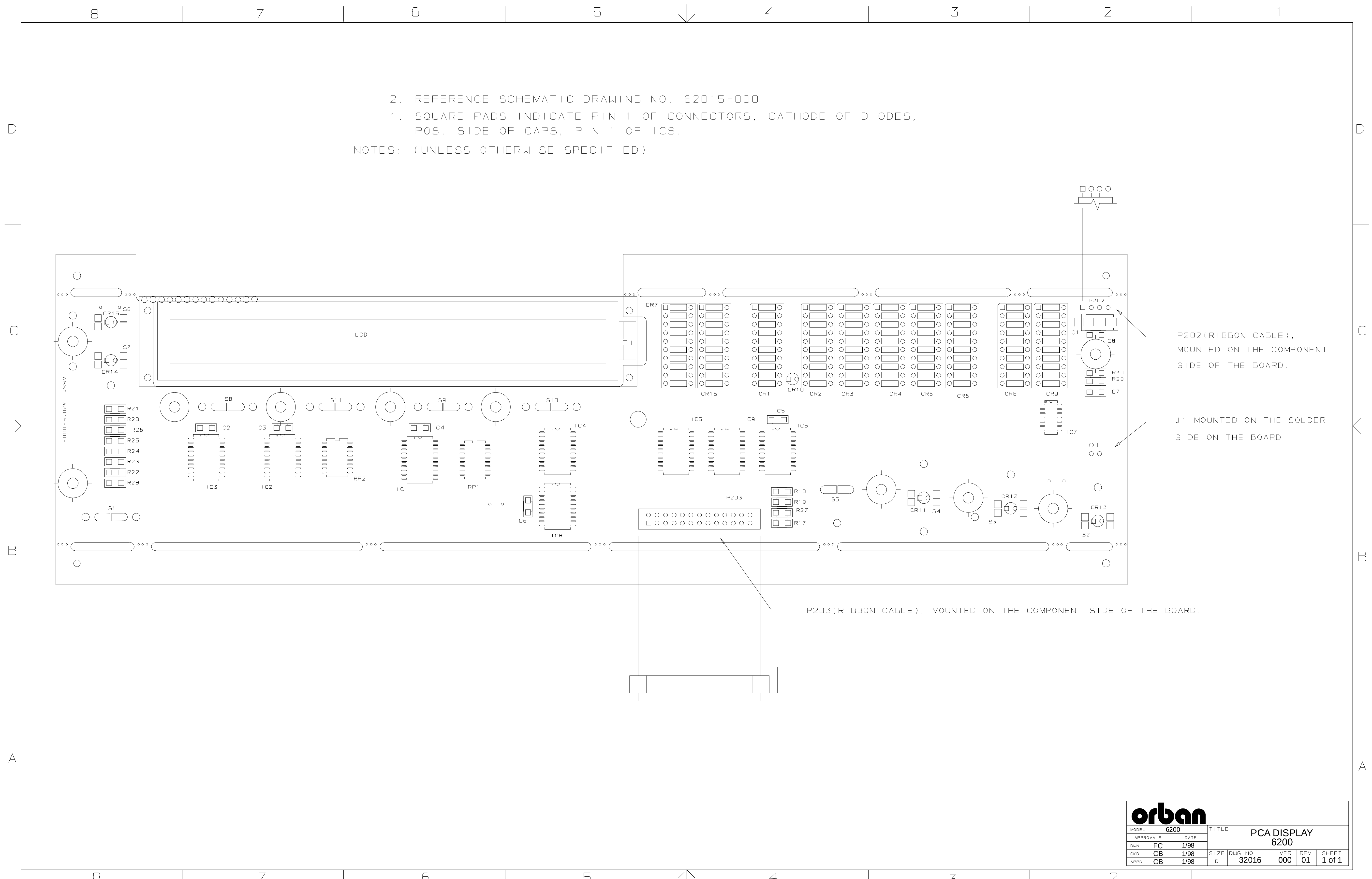


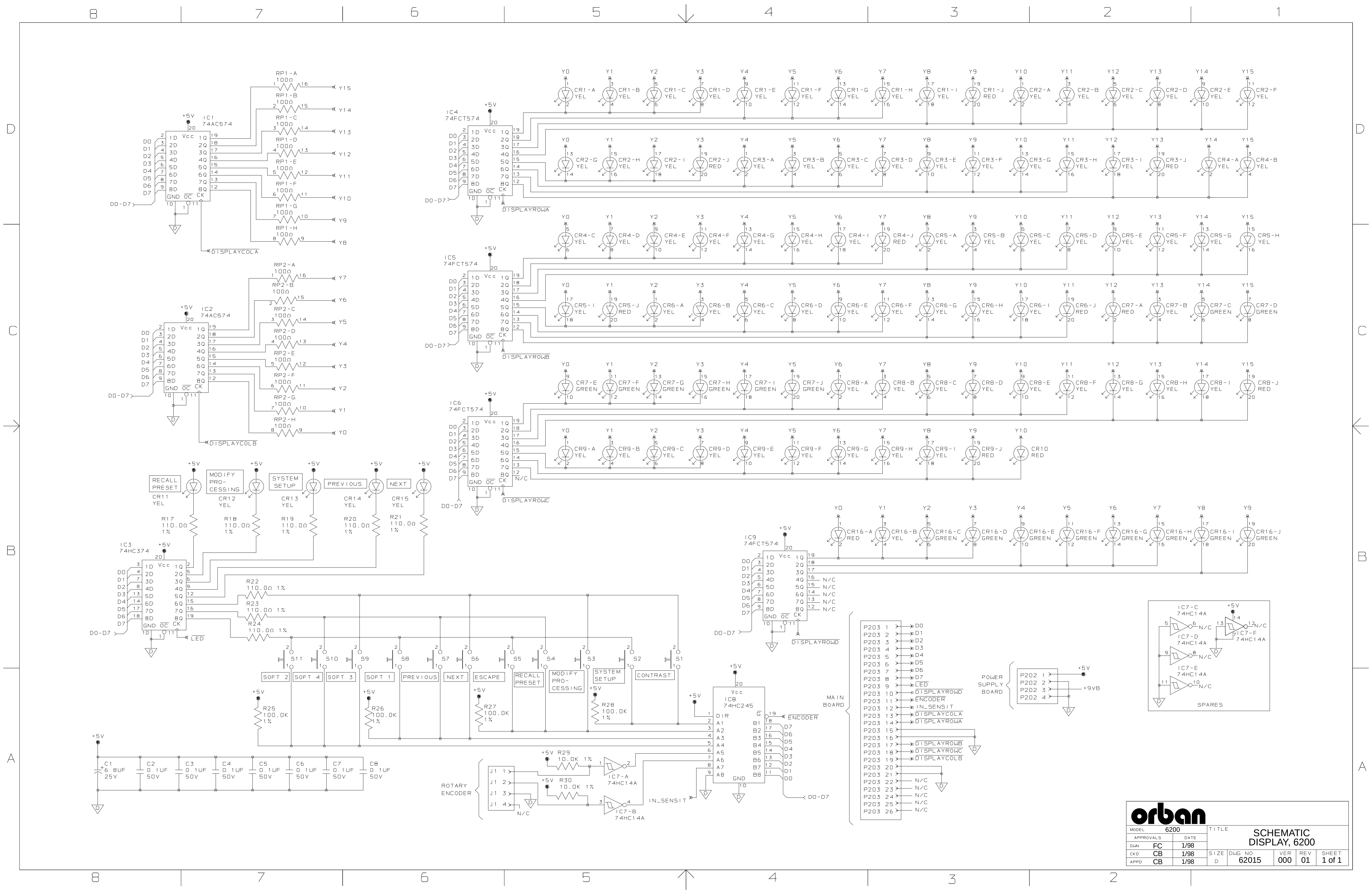
orban		TITLE	SCHEMATIC CLOCK & DIGITAL I/O	
MODEL	6200			
APPROVALS	DATE			
DWN FC	1/98			
CKD CB	1/98	SIZE D	DWG NO 62020	VER 000 REV 04 SHEET 4 of 7
APPD CB	1/98			


orban
MODEL 6200
APPROVALS DATE
DWN FC 1/98
CKD CB 1/98
APPD CB 1/98
TITLE
SCHEMATIC
DSP 1
SIZE D
DWG NO 62020
VER 000
REV 04
SHEET 5 of 7

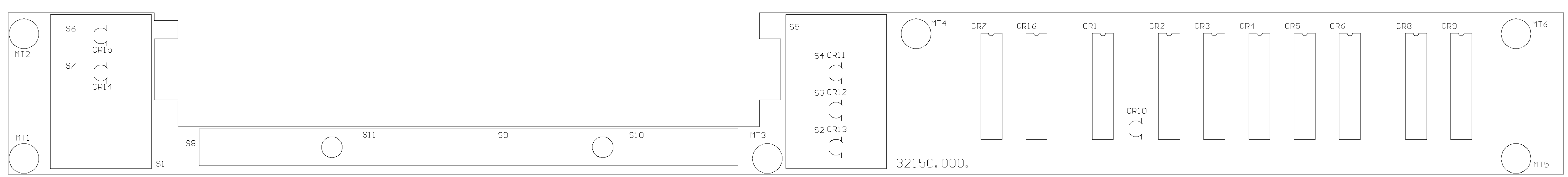

(SHT4) IRQBA
(SHT4) IRQBB
(SHT4) 6.144MHZA
(SHT4) 6.144MHZB
+5V
IC700-B DSP56009FJ88
IC701-B DSP56009FJ88
IC702-B DSP56009FJ88
IC703-B DSP56009FJ88
IC704-B DSP56009FJ88
IC705-B DSP56009FJ88
IC706-B DSP56009FJ88
IC707-B DSP56009FJ88
C800 0.0068UF 5% 50V MY
C801 0.1UF 50V
C802 0.0068UF 5% 50V MY
C803 0.1UF 50V
C804 0.0068UF 5% 50V MY
C805 0.1UF 50V
C806 0.0068UF 5% 50V MY
C807 0.1UF 50V
C808 0.0068UF 5% 50V MY
C809 0.1UF 50V
C810 0.0068UF 5% 50V MY
C811 0.1UF 50V
C812 0.0068UF 5% 50V MY
C813 0.1UF 50V
C814 0.0068UF 5% 50V MY
C815 0.1UF 50V
AVCC QVCC DVCC SVCC
MODA/IRQA MODB/IRQB MODC/NMI
EXTAL PINIT PCAP PVCC PGND
QGND SGND AGND DGND
HREQ GPIO2 GPIO3
MA0 MA1 MA2 MA3 MA4 MA5 MA6 MA7 MA8 MA9 MA10 MA11 MA12 MA13 MA14 MA15
CAS RAS MSCO MWR MRD
MD0 MD1 MD2 MD3 MD4 MD5 MD6 MD7
N/C
orban
MODEL 6200
APPROVALS DATE
DWN FC 1/98
CKD CB 1/98
APPD CB 1/98
TITLE SCHEMATIC DSP 2
SIZE D
DWG NO 62020
VER 000
REV 04
SHEET 6 of 7


POWER SUPPLY BOARD
J901 1 … J901 16
J900 1 … J900 16
+5V
+9VB
LEDPULSE (SHT 4)
+5VA
-5VA
+15V
-15V
TP1
TP2
TP3
TP4
TP5
TP6
TP901
C82 10UF 10% 25V
C83 10UF 10% 25V
C92 10UF 10% 25V
C84 10UF 10% 25V
C91 10UF 10% 25V
C102 2.2UF 63V
0.1UF 50V
M11
M50
M51
M59
M115
M116
M117
M118
M119
M120
J2
J3
THE +5V AND DIGITAL GROUND SIDES OF THESE HEADERS ARE ATTACHED TO INTERNAL FLOATING PLANES AND ARE NOT ELECTRICALLY CONNECTED TO +5V AND DIGITAL GROUND. THESE PLANES ARE CONNECTED TO CHASSIS GROUND VIA COPPER ON THE COMPONENT SIDE OF THE PCB.
orban
MODEL 6200
APPROVALS DATE
DWN FC 1/98
CKD CB 1/98
APPD CB 1/98
TITLE SCHEMATIC POWER DISTRIBUTION
SIZE D
DWG NO 62020
VER 000
REV 04
SHEET 7 of 7

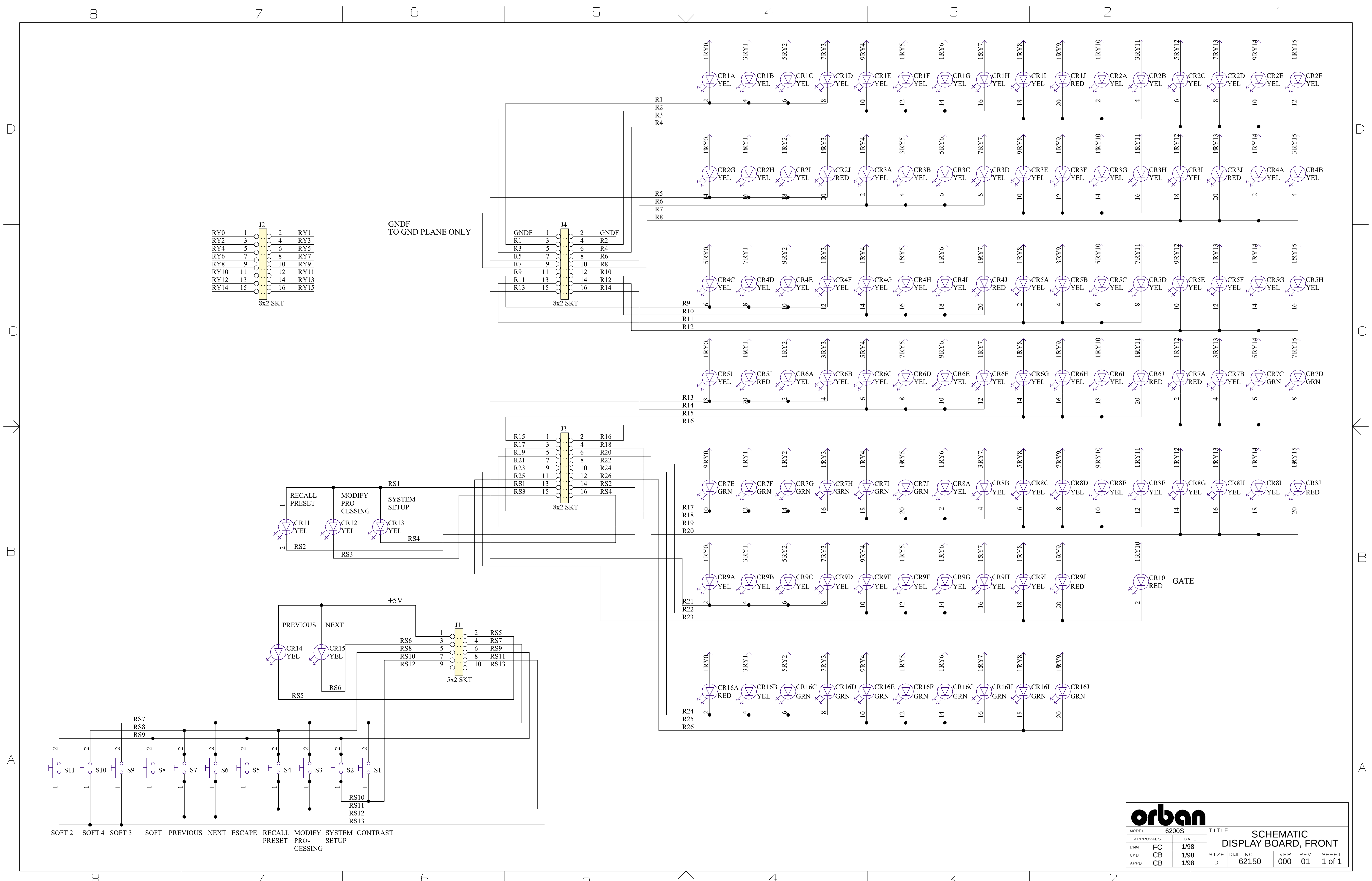

2. REFERENCE SCHEMATIC DRAWING NO. 62015-000
1. SQUARE PADS INDICATE PIN 1 OF CONNECTORS, CATHODE OF DIODES, POS. SIDE OF CAPS, PIN 1 OF ICS.
NOTES: (UNLESS OTHERWISE SPECIFIED)
P202(RIBBON CABLE), MOUNTED ON THE COMPONENT SIDE OF THE BOARD.
J1 MOUNTED ON THE SOLDER SIDE ON THE BOARD
P203(RIBBON CABLE), MOUNTED ON THE COMPONENT SIDE OF THE BOARD
LCD
ASSY 32015-000-
orban
MODEL 6200
TITLE PCA DISPLAY 6200
DWN FC 1/98
CKD CB 1/98
APPD CB 1/98
SIZE D
DWG NO 32016
VER 000
REV 01
SHEET 1 of 1



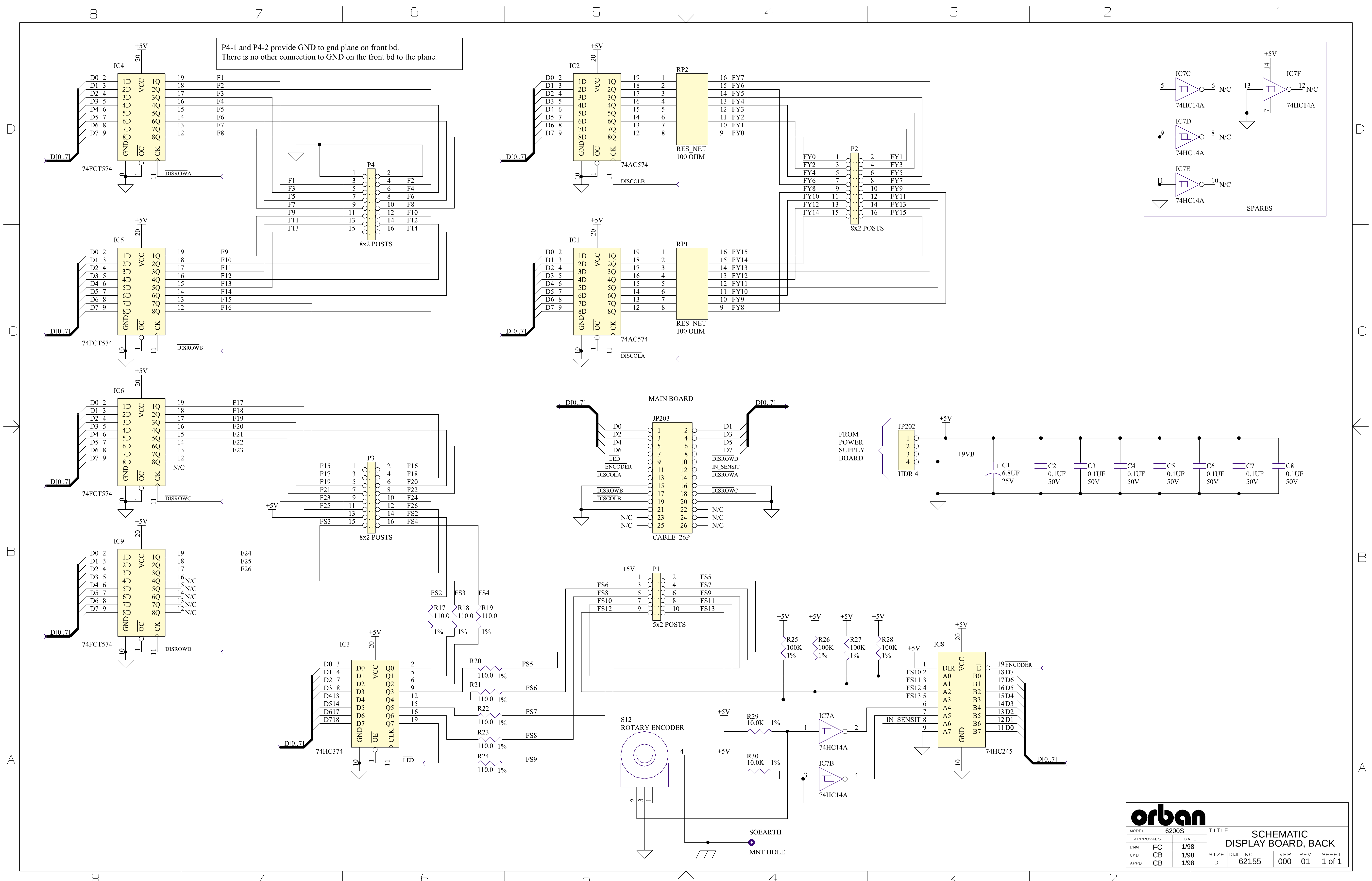
orban
MODEL 6200
APPROVALS DATE
DWN FC 1/98
CKD CB 1/98
APPD CB 1/98
TITLE
SCHEMATIC
DISPLAY, 6200
SIZE D
DWG NO 62015
VER 000
REV 01
SHEET 1 of 1
IC1 74AC574
IC2 74AC574
IC3 74HC374
IC4 74FCT574
IC5 74FCT574
IC6 74FCT574
IC8 74HC245
IC9 74FCT574
RECALL PRESET
MODIFY PRO-CESSING
SYSTEM SETUP
PREVIOUS
NEXT
ESCAPE
CONTRAST
SOFT 1
SOFT 2
SOFT 3
SOFT 4
ROTARY ENCODER
MAIN BOARD
POWER SUPPLY BOARD
SPARES
DISPLAYCOLA
DISPLAYCOLB
DISPLAYROWA
DISPLAYROWB
DISPLAYROWC
DISPLAYROWD
ENCODER
IN_SENSIT
LED

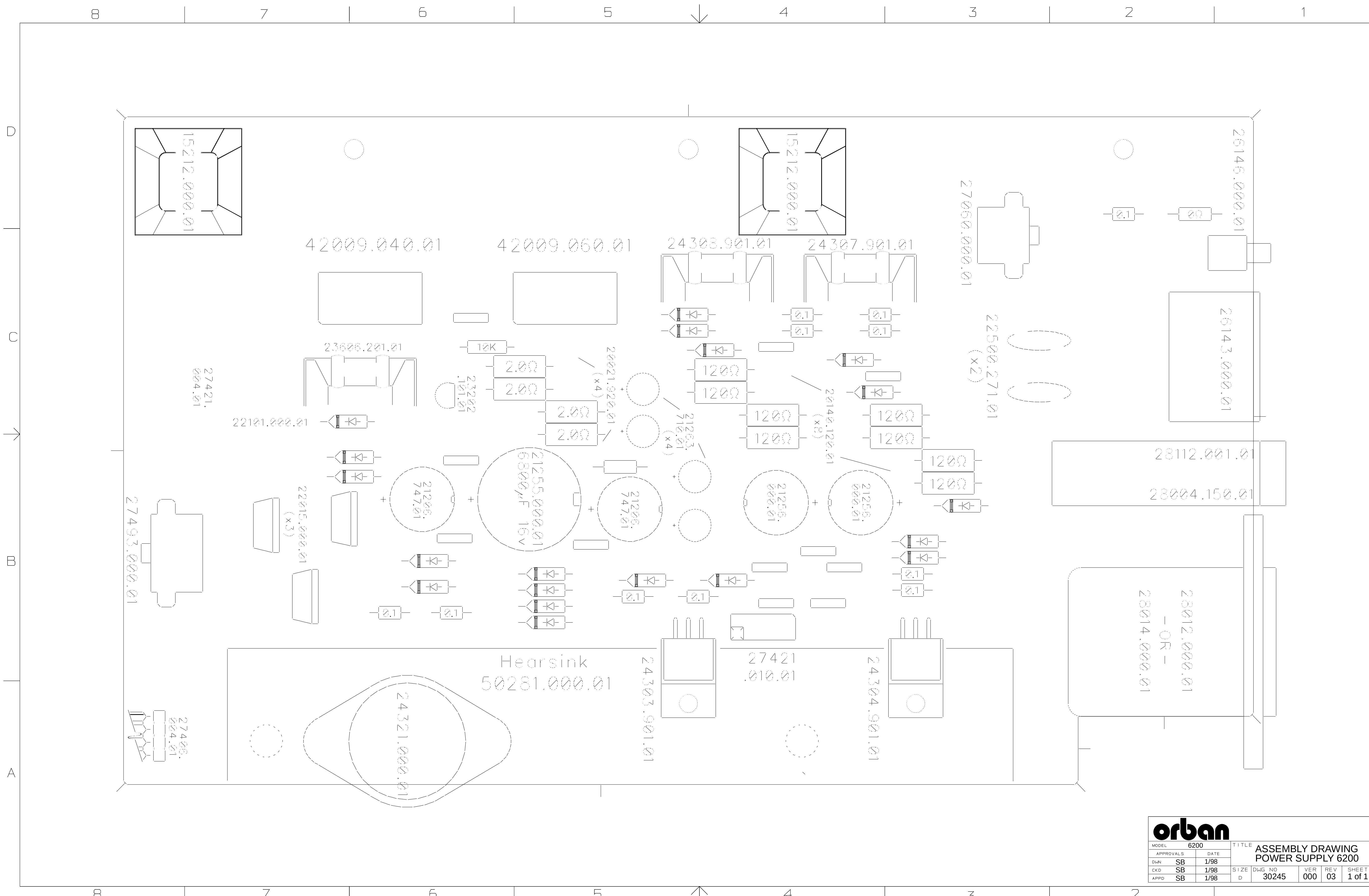


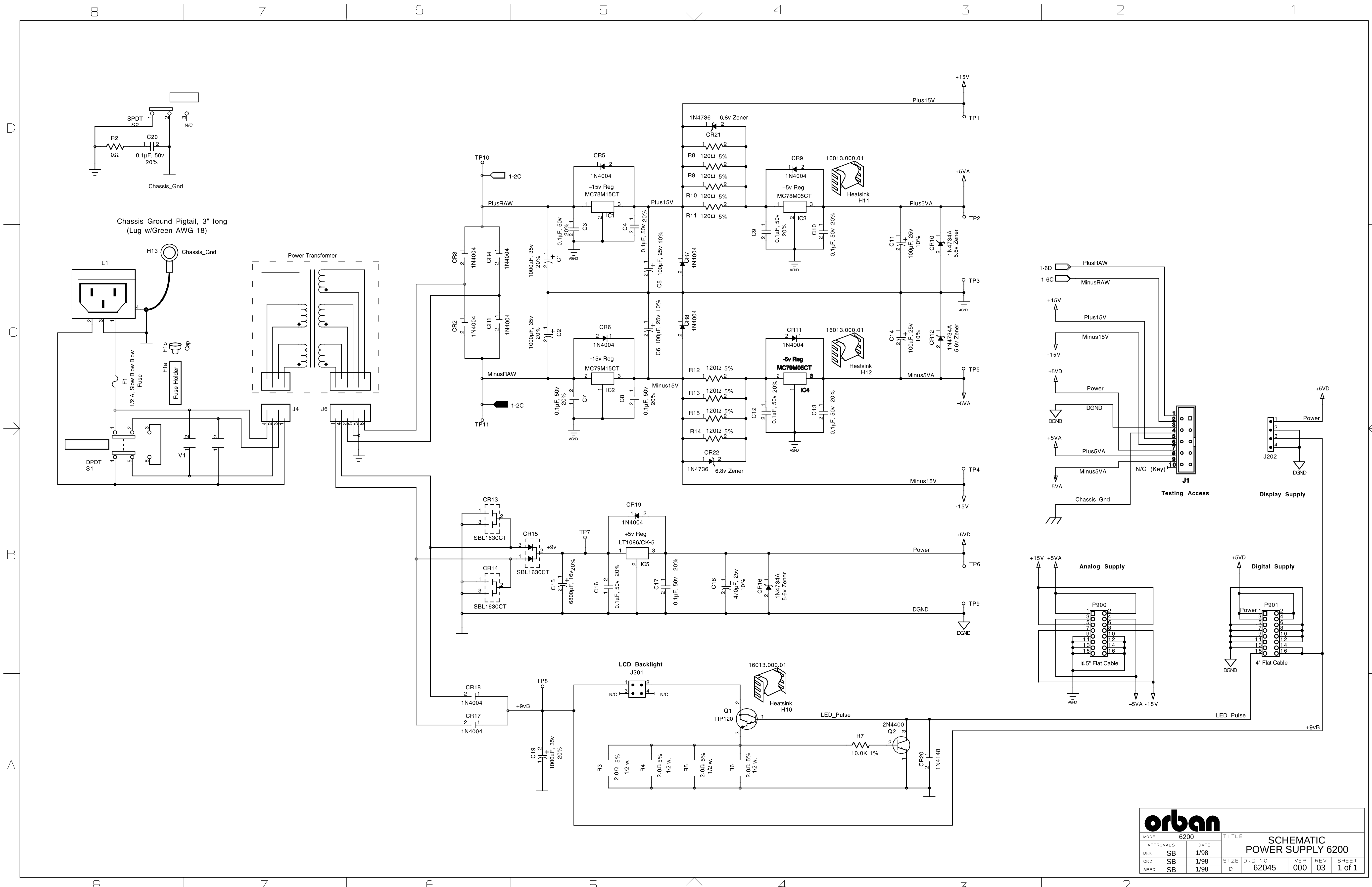
orban		TITLE	PCA DISPLAY, FRONT 6200S
MODEL	6200S		
APPROVALS	DATE		
DWN SB	1/98	SIZE D	DWG NO 32150
CKD CB	1/98	VER 000	REV 01
APPD CB	1/98	SHEET	1 of 1


GNDF
TO GND PLANE ONLY
J2
8x2 SKT
J4
8x2 SKT
J3
8x2 SKT
J1
5x2 SKT
RECALL
PRESET
MODIFY
PRO-
CESSING
SYSTEM
SETUP
PREVIOUS
NEXT
+5V
GATE
SOFT 2
SOFT 4
SOFT 3
SOFT
PREVIOUS
NEXT
ESCAPE
RECALL
PRESET
MODIFY
PRO-
CESSING
SYSTEM
SETUP
CONTRAST
orban
MODEL 6200S
APPROVALS DATE
DWN FC 1/98
CKD CB 1/98
APPD CB 1/98
TITLE
SCHEMATIC
DISPLAY BOARD, FRONT
SIZE D
DWG NO 62150
VER 000
REV 01
SHEET 1 of 1


orban
MODEL 6200S
APPROVALS DATE
DWN SB 1/98
CKD CB 1/98
APPD CB 1/98
TITLE PCA DISPLAY, BACK
6200S
SIZE D
DWG NO 32155
VER 000
REV 01
SHEET 1 of 1
MT8
P1
R20
R21
R22
R24
R23
R27
R28
R25
R26
MT7
IC8
C8
IC3
MT9
R17
R19
R18
JP203
MT10
P3
IC6
C3
IC9
C4
IC2
C5
IC1
C6
IC4
C7
IC5
C1
RP2
RP1
P2
R29
R30
IC7
C2
MT12
P4
JP202
SOEARTH
S12
MT11
MT6


P4-1 and P4-2 provide GND to gnd plane on front bd.
There is no other connection to GND on the front bd to the plane.
MAIN BOARD
FROM POWER SUPPLY BOARD
SPARES
S12 ROTARY ENCODER
SOEARTH
MNT HOLE
orban
MODEL 6200S
SCHEMATIC
DISPLAY BOARD, BACK
DWN FC 1/98
CKD CB 1/98
APPD CB 1/98
SIZE D
DWG NO 62155
VER 000
REV 01
SHEET 1 of 1


15212.000.01
42009.040.01
42009.060.01
24308.901.01
24307.901.01
27060.000.01
26146.000.01
26143.000.01
22500.271.01
(x2)
23606.201.01
27421.
004.01
22101.000.01
23222
.101.01
20021.920.01
(x4)
21263.
710.01
(x4)
20140.120.01
(x8)
120Ω
2.0Ω
12K
21206.
747.01
21255.000.01
6800μF 16v
21256.
000.01
28112.001.01
28004.150.01
27493.000.01
22015.000.01
(x3)
28012.000.01
– OR –
28014.000.01
Hearsink
50281.000.01
24303.901.01
27421
.010.01
24304.901.01
24321.000.01
27406.
004.01
orban
MODEL 6200
APPROVALS DATE
DWN SB 1/98
CKD SB 1/98
APPD SB 1/98
TITLE ASSEMBLY DRAWING
POWER SUPPLY 6200
SIZE D
DWG NO 30245
VER 000
REV 03
SHEET 1 of 1


orban
MODEL 6200
APPROVALS DATE
DWN SB 1/98
CKD SB 1/98
APPD SB 1/98
TITLE
SCHEMATIC
POWER SUPPLY 6200
SIZE D
DWG NO 62045
VER 000
REV 03
SHEET 1 of 1

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
AT	"advanced technology" — IBM PC with 80286 or higher processor
BAL	balance
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. 0dBm = 1mW applied to a specified load. In audio, the load is usually 600Ω.
dBu	decibel voltage measurement. 0dBu = 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
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
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
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 23kHz (monophonic) or 57kHz (stereophonic)
S/P-DIF	Sony/Philips digital interface
TRS	tip-ring-sleeve (2-circuit phone jack)
THD	total harmonic distortion
TX	transmitter
μs	microseconds
VCA	voltage-controlled amplifier
VU	volume unit (meter)
XLR	a common style of 3-conductor audio connector
XTAL	crystal