Operating Manual

OPTIMOD-FM 2200

Digital Audio Processor

Models 2200 and 2200-D

orban ®

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

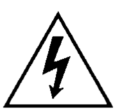
Model Number:	Description:
2200-D/U	OPTIMOD-FM 2200 DIGITAL, Stereo Encoder, Digital I/O, Protection Structure, Two-Band Structure, set to 115V (for 98-130V operation), switchable to 50µs or 75µs.
2200-D/E	OPTIMOD-FM 2200 DIGITAL, Stereo Encoder, Digital I/O, Protection Structure, Two-Band Structure, set to 230V (for 196-264V operation), switchable to 50µs or 75µs.
2200-D/J	OPTIMOD-FM 2200 DIGITAL, Stereo Encoder, Digital I/O, Protection Structure, Two-Band Structure, set to 100V (for 89-120V operation), switchable to 50µs or 75µs.
2200/U	OPTIMOD-FM 2200 DIGITAL, Stereo Encoder, Protection Structure, Two-Band Structure, set to 115V (for 98-130V operation), switchable to 50µs or 75µs.
2200/E	OPTIMOD-FM 2200 DIGITAL, Stereo Encoder, Protection Structure, Two-Band Structure, set to 230V (for 196-264V operation), switchable to 50µs or 75µs.
2200/J	OPTIMOD-FM 2200 DIGITAL, Stereo Encoder, Protection Structure, Two-Band Structure, set to 100V (for 89-120V operation), switchable to 50µs or 75µs.

MANUAL:

Part Number:	Description:
96079-000-04	2200/2200-D 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.

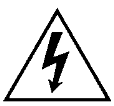
The Appliance should be used only with a cart or stand that is recommended by the manufacturer.

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 cäble. Si un cäble est endommagé, le remplacer immédiatement.

Ne jamais exposer l'appareil ou le cäble 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:

Tighten all screws on any barrier strip(s) so the screws do not fall out from vibration.
Wrap the unit in its original plastic bag to avoid abrading the paint.
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 Card
- The Line Cord
- All Miscellaneous Hardware (including the Rack Screws)
- The Extender Card
- The COAX Connecting Cable

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 Section 5: Troubleshooting.

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.

Condensed Setup of OPTIMOD-FM Models 2200 and 2200-D

This setup guide is for qualified personnel only. Use this guide to help you get your 2200 installed and operating as quickly as possible. If, at any time, you require more details to complete installation, refer to the relevant steps in Section 2 of the 2200 Operating Manual.

1. Unpack and inspect. (Refer to page 2-2.)

A ☐ If you note obvious physical damage, contact the carrier immediately to make a damage claim. A complete list of items included in the 2200 package is on page 2-2.

B ☐ *Save all packing materials*, in case you should ever have to ship the 2200.

C ☐ Complete the Registration Card and return it to Orban.

2. Change standard factory configuration, if required. (Refer to page 2-3.)

[Skip this step if your installation does not have any special requirements.]

The 2200 is supplied from the factory with its jumpers set to the configuration correct for most installations.

Stereo Encoder Composite Output Impedance	0Ω
Input Impedance	10kΩ
Input Sensitivity	–10dBu or greater (+5dBu to +27dBu peak)

A ☐ If you are changing any jumper settings, remove the top cover of the 2200 to access the main circuit board. (Make sure power is not connected.)

B ☐ Refer to Figure 2-1 in the manual to find the jumpers on the main circuit board and to position them according to your application.

C ☐ Replace the 2200 top cover.

3. Check the line voltage, fuse and power cord. (Refer to page 2-7.)

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

B ☐ Check the voltage selector on the rear panel, and change the setting if it is incorrect.

Refer to the unit's rear panel for your Model Number and the inside of the front cover of the 2200 manual for your Model Number's line voltage setting.

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

For safety, the fuse must be Slow-Blow $\frac{1}{2}$-amp for 115V, or 250mA ($\frac{1}{4}$-amp) "T" type for 230V.

D ☐ Check power cord.

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

4. Set Ground Lift switch. (Refer to page 2-8.)

The GND LIFT switch, located on the rear panel, is shipped from the factory set to GND (to connect the 2200's circuit ground to its chassis ground). If you are using the 2200's stereo encoder, and are driving its composite output into an *unbalanced* exciter input, set the GND LIFT switch to LIFT. If you are not using the 2200's stereo encoder, or are using it to drive a balanced exciter input, set the GND LIFT switch to GND.

5. Mount the 2200 in a rack. (Refer to page 2-8.)

The 2200 requires one standard rack unit (1¾ inches/4.4 cm). There should be a good ground connection between the rack and the 2200 chassis — check this with an ohmmeter to verify that the resistance is less than 0.5Ω.

6. Connect remote control (optional). (Refer to page 2-2.)

7. Connect inputs and outputs. (Refer to page 2-9.)

See the hook-up and grounding information in the manual.

8. Power up the 2200. (refer to page 2-14.)

A ☐ Plug in the 2200's power cord.

With no input program material, the red Gated LED and one of the green Function LEDs should be on. The AGC meter should indicate 10.0. The main screen appears in the front window display.

9. Recall Classical Protect preset. (Refer to page 2-17.)

A ☐ Press Recall Preset button, turn control knob until next: is CLASSICAL PROTECT, then press the RECALL NEXT soft key button.

10. Set pre-emphasis to the standard used in your country. (Refer to page 2-18.)

A ☐ Press System Setup button, press STEREO ENCODER soft key button, then set processing (PROC PRE-E) to your country's standards:

75µs	NORTH, CENTRAL, SOUTH AMERICA
50µs	EUROPE, ASIA, AFRICA, PACIFICA
	EXCEPT
75µs	TAIWAN, KOREA, THAILAND

11. Set Analog Output pre-emphasis. (Refer to page 2-18.)

[Skip this step if you are not using the 2200's analog outputs.]

A ☐ Access ANLG OUTP CALIB control screen: Press System Setup button, press I/O CALIB soft key button, then press ANLG OUTP CALIB soft key button.

B ☐ Set Analog Output pre-emphasis to [flat] or [pre-emph].

This controls whether the analog left/right outputs produce a flat signal, or a pre-emphasized signal, following the pre-emphasis set with Stereo Encoder PROC PRE-E control in the previous step.

12. Model 2200-D only: Set Digital Output pre-emphasis status. (Refer to page 2-19.)

[Skip this step if you are not using the 2200-D's digital output.]

A ☐ Access I/O CALIB control screen: Press System Setup button, press the I/O CALIB soft key button, then press DIG OUTP CALIB soft key button..

B ☐ Set DO PRE-E (Digital Output pre-emphasis status).

See page 2-19 for an explanation of the the following options:
[flat], [pre-emph], [J.17], or [J.17+pre-e].

13. Model 2200-D only: Enable Analog Inputs. (Refer to page 2-20.)

[Skip this step if you are not using the 2200-D's analog inputs.]

A ☐ Access ANLG INP CALIB control screen: Press System Setup button, press the I/O CALIB soft key button, then press ANLG INP CALIB soft key button.

B ☐ Enable Analog Inputs.

14. Adjust analog left/right input peak clipping level. (Refer to page 2-20.)

[Skip this step if you are not using the 2200's analog inputs.]

A ☐ Press the meter button so that the L/R Channel Input meters are active.

B ☐ Access ANLG INP CALIB control screen: Press System Setup button, press I/O CALIB soft key button, then press ANLG INP CALIB.

C ☐ Set Analog Input Clip level.

This step calibrates the level at which the 2200's A-D (Analog-to-Digital) converter clips to the absolute maximum peak level that your installation supplies to the 2200's analog input.

This setup maximizes the 2200's signal-to-noise ratio. If the clip level is set too low, the 2200'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.

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 2200's AI CLIP so that the program peaks just reach to –3dB on the L/R Channel Input meters.

Hold down the button directly below the words "AI CLIP," turn the control knob to scroll from +5.0dBu to +27.0dBu (or –17.0dBu to +5dBu, if input sensitivity jumpers were reset), then release the button.

Observe the L/R Channel Input meters on the 2200 for a long period of time; be sure to observe live announcer voice. If this setting is mis-adjusted, distortion will result.

0dB indicates input clipping on the 2200. These meters should never peak as high as 0dB with program material.

c) If you are using an Orban 4000A Transmission Limiter or Orban 8200ST OPTIMOD ahead of the 2200, activate the tone oscillator on either unit. Then adjust the 2200's AI PEAK so that the 2200's L/R Channel Input meters reads –3dB.

15. Calibrate analog inputs to your standard studio level. (Refer to page 2-22.)

[Skip this step if you are not using the 2200's analog inputs.]

A ☐ Press the meter button so that the L/R Channel Input meters are active.

B ☐ Access ANLG INP CALIB control screen: Press System Setup button, press I/O CALIB soft key button, then press ANLG INP CALIB.

C ☐ Set Analog Input Reference level.

This step calibrates the 2200 to the level to which your studio operators peak their program material on the studio meters. This assures that the 2200's processing presets will operate in their preferred range.

If you are able to interrupt or distort programming, use a standard reference/line-up level tone from your studio or play program material; this will achieve the most precise calibration. Adjust the appropriate 2200 reference level control (either AI REF VU or AI REF PPM) for an average of –10dB on the Master Gain Reduction meter when audio is peaking at normal levels (e.g., 0VU).

If you cannot interrupt or distort programming, calibrate by numbers, adjusting the appropriate 2200 reference level control (either AI REF VU or AI REF PPM) to your studio's reference level.

16. Model 2200-D only: Enable Digital Input. (Refer to page 2-23.)

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

A ☐ Access DIG INP CALIB control screen: Press System Setup button, press the I/O CALIB soft key button, then press DIG INP CALIB.

Note: If DIG STAT is no lock, then the AES/EBU digital input is not valid. Check connections, cabling, and digital source.

B ☐ Enable Digital Input.

17. Calibrate Digital Input to your standard studio level. (Refer to page 2-22.)

[Skip this step if you do not have Model 2200-D or if you are not using the 2200-D's digital input.]

A ☐ Press System Setup button, press the I/O CALIB soft key button, then press DIG INP CALIB.

B ☐ Set Digital Input Reference level.

This step calibrates the 2200 to the level to which your studio operators peak their program material on the studio meters. This assures that the 2200's processing presets will operate in their preferred range.

If you are able to interrupt or distort programming, play program material; this will achieve the most precise calibration. Then adjust the appropriate 2200 reference level control (either DI REF VU or DI REF PPM) for an average of −10dB on the AGC Gain Reduction meter when audio is peaking at normal levels (e.g., 0VU).

If you cannot interrupt or distort programming, calibrate by numbers, adjusting the appropriate 2200 reference level control (either DI REF VU or DI REF PPM) to your studio's reference level. Note that the numbers you see represent dB below digital full-scale.

18. Adjust Composite Output level controls. (Refer to page 2-25.)

[Skip this step if you are not using the 2200's composite outputs. These are the outputs of the 2200's stereo encoder.]

A ☐ Feed the 2200 with program material or activate the built-in 400Hz TEST tone.

To turn on the TEST tone: Press System Setup button, press TEST soft key button, set TONE to 400Hz (hold down the TONE soft key button, turn the control knob to 400 Hz, then release the button), and activate 400Hz test tone (hold down the MODE soft key button, turn the control knob to scroll to tone, then release the button).

To turn off TEST tone, hold down the MODE tone soft key button, turn the control knob to scroll to operate, then release the button.

B ☐ Adjust the 2200's Comp 1 and Comp 2 level controls — screwdriver slots on the left side of the front panel — for 100% Total Peak Modulation of your FM exciter, as indicated on a modulation monitor, or modulation indicator on your exciter.

In the U.S., you can modulate higher than 100% when using SCAs. Refer to the appropriate FCC rules.

19. Adjust Analog Left/Right or Digital Output level controls. (Refer to page 2-25.)

[Skip this step if you are not using the analog Left/Right or Digital Outputs.]

A ☐ Access analog or digital output level control: Press System Setup button, then press ANLG OUTP CALIB or DIG OUTP CALIB as required.

B ☐ Set Output level.

Hold down AO 100% or DO 100%, as applicable, and adjust the knob.

Adjust the output level controls for 100% Total Modulation of your FM exciter, or discrete left/right STL, as indicated on a modulation monitor, or modulation indicator on your exciter or STL. In the U.S., you can modulate higher than 100% when using SCAs. Refer to the appropriate FCC rules.

20. Select a preset that complements the program format of your station. (Refer to page 2-26.)

A ☐ Press Recall Preset button to access the Recall Preset screen, then select a preset.

21. Quick Setup Completed!

If you want to set up additional input/output parameters, or reset any setup adjustments, continue to “System Setup Controls,” starting on page 2-27. If you are ready to use the 2200, proceed to Section 3 for important 2200 operation information.

Operating Manual

OPTIMOD-FM 2200

Digital Audio Processor

Models 2200 and 2200-D

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WARNING

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

WARNING

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

IMPORTANT

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

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

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

The OPTIMOD-FM 2200 Digital Audio Processor is protected by U.S. patents 4,208,548; 4,249,042; 4,412,100; 4,460,871; 5,050,217; and U.K. patent 2,001,495. Other patents pending.

This manual is part number 96079-000-04

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Section 1
Introduction

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OPTIMOD-FM 2200 DIGITAL Audio Processor

Orban's OPTIMOD-FM 2200 DIGITAL Audio Processor is a complete audio processing system for FM broadcast. Model 2200 is configured for analog inputs and outputs. Model 2200-D also includes digital inputs and outputs. Features for all versions include:

- Universal transmitter protection and audio processing for FM broadcast. The 2200 can be configured to interface ideally with any commonly-found transmission system in the world.
- User-friendly Interface: A simple liquid-crystal display (LCD) makes setup, adjustment and programming easy. Front panel bargraphs show metering functions of the processing preset in use. Push one of the clearly labeled soft keys to RECALL a preset, to MODIFY PROCESSING, or to access SYSTEM SETUP controls.
- 8 factory-programmed presets, based on Orban's Two-Band and Protection/Limiting Processing Structures. These presets can be modified and saved.
- 8 user presets to store customized settings.
- Pre-emphasis limiting for the internationally-used pre-emphasis curves of 50μs and 75μs. The pre-emphasis control is almost never audibly apparent, producing a clean, open sound with subjective brightness matching the original program.
- Extremely tight peak control; overshoot is limited to ±0.3dB!
- DSP-based stereo encoder (stereo generator) produces a circuit with excellent specs, high stability, and uncompromising baseband spectrum control.
- Remote-control, via optically-isolated terminals that can be operated with contact closures (to facilitate interfacing to older-technology remote controls).
- Built-in line-up tone generator facilitating quick and accurate level setting in any system and a Bessel Null tone for calibrating modulation.

Presets in the 2200

There are two distinct kinds of presets in the 2200: Factory Processing Presets and User Presets.

The 8 Factory Processing Presets include a protection/limiting preset, a two-band general purpose preset, and 6 other presets derived from the two-band structure. All of the factory processing presets are stored in the 2200'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. The factory preset remains unchanged. You may store your new settings in one of the 8 numbered User Presets.

User Presets cannot be created from "scratch." You must always start by recalling a Factory Preset, make changes, then store the changes in a User Preset.

The Two-Band Structure

The Two-Band Structure consists of a slow single-band gated AGC (Automatic Gain Control) for gain riding, followed by a gated two-band compressor and a high-frequency limiter. A two-band equalizer is available before the input of the Two-Band structure.

The Two-Band Structure is an improved version of Orban's classic 8100A OPTIMOD-FM, but with increased high frequency clarity. (This is the same structure used in our OPTIMOD-FM 8200.) It is operated after a phase rotator (time-dispersion filter) to improve its loudness capability by making positive and negative peaks more symmetrical, particularly with voice.

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 bass coupling control — BASS COUPL — is operated toward 0%) can gently correct it without introducing obvious coloration.

Two-Band Presets

The Two-Band Factory Programming Presets are designed to help you get on the air quickly. In most cases, they will suit your needs without the need for further adjustment. Or, if you desire, you can always experiment with the LESS-MORE control to fine-tune the processing to your taste.

The basic Two-Band preset, 2B General Purpose, provides an average amount of processing. The other Two-Band presets provide a sound tailored for a specific program format. For example, Music-Light produces a very open, unprocessed sound while Music+Bass Medium provides a very punchy, clean, open sound. Presets include: 2B GENERAL PURPOSE, TALK, MUSIC-LIGHT, MUSIC-MEDIUM, MUSIC HEAVY, MUSIC+BASS MEDIUM and MUSIC+BASS HEAVY.

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. As a result, 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-FM controls the level of the music in ways that are, for all practical purposes, inaudible to the listener. Low-level passages are increased in level by up to 10dB, while the dynamics of crescendos are maintained.

The same station may wish to switch to the Protection Structure during the evening hours when the audience is more likely to listen critically.

Protection Limiting: Orban’s Approach

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 it’s audience is likely to listen in a more critical manner. While the Protection Limiter Structure can readily reduce the dynamic range, it is designed to do so without increasing program density, loudness, or the consistency of sound from different sources. It’s primary function is to protect the transmitter from over-deviation while preserving the spectral and textured quality of the source material.

The 2200 has one Protection Limiter factory preset , named CLASSICAL PROTECT.

Traditionally, protection limiters have used peak-sensing automatic gain control (AGC) processors to control peak levels. This 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. 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*.

Input/Output Configurations

The OPTIMOD-FM 2200 DIGITAL is designed to simultaneously accommodate:

- analog left/right inputs and outputs
- Digital AES/EBU left/right inputs and outputs (Model 2200-D only)
- stereo analog baseband composite output

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. Outputs can drive up to +20dBm into 600Ω.

Level control of the analog inputs and outputs is via software control through SYSTEM SETUP. (See page 2-20 and 2-6.)

Digital AES/EBU Left/Right Input/Output (Model 2200-D only)

The digital input and output follow the professional AES/EBU standard. 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.

The 2200-D is designed to simultaneously accommodate digital and analog inputs and outputs. You select whether the 2200-D uses the digital or the analog input via software control through System Setup (DIG INP CALIB or ANLG INP 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.

The 2200-D's digital I/O operates at a 32kHz, 44.1kHz, or 48kHz sample rate. Its output can be flat, pre-emphasized (to the 50µs or 75µs curve selected for the processing), J.17 pre-emphasized, or J.17 plus 50µs or 75µs pre-emphasized.

> Please note that the AES/EBU standard is not the same as the S/P-DIF (Sony/Philips Digital Interface) standard used in consumer digital applications, such as the "digital outputs" of CD players. The AES/EBU interface will not work with S/P-DIF signals.

A 2200 cannot be upgraded to a 2200-D.

Stereo Analog Baseband Composite Output

The stereo encoder has two unbalanced analog baseband outputs on two BNC connectors on the rear panel. Each output can be strapped for 0Ω or 75Ω source impedance, and can drive up to 8V peak-to-peak into 75Ω in parallel with up to 0.047µF (100ft/30m of RG-59/U cable) before any noticeable performance degradation occurs.

Level control of each output is via a separate screwdriver control accessible from the front panel.

A ground lift switch is available on the rear panel. This is useful to prevent ground loops between the 2200 and the transmitter.

Location of OPTIMOD-FM

Best Location for OPTIMOD-FM

The best location for OPTIMOD-FM is as close as possible to the transmitter, so that its stereo encoder output can be connected to the transmitter through a circuit path that introduces the least possible change in the shape of OPTIMOD-FM's carefully peak-limited waveform — a short length of coaxial cable.

We very strongly recommend that you use the 2200's internal stereo encoder and feed its output directly to the baseband input of the exciter through less than 100 feet (30 meters) of coaxial cable. You will achieve a louder sound on the air, with better control of peak modulation, than if you use an external stereo encoder.

About Transmission Levels and Metering

Meters

Studio engineers and transmission engineers consider audio levels and their measurements differently, so they typically use different methods of metering to monitor these levels.

The VU meter is an average-responding meter (measuring the approximate RMS level) with a 300ms rise time and decay time; the VU indication usually under-indicates the true peak level by 8 to 14dB.

The Peak Program Meter (PPM) indicates a level between RMS and the actual peak. The PPM has an attack time of 10ms, slow enough to cause the meter to ignore narrow peaks and under-indicate the true peak level by 5dB or more.

The absolute peak-sensing meter or LED indicator shows the true peak level. It has an instantaneous attack time, and a release time slow enough to allow the engineer to easily read the peak level.

Figure 1-1 shows the relative difference between the absolute peak level, and the indications of a VU meter and a PPM.

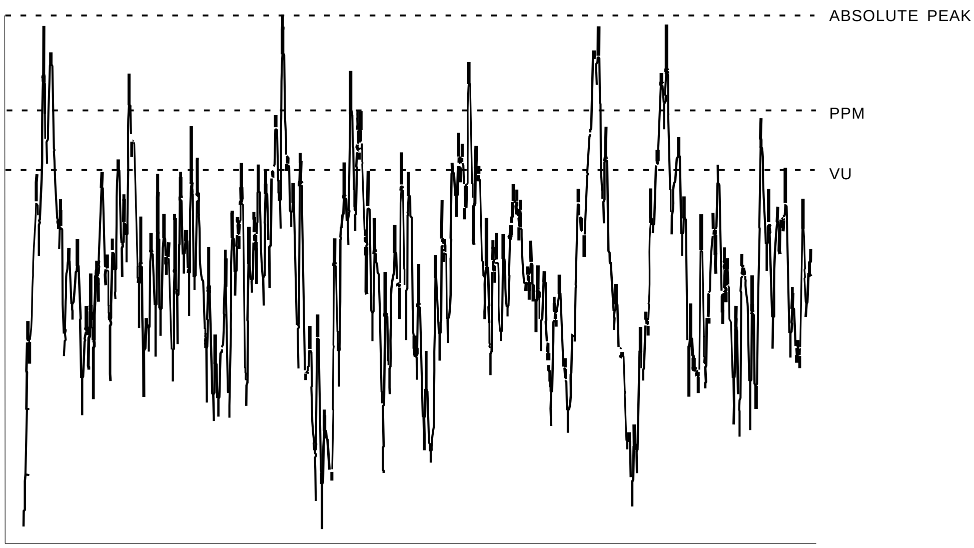


Figure 1-1: Absolute Peak Level, VU and PPM Reading
For a Few Seconds of Music Program

Studio Line-up Levels and Headroom

The studio engineer is primarily concerned with calibrating the equipment to provide the required input level for proper operation of each device, and so that all devices operate with the same input and output levels. This facilitates patching devices in and out without recalibration. For line-up, the studio engineer uses a calibration tone at a studio standard level, commonly called line-up level, reference level, or operating level.

Metering at the studio is by a VU meter or PPM (Peak Program Meter). As discussed above, the VU or PPM indication lags the true peak level. Most modern studio audio devices have a clipping level of no less than +21dBu, and often +24dBu or more. So the studio standardizes on a maximum program indication on the meter that is lower than the clipping level, so that peaks that the meter doesn't indicate will not be clipped. Line-up level is usually at this same maximum meter indication.

In facilities that use VU meters, this level is usually at 0VU, which corresponds to the studio standard level, typically +4 or +8dBu. For facilities using +4dBu standard level, instantaneous peaks can reach +18dBu or higher (particularly if the operator overdrives the console or desk). Older facilities with +8dBu standard level and equipment that clips at +18 or +21dBu will experience noticeable clipping on some program material.

In facilities that use the BBC-standard PPM, maximum program level is usually PPM4 for music, PPM6 for speech. Line-up level is usually PPM4, which corresponds to +4dBu. Instantaneous peaks will reach +17dBu or more on voice.

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

Transmission Levels

The transmission engineer is primarily concerned with the peak level of a program to prevent overloading or over-modulation of the transmission system. This peak overload level is defined differently, system to system. In FM modulation (FM/VHF radio and television broadcast, microwave or analog satellite links), it is the maximum-permitted RF carrier frequency deviation. In AM modulation, it is negative carrier pinch-off. In analog telephone/post/PTT transmission, it is the level above which serious crosstalk into other channels occurs, or the level at which the amplifiers in the channel overload. In digital, it is the largest possible digital word.

For metering, the transmission engineer uses an oscilloscope, absolute peak-sensing meter, calibrated peak-sensing LED indicator, or a modulation meter. A modulation meter usually has two components — a semi-peak reading meter (like a PPM), and a peak-indicating light which is calibrated to turn on whenever the instantaneous peak modulation exceeds the over-modulation threshold.

Line-Up Facilities

Metering of Levels

The 2200's front panel metering includes bargraphs displaying the following I/O levels: L/R Input, L/R Output and Composite Output.

L/R Input Level

Left and right input level is shown on a VU-type scale (0dB to –27dB), while the metering indicates *absolute instantaneous peak* (much faster than a standard PPM or VU meter). 0dB indicates A/D converter clipping (analog input) or digital full-scale (digital input, 2200-D only).

Left/Right Output Level

Left and right output level is shown on a VU-type scale (0dB to –27dB), where the metering indicates *absolute instantaneous peak* (much faster than a standard PPM or VU meter).

The meter is scaled so that 0dB is calibrated to the highest left and right peak modulation level, before de-emphasis, that the processing will produce, under any program, processing, or setup condition (except when the processing is switched to bypass). The meter indication is not affected by the setting of the analog or digital output level control.

Composite Output Level

The Orban 2200 Audio Processor controls instantaneous, absolute peak levels to a tolerance of approximately ±0.3dB. Composite modulation is indicated in % modulation, absolute instantaneous peak indicating. 100% is calibrated to the highest composite peak modulation level that the processing will produce, including the pilot tone, under any program, processing, or setup condition (except when the processing is switched to bypass). 100% ordinarily corresponds to ±75kHz carrier deviation.

Built-in Calibrated Line-up Tones

To facilitate matching the output level of the 2200 to the transmission system that it is driving, the 2200 contains an adjustable tone that produces sine waves at 2200's analog, digital and composite outputs. The frequency of the line-up tones can be adjusted from the front panel. The modulation is always 100%.

When the 2200's left/right analog output is switched to flat, a de-emphasis filter is inserted between the output of the 2200's audio processing and its line output. Thus, as the frequency of the tone preset is changed, the level at the 2200's line output will follow the selected de-emphasis curve.

In most cases the pre-emphasis filter in the driven equipment will undo the effect of the 2200's internal de-emphasis, and the 2200's output level should be adjusted so that the tone produces 100% modulation of the transmission link as measured after the link's pre-emphasis filter. At 100Hz, switching the de-emphasis out or in will have negligible effect on the level appearing at the 2200's left and right audio outputs.

Built-in Calibrated Bypass

Bypass is available to transparently pass line-up tones generated earlier in the system. It will also pass program material, with no gain reduction or protection against overmodulation. It can transparently pass any line-up tone applied to its input up to about 130% output modulation, at which point clipping may occur. Access bypass in the System Setup TEST MODE screen.

Note that BYPASS applies 50µs or 75µs pre-emphasis, as determined by the setting of PROC PRE-E, in the System Setup STEREO ENCODER screen.

Warranty, Feedback

Warranty

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 on page 5-9.

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

page contents

CAUTION

The installation and servicing instructions in this manual are for use by qualified personnel only. To avoid electric shock, do not perform any servicing other than that contained in the Operating Instructions unless you are qualified to do so. Refer all servicing to qualified service personnel.

Installation of 2200

Allow about 2 hours for installation.

Installation consists of: (1) unpacking and inspecting the 2200, (2) optional resetting of jumpers for 2200 options (composite output impedance, input termination, input sensitivity), (3) checking the line voltage setting, fuses and power cord, (4) mounting the 2200 in a rack, (5) connecting inputs, outputs and power, (6) setting the GROUND LIFT switch, (7) and optional connecting of remote control leads.

When you have finished installing the 2200, proceed to “System Setup,” on page 2-16.

1. Unpack and inspect.

A ☐ If you note obvious physical damage, contact the carrier immediately to make a damage claim. Packed with the 2200 are:

1	Operating Manual
1	Quick Setup Guide
1	Line Cord
2	½A Replacement Fuses for “U” and “J” Versions
2	250mA Replacement Fuses for “E” Version
1	Orban green screwdriver (Xcelite R3323)
1	Booklet: *Audio Quality in the FM Plant*

B ☐ *Save all packing materials*! If you should ever have to ship the 2200 (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, 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 or give away our customer’s names to anyone.

2. Change standard factory configuration, if required.

[Skip this step if your installation does not have any special requirements.]

The 2200 is supplied from the factory with its jumpers set to the configuration correct for most installations.

Stereo Encoder Composite Output Impedance	0Ω
Input Impedance	10kΩ
Input Sensitivity	–10dBu or greater (+5dBu to +27dBu peak)

A ☐ To change any jumpers you must remove the top cover of the 2200 to access the main circuit board. (Make sure power is not connected.) Remove all screws holding the cover in place, then lift it off. (Be careful not to strip threads when opening the cover.)

B ☐ Refer to Figure 2-1 to find the jumpers on the main circuit board and to position them according to your application.

The following information is provided to explain each jumper and its settings in detail.

- Stereo encoder composite output impedance.

  [Do not change the default 0Ω jumper setting unless your installation needs 75Ω source impedance.]

  The stereo encoder is shipped from the factory with 0Ω source impedance. This is correct for virtually all installations. However, the 2200 stereo encoder can be changed to 75Ω source impedance if desired.

  The frequencies in the stereo baseband are low by comparison to RF or video, and the characteristic impedance of coaxial cable is not 75Ω at lower frequencies, so the transmission system will have more accurate amplitude and phase response (and thus, better stereo separation) if the cable is driven by a very low source impedance (0Ω) and is terminated with greater than 1kΩ at the exciter.

  However, a few broadcast organizations require that FM composite be transmitted in impedance-matched coaxial cable with 75Ω source and load impedances.

  To change the source impedance of one or both of the composite outputs:

  To change the source impedance of composite output #1, move jumper JA to the “75Ω” position (Fig 2-1). To change the source impedance of composite output #2, move jumper JB to the “75Ω” position (Fig 2-1).

- Analog left/right input termination.

  [Do not change the default setting unless your installation requires 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 2200 analog inputs can be changed to 600Ω input impedance.

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

Move jumpers J301 and J305 according to Figure 2-1. Jumper J301 sets the left channel and jumper J305 sets the right channel.

- Set analog left/right input sensitivity.

[Skip this step if your installation will supply the 2200 with nominal input level of –10dBu or greater (+5dBu to +27dBu peak).]

The analog left/right inputs are shipped from the factory with input sensitivity to accommodate inputs whose absolute maximum peak level is between +5dBu and +27dBu.

If VU meters are used, +5dBu to +27dBu absolute peak corresponds to a 0VU level of approximately –9dBu to +13dBu.

If PPMs are used, +5dBu to +27dBu absolute peak corresponds to a PPM level of approximately –2dBu to +20dBu.

However, in unusual circumstances where the input level is very low, the 2200 analog inputs can be changed for greater sensitivity. This usually occurs only when the studio-to-transmitter link is a long telephone or post line with a passive equalizer at the receive end and no amplifier to make up the loss of the line and the equalizer.

To increase the input sensitivity of the analog input to accommodate absolute peak levels of –17dBu to +5dBu (nominal levels down to –30dBu):

Move jumpers J302, J303, J306, J307 and J308 according to Figure 2-1. Jumpers J302 and J303 set the left channel and jumpers J306 and J307 set the right channel. Jumper J308 sets the control circuit to recognize the new input sensitivity.

C ☐ Replace the 2200 top cover.

Replace all screws snugly. (Be careful not to strip threads by fastening the screws too tightly.)

2200 Rear Panel

Voltage Selector (for Model Numbers 2200/U, 2200-D/U, 2200/E and 2200-D/E) can be set to 115V (for 100-132V operation) or 230V (for 200-264V operation); (for Model 2200/J and 2200-D/J) set to 115V (for 89-120V operation).

Fuse values can be changed to support 115V or 230V operation. Fuse must be 3AG Slow-Blow, ½-amp for 115V, or ¼-amp (250mA) "T" type for 230V.

Power Cord is detachable and is terminated in a "U-ground" plug (USA standard), or CEE7/7 plug (Continental Europe), as appropriate to your 2200's Model Number.

GND LIFT (Ground Lift) Switch can be set to GND (to connect the 2200's circuit ground to its chassis), or to LIFT (if you are using the 2200's stereo encoder, and are driving its composite output into an *unbalanced* exciter input).

Remote Control Interface is provided to connect the 2200 to a remote control. The 2200 remote control accepts a DB-25 connector and supports user-programmable selection of up to eight inputs for any one of the following parameters: user presets, factory presets, bypass, tone, exit test, stereo, mono left, mono right, mono sum, analog input, digital input (Model 2200-D only), digital input + J.17 pre-emphasis (Model 2200-D only).

> A valid signal is a momentary transition from no-current to current flowing through the particular remote signal pins. Current must flow for at least 50ms for the signal to be interpreted as valid. It is acceptable to apply current continuously to an input, DC or AC. Do not exceed 9 volts unless you use an external current-limiting resistor that limits current to 10mA.

COMPOSITE 1 OUTPUT and COMPOSITE 2 OUTPUT are provided, each with independent output level control (via front panel Comp 1 and Comp 2 controls). Each output uses a BNC connector.

ANALOG INPUT and ANALOG OUTPUT provided to support left and right audio signals through XLR-type connectors.

Digital AES/EBU INPUT and AES/EBU OUTPUT (Model 2200-D only) are provided to support two-channel AES/EBU-standard digital audio signals through XLR-type connectors.

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.

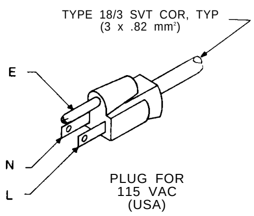
Refer to the unit's rear panel for your Model Number and the inside of the front cover of this manual for your Model Number's line voltage setting.

Model Numbers 2200/U, 2200-D/U, 2200/E and 2200-D/E are shipped configured for either 100-132V or 200-264V, 50Hz or 60Hz operation, as indicated on the rear panel. To change the operating voltage, set the voltage selector to 115V (for 100-132V) or 230V (for 200-264V) as appropriate.

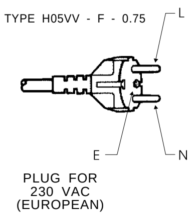
Model Number 2200/J and 2200-D/J are shipped for 89-120V, 50/60Hz operation. The voltage selector should be set to 115V (for 89-120V).

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

For safety, the fuse must be Slow-Blow ½-amp for 115V, or 250mA (¼-amp) "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 2-1: AC Line Cord Wire Standard

D ☐ Check power cord.

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

The power cord is terminated in a "U-ground" plug (USA standard), or CEE7/7 plug (Continental Europe), as appropriate to your 2200's Model Number. The green/yellow wire is connected directly to the 2200 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 switch is located on the rear panel.

The GND LIFT switch is shipped from the factory set to ground (to connect the 2200's circuit ground to its chassis ground). If you are using the 2200's stereo encoder, and are driving its composite output into an *unbalanced* exciter input, set the GND LIFT switch to LIFT.

This will break a ground loop that could otherwise occur.

Unbalanced exciter inputs can cause hum and noise because it is difficult to control the system grounding. If hum or noise appears that cannot be cured by resetting the GND LIFT switch, we suggest that you install the optional Orban CIT25 Composite Isolation Transformer at the exciter's input to balance it. If you use the CIT25, set the 2200's GND LIFT switch to GND.

If you are not using the 2200's stereo encoder, set the GND LIFT switch to ground.

5. Mount the 2200 in a rack.

The 2200 requires one standard rack unit (1¾ inches/4.4 cm).

There should be a good ground connection between the rack and the 2200 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 113°F/45°C when equipment is powered.

Equipment life will be extended if the unit is mounted away from sources of vibration, such as large blowers.

The shorter the baseband cable run from the 2200 to exciter, the less likely that ground loops or other noise problems will occur in the installation. If you require a long cable run, it is usually best to mount the RF exciter close to the 2200, and to make the long cable carry the RF output from the exciter to the transmitter's RF power amplifiers.

6. Connect remote control. (optional)

The 2200 has extensive remote control provisions, which are described on page 2-34.

Optically-isolated remote control connections are terminated in a type DB-25 male connector located on the rear panel. It is wired according to Fig. 2-2. To select the desired function, apply a 5-12V AC or DC pulse between the appropriate REMOTE 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 3.6K 10%, 1-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 the connector shell 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+
10-13. N/C
   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

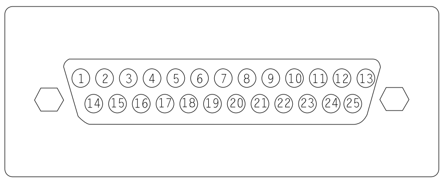


Figure 2-2: Wiring the 25-pin Remote Control Connector

7. Connect inputs and outputs.

See the hook-up and grounding information on the following pages.

Audio Input and Audio Output Connections	Page 2-10
Composite Output	Page 2-12
AES/EBU Digital Input and Output (2200-D only)	Page 2-11
Grounding	Page 2-12

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 –30dBu and +8dBu will result in normal operation of the 2200. (See step 2 on page 2-3 for a full discussion).

  (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 control. It is adjustable from –17dBu to +27dBu in two ranges.

- 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. (Note: International Standard is pin 2 HIGH .)

- *In **low RF fields** (like a studio site), do not connect the cable shield at the 2200 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 2200 input.

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

Analog Audio Output

- **Electronically-balanced and floating output**s simulate a true transformer output. The source impedance is 30Ω. The output is capable of driving loads of 600Ω or higher; maximum output level is +20dBu. 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 2200'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 (2200-D Only)

Model 2200-D includes an AES/EBU digital input/output connector. These follow the AES/EBU standard.

- Use 110Ω shielded twisted-pair cable (Belden 1800A, or equivalent). The length of cable, cable characteristics, and quality of termination become extremely important because AES/EBU signals have a spectrum with significant energy to 6MHz. Do not use standard audio cable for AES/EBU applications.

  Impedance mismatching and noise are the most common causes of trouble with AES/EBU signals, usually manifesting as pops or clicks in the output audio, or total loss of synchronization lock. Impedance mismatching can be caused by use of an inappropriate cable type, poor cable termination technique, or improper termination of the transmission line either in the source, or receiver equipment. Induced noise may prevent the AES/EBU receiver from locking to the incoming data stream. For this reason, cable length is generally recommended to be less than 100 meters (approx. 328').

  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 (0dBFS). Since the input meters monitor clip levels, the maximum digital input will make the meters display 0dB. The reference level is adjustable using either the DI REF VU or DI REF PPM controls. The reference level determines the amount of gain reduction. The output level is adjustable using the DO 100% control. This control is in dB, referenced to full scale, and is adjustable from 0.0dBFS to –20dBFS.

Composite Output

- There are two independent composite baseband outputs (containing the encoded stereo signal and the stereo pilot tone).

  Each output has an independent output level control and can be strapped for 0Ω or 75Ω source impedance. Each output can drive up to 8V peak-to-peak into 75Ω in parallel with up to 0.047µF in cable and input capacitance before any noticeable performance degradation occurs.

- Connect the 2200's composite output to the exciter input with up to 100 feet (30.5m) of RG-58/U or RG-59/U coaxial cable terminated with BNC connectors.

  Longer runs of coax may increase problems with noise, hum, and RF pickup at the exciter. In general, the least troublesome installations place the 2200 close to the exciter and limit the length of the composite cable to less than 6 feet (1.8m).

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

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

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

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 2200 has balanced inputs.

(Grounding continued)

- All equipment *circuit grounds* must be connected to each other; all equipment *chassis grounds* must be connected together.
- In a low RF field, *audio 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 2200 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 2200's GND LIFT switch to GND, *except* when the 2200's optional stereo encoder composite output is driving an **unbalanced exciter input**. This is an unbalanced-to-unbalanced connection, so the ground loop that would otherwise occur must be broken by setting the 2200's GROUND LIFT switch to LIFT.

  Alternately, you can balance and float the exciter input with the Orban CIT25 Composite Isolation Transformer — (call Orban Customer Service).

- *In high RF fields*, the system is usually grounded through the equipment rack in which the 2200 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.

Older Exciters

Most exciters have straightforward wideband inputs, and no special considerations are involved. If you have an older exciter requiring special wideband interfaces, contact Orban Customer Service.

8. Power up the 2200.

A ☐ Plug in the 2200's power cord.

With no input program material, the red Gated LED and one of the green Function LEDs should be on. The AGC meter should indicate 10.0.

The main screen appears in the front window display.

```
on air:CLASSICAL PROTECT
AnlgIn—Stereo
```

The main or "home" screen shows which processing preset is selected to be on-air, the type of input (analog or digital), and the stereo encoder mode.

If the main screen does not appear, repeatedly press the Escape button until it does appear.

9. Physical installation is complete.

A ☐ Continue with system software configuration, on page 2-16.

We recommend you browse through the explanation of 2200 front panel controls and meters (on the following pages) before you being configuring system software.

2200 Front Panel

Screen Display labels the four soft key buttons and provides control setting information.

Screen Saver feature: The screen has a built-in screen saver that turns off the backlight after approximately one hour. The screen turns back on when any front panel control is touched. Note that buttons don't perform their normal function when the screen is blank. Similarly, the control knob's first turn is not read, until it stops for a second or so.

Contrast Button adjusts the optimum viewing angle of the screen display. Press this button to cycle through four contrast settings for the screen display.

Four Soft Key Buttons provide access to all 2200 functions and controls. The functions of the buttons change with each screen according to the labels at the bottom of each screen. Push a button:

To select options (always identified on the screen by all-capital-letter words surrounded by left and right vertical bars), press the button directly below the option.

To change a parameter setting (always identified by lower-case letters or numerals), hold down the button directly below the parameter setting, turn the control knob to scroll through choices, and release the button to set the parameter.

Control Knob is used for changing data in one of three methods.

To scroll through submenu choices: Presets (on Recall Preset screen), FULL CONTROL parameters (on Modify Processing FULL CONTROL screen) and 8 Remote Interfaces (on System Setup REMOTE INTERFACE screen).

To change a parameter setting, by ***simultaneously holding down a soft key***. The parameters being changed take effect immediately, except for the following system level controls: MODE (on test screen), MODE (on Stereo Encoder screen), INPUT, AO PRE-E, DO PRE-E, DO RATE, DO SYNC and XTLK TEST. The setting for these controls do not take effect until the soft key is released.

To adjust the LESS-MORE control. Adjusting this control changes the sound immediately.

Escape Button returns the user to the previous screen; pressing this button repeatedly will always return you to main screen, which shows the on-air preset name.

Recall Preset Button brings up a screen that displays the current on-air preset and next preset (which can be changed by turning the control knob). To put a different preset on-air, turn the control knob to find the preset desired, then press the RECALL NEXT soft key.

When the button's yellow LED is lit, the Recall Preset screen is displayed.

Modify Processing Button brings up a screen to modify parameters for the current on-air preset. For Two-Band presets: LESS-MORE, EQ and FULL CONTROL. For the Protection preset, DRIVE and 30HZHPF.

When the button's yellow LED is lit, the Modify Processing screen (or one of its submenus) is displayed.

System Setup Button brings up a screen to modify system settings (such as I/O levels). There are four System Setup submenus: I/O CALIB, STEREO ENCODER, REMOTE INTERFACE, TEST.

When the button's yellow LED is lit, the System Setup screen (or one of its submenus) is displayed.

Gain Reduction Meters indicate AGC, Bass, and Master Gain Reduction. All three meters operate over a –25.0dB to 2.5dB range. Note that the AGC and Bass meters are off when the Protect preset is on-air.

HF Limiting LEDs light when the high-frequency content of audio is being limited by the very fast high-frequency limiters. These LEDs indicate when greater than 0.5dB HF limiting is occurring.

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

Composite Meter is a 10-segment bargraph showing the stereo encoder's composite output level before the composite level controls.

Function Button selects which of three functions are displayed in the Function meters: Enhance, L/R Input or L/R Output.

Function LEDs indicate which function is currently displayed by the Function meters: Enhance, L/R Input or L/R Output. Press the Function button to toggle between the three functions.

Function Meter indicates level of Enhance, L/R Channel Input or Output, as selected with Function button. The meters operate over a –27dB to 0dB range. Input meters are referenced to clip level. Output meters are referenced to 100% modulation level. HF Enhance meter shows the active amount of enhancement activity. Since the HF Enhancement is program-dependent, it will vary with source material and the HF parameter. Note: HF Enhance is displayed only on the left-hand meter, below "HF."

Stereo Encoder Screwdriver-Adjustable Controls

Orban supplies a special green-handled flat-blade screwdriver (Xcelite R3323) to adjust the stereo encoder controls.

Comp 1 sets the output level of Composite Output 1.

Comp 2 sets the output level of Composite Output 2.

Basic System Setup

Allow about about 30 minutes for system setup.

You can set up all of the 2200's required settings through Basic System Setup. It is a guided procedure for adjusting all of the setup adjustments needed to get you on the air. System setup consists of: (1) setting the pre-emphasis, (2) setting the analog input peak level, (3) setting the input reference level, (4) adjusting composite, analog or digital output level and (5) recalling a processing preset.

The 2200 also contains a few special-feature parameters that are not included in this section. These parameters can be adjusted immediately after Basic System Setup is completed, or they can be implemented during normal operation. All the I/O CALIB, STEREO ENCODER, REMOTE INTERFACE and TEST parameters are covered in depth, beginning on page 2-27.

To complete the following steps, you need to have a basic understanding of how to use the 2200's front panel controls:

> **To change display contrast**, repeatedly press the Contrast button to adjust the display for best clarity.
>
> **To select options** (always identified on the screen by all-capital-letter words surrounded by left and right vertical bars), press the button directly below the option.
>
> **To change a parameter setting** (always identified by lower-case letters or numerals), hold down the button directly below the parameter setting, turn the control knob to scroll through choices, and release the button to set the parameter. Note that most parameters take effect immediately when you turn the control knob. Some settings (e.g., INPUT status) will not take effect until the soft key button is released.

1. Begin System Setup.

A ☐ If you have not done so already, plug in the 2200's power cord. The main screen (shown below) appears in the window display.

> If the main screen does not appear, repeatedly press ESC until it does appear.

```
on air:CLASSICAL PROTECT
AnlgIn—Stereo
```

> **Change preset to CLASSICAL PROTECT — This preset facilitates the most accurate initial setup:** If on-air: is not CLASSICAL PROTECT, press the Recall Preset button, hold down the next: soft key button and turn the control knob to scroll through preset list to CLASSICAL PROTECT, then release the button and press RECALL NEXT soft key button.
>
> Note: User can choose their preferred preset *after* setup is completed.

2. Set pre-emphasis to the standard used in your country.

A ☐ Press System Setup button to access the System Setup screen.

```
| I/O   |  STEREO | REMOTE   |  TEST  |
| CALIB | ENCODER | INTERFACE|        |
```

B ☐ From the System Setup screen, press the STEREO ENCODER soft key button.

```
|PROC PRE-E|  MODE  |PILOTLVL|XTLK TEST|
|   75   us| stereo |   9.0 %| normal  |
```

C ☐ Set processing Pre-Emphasis

[50µs] or [75µs]

This controls the pre-emphasis of the internal processing's high-frequency limiters, and the pre-emphasis of the stereo encoder's output. It does not control whether analog or digital left/right outputs are flat or pre-emphasized.

Set this control to the pre-emphasis standard in your country:

75µs	NORTH, CENTRAL, SOUTH AMERICA
50µs	EUROPE, ASIA, AFRICA, PACIFICA
	EXCEPT
75µs	TAIWAN, KOREA, THAILAND

Hold down the button directly below the words "PROC PRE-E," turn the control knob counterclockwise to choose 50µs, or clockwise to choose 75µs, then release the button.

3. Set Analog Output pre-emphasis.

[Skip this step if you are not using the 2200's analog outputs.]

A ☐ Press System Setup button to access the main System Setup screen.

B ☐ From the System Setup screen, press the I/O CALIB soft key button.

```
|ANLG INP| DIG INP | ANLG OUTP|DIG OUTP|
| CALIB  |  CALIB  |   CALIB  |  CALIB |
```

C ☐ Press ANLG OUTP CALIB soft key button.

```
| AO 100%| AO PRE-E |
|—22.1dBu|pre-emph  |
```

D ☐ Set Analog Output pre-emphasis status.

[flat] or [pre-emph]

This controls whether the analog left/right outputs produce a flat signal, or a pre-emphasized signal, following the pre-emphasis set with Stereo Encoder PROC PRE-E control in step 2 above.

Hold down the button directly below the words “AO PRE-E,” turn the control knob counterclockwise to choose flat, or clockwise to choose pre-emphasis, then release the button.

4. Set Digital Output pre-emphasis status.

[Skip this step if you are not using the 2200-D’s digital output. Digital I/O parameters only available with Model 2200-D.]

A ☐ Press System Setup button to access the main System Setup screen.

B ☐ From the System Setup screen, press the I/O CALIB soft key button.

```
|ANLG INP| DIG INP | ANLG OUTP|DIG OUTP|
| CALIB  |  CALIB  |   CALIB  |  CALIB |
```

C ☐ Press DIG OUTP CALIB soft key button.

```
| DO 100% | DO PRE-E |DO RATE | DO SYNC|
| -2,8dBFS|   flat   | 32kHz  |internal|
```

D ☐ Set DO PRE-E (Digital Output pre-emphasis status).

[flat], [pre-emph], [J.17], or [J.17+pre-e]

Within the audio processing, the audio signal is pre-emphasized to either 50µs or 75µs (as set with Stereo Encoder PROC PRE-E control in step 2 above).

Hold down the button directly below the words “DO PRE-E,” turn the control knob to scroll through the output pre-emphasis choices, then release the button to set the parameter.

[flat]: Set the 2200-D's digital output to flat if the output feeds a digital STL modem using lossy bit-rate reduction (such as Marti MD-2, Moseley DSP 6000, and TFT DMM92); these modems are ***not*** designed to carry pre-emphasized AES/EBU data.

[pre-emph]: Use the pre-emph setting when the AES/EBU digital output of the 2200-D is sent directly to the AES/EBU input of a digital exciter, such as the Harris "Digit." Also, if you are using a digital link with no data rate reduction, the 2200-D's output should remain pre-emphasized.

[J.17], or [J.17+pre-e]: Use these settings to apply J.17 pre-emphasis or J.17 plus the pre-emphasis set in step 2 above.

5. Enable Analog Inputs.

[Skip this step if you do not have Model 2200-D or if you are not using the analog inputs.]

A ☐ Press System Setup button to access the main System Setup screen.

B ☐ From the System Setup screen, press the I/O CALIB soft key button.

ANLG INP CALIB	DIG INP CALIB	ANLG OUTP CALIB	DIG OUTP CALIB

C ☐ Press ANLG INP CALIB soft key button.

INPUT	AI REF VU	AI REF PPM	AI CLIP
analog	+4.0 dBu	+9.0 dBu	+20.0dBu

D ☐ Enable Analog Inputs.

[analog], [digital] or [dig+J17]

Hold down the button directly below the word "INPUT," turn the control knob counterclockwise to choose analog, then release the button to set the parameter.

6. Adjust analog left/right input peak clipping level.

[Skip this step if you are not using the 2200's analog inputs.]

A ☐ Press the meter button so that the L/R Channel Input meters are active.

B ☐ Press System Setup button to access the System Setup screen.

```
|  I/O    |  STEREO | REMOTE  |  TEST   |
| CALIB   | ENCODER |INTERFACE|         |
```

C ☐ From the System Setup screen, press I/O CALIB soft key button.

```
|ANLG INP| DIG INP | ANLG OUTP|DIG OUTP|
| CALIB  |  CALIB  |   CALIB  |  CALIB |
```

D ☐ Press ANLG INP CALIB.

```
| INPUT  | AI REF VU|AI REF PPM| AI CLIP|
|analog  | +4.0  dBu| +9.0 dBu |+20.0dBu|
```

E ☐ Set Analog Input Clip level.

This step calibrates the level at which the 2200's A-D (Analog-to-Digital) converter clips to the absolute maximum peak level that your installation supplies to the 2200's analog input.

This setup maximizes the 2200's signal-to-noise ratio. If the clip level is set too low, the 2200'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.

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 2200's AI CLIP so that the program peaks just reach to about –3dB on the L/R Channel Input meters.

   Hold down the button directly below the words "AI CLIP," turn the control knob to scroll from +5.0dBu to +27.0dBu (or –17.0dBu to +5dBu, if input sensitivity jumpers were reset), then release the button.

   Observe the L/R Channel Input meters on the 2200 for a long period of time; be sure to observe live announcer voice. If this setting is mis-adjusted, distortion will result.

   0dB indicates input clipping on the 2200. These meters should never peak as high as 0dB with program material.

c) If you are using an Orban 4000A Transmission Limiter or Orban 8200ST OPTIMOD ahead of the 2200, activate the tone oscillator on either unit. Then adjust the 2200's AI PEAK so that the 2200's L/R Channel Input meters reads –3dB.

7. Calibrate analog inputs to your standard studio level.

[Skip this step if you are not using the 2200's analog inputs.]

A ☐ Press the meter button so that the L/R Channel Input meters are active.

B ☐ Press System Setup button to access the System Setup screen.

```
|   I/O    |  STEREO  |  REMOTE  |   TEST    |
|  CALIB   | ENCODER  |INTERFACE |           |
```

C ☐ From the System Setup screen, press I/O CALIB soft key button.

```
|ANLG INP| DIG INP | ANLG OUTP|DIG OUTP|
| CALIB  |  CALIB  |   CALIB  | CALIB  |
```

D ☐ Press ANLG INP CALIB.

```
| INPUT | AI REF VU|AI REF PPM| AI CLIP|
|analog | +4.0  dBu| +9.0 dBu |+20.0dBu|
```

E ☐ Set Analog Input Reference level.

This step calibrates the 2200 to the level to which your studio operators peak their program material on the studio meters. This assures that the 2200's processing presets will operate in their preferred range.

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.

Calibration with Audio: Follow these steps if you are able to interrupt or distort programming. This will achieve the most precise calibration. You may adjust this level with a standard reference/line-up level tone from your studio or with program material.

Calibration by Numbers: Follow these steps if you cannot interrupt or distort programming.

Calibration with Audio

a) Depending on whether you monitor program levels by VU or PPM meters at the console or mixer, adjust the appropriate 2200 reference level control (either AI REF VU or AI REF PPM) for an average of –10dB on the Master Gain Reduction meter when audio is peaking at normal levels (e.g., 0VU).

Hold down the appropriate reference level soft key button, turn the control knob to scroll to the appropriate level, then release the button.

Calibration by Numbers

a) Depending on whether you monitor program levels by VU or PPM meters at the console or mixer, adjust the appropriate 2200 reference level control (either AI REF VU or AI REF PPM) to your studio's reference level.

Hold down the appropriate reference level soft key button, turn the control knob to scroll to the appropriate level, then release the button.

8. Enable Digital Input.

[Skip this step if you do not have Model 2200-D or if you are not using the digital input.]

A ☐ Press System Setup button to access the main System Setup screen.

B ☐ From the System Setup screen, press the I/O CALIB soft key button.

```
|ANLG INP| DIG INP | ANLG OUTP|DIG OUTP|
| CALIB  |  CALIB  |   CALIB  |  CALIB |
```

C ☐ Press DIG INP CALIB soft key button.

```
| INPUT  |DIG STAT| DI REF VU|DI REF PPM|
|analog  |  lock  |–19.5dBFS |–19.5dBFS |
```

Note: If DIG STAT is no lock, then the AES/EBU digital input is not valid. Check connections, cabling, and digital source.

D ☐ Enable Digital Input.

[analog], [digital] or [dig+J17]

Hold down the button directly below the word "INPUT," turn the control knob clockwise to choose digital, or digital+J17, then release the button to set the parameter.

9. Calibrate Digital Input to your standard studio level.

[Skip this step if you do not have Model 2200-D or if you are not using the 2200-D's digital input.]

A ☐ Press System Setup button to access the System Setup screen.

```
| I/O     |  STEREO |  REMOTE  |  TEST  |
| CALIB   | ENCODER | INTERFACE|        |
```

B ☐ From the System Setup screen, press the I/O CALIB soft key button.

```
|ANLG INP| DIG INP | ANLG OUTP|DIG OUTP|
| CALIB  |  CALIB  |   CALIB  |  CALIB |
```

C ☐ Press DIG INP CALIB.

```
| INPUT  |DIG STAT| DI REF VU|DI REF PPM|
|analog  |  lock  |–19.5dBFS |–19.5dBFS |
```

D ☐ Set Digital Input Reference level.

This step calibrates the 2200 to the level to which your studio operators peak their program material on the studio meters. This assures that the 2200's processing presets will operate in their preferred range.

Calibrate with Audio if you are able to interrupt or distort programming. This will achieve the most precise calibration.

Calibrate by Numbers if you cannot interrupt or distort programming.

Calibrate with Audio

a) 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).

b) Adjust the appropriate 2200 reference level control (either DI REF VU or DI REF PPM) for –10dB on the AGC Gain Reduction meter.

   Hold down the appropriate reference level soft key button, turn the control knob to scroll to the appropriate level, then release the button.

Calibrate by Numbers

a) Adjust the appropriate 2200 reference level control (either DI REF VU or DI REF PPM) for your studio's reference level. Note that the numbers you see represent dB below digital full-scale.

   Hold down the appropriate reference level soft key button, turn the control knob to scroll to the appropriate level, then release the button.

10. Adjust Composite Output level controls.

[Skip this step if you are not using the 2200's composite outputs.]

A ☐ Feed the 2200 with program material or the built-in 400Hz TEST tone.

To turn on the TEST tone:

a) Press System Setup button.

b) Press TEST soft key button.

c) Set TONE to 400Hz: Hold down the TONE soft key button, turn the control knob to 400 Hz, then releasing the button.

d) Activate 400Hz test tone: Hold down the MODE soft key button, turn the control knob to scroll to tone, then release the button.

To turn off TEST tone:

Hold down the MODE tone soft key button, turn the control knob to scroll to operate, then release the button.

B ☐ Adjust the 2200's Comp 1 and Comp 2 level controls — screwdriver slots on the left side of the front panel — for 100% Total Peak Modulation of your FM exciter, as indicated on a modulation monitor, or modulation indicator on your exciter. In the U.S., you can modulate higher than 100% when using SCAs. Refer to the appropriate FCC rules.

If using a composite STL, adjust the 2200's Comp 1 and Comp 2 level controls for 100% Total Modulation of your composite STL transmitter, as indicated on the STL's modulation indicator. Then adjust your STL's receiver output level control and/or FM exciter composite input level control for 100% Total Modulation of your FM exciter, as indicated on a modulation monitor, or modulation indicator on your exciter.

C ☐ Continue to step 12.

11. Adjust Analog Left/Right or Digital Output level controls.

[Skip this step if you are not using the analog Left/Right or Digital Outputs.]

A ☐ Press System Setup button to access the main System Setup screen.

B ☐ Access analog or digital output level control. (Press ANLG OUTP CALIB or DIG OUTP CALIB as required.)

```
| AO 100%| AO PRE-E |
|–22.1dBu| pre-emph |
```

or

```
| DO 100% | DO PRE-E |DO RATE | DO SYNC|
| –2.8dBFS|   flat   | 32kHz  |internal|
```

C ☐ Set Output level.

Hold down AO 100% or DO 100%, as applicable, and adjust the knob.

Adjust the output level controls for 100% Total Modulation of your FM exciter, or discrete left/right STL, as indicated on a modulation monitor, or modulation indicator on your exciter or STL. In the U.S., you can modulate higher than 100% when using SCAs. Refer to the appropriate FCC rules.

12. Select a preset that complements 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 Recall Preset button to access the Recall Preset screen.

```
on air:CLASSICAL PROTECT          |RECALL|
next:  2B GENERAL PURPOSE         | NEXT |
```

B ☐ From the Recall Preset screen, select a preset.

Turn the control knob to scroll through the preset list. When you find a desired preset, press the RECALL NEXT soft key button. Press Escape button to return to main screen.

13. System Setup Completed!

If you want to set up additional input/output parameters, or reset any setup adjustments, continue to “System Setup Controls,” on the following pages. If you are ready to use the 2200, proceed to Section 3 for important 2200 operation information.

System Setup Controls

System Setup includes access to the following 2200 controls: I/O Calibration, Stereo Encoder, Remote Interface and Test mode. This section provides steps to access these System Setup controls, and descriptions of their respective controls and parameters.

I/O CALIB (I/O Calibration)

I/O Calibration provides the user with control of the audio input and output settings. Model 2200 provides control of analog I/O settings. Model 2200-D provides control for both analog and digital I/O settings.

To access I/O Calibration controls:

A ☐ Press the meter Function button so that the L/R Channel Input meters are active.

B ☐ Press System Setup button to access the System Setup screen.

```
|   I/O    |  STEREO  |  REMOTE   |   TEST   |
|  CALIB   | ENCODER  | INTERFACE |          |
```

C ☐ From the System Setup screen, press I/O CALIB soft key button.

```
|ANLG INP| DIG INP | ANLG OUTP|DIG OUTP|
|  CALIB |  CALIB  |   CALIB  |  CALIB |
```

D ☐ Access the desired input and output controls. Press the appropriate soft key, located directly below the words that indicate the I/O control you wish to adjust.

To change any of the I/O Calibration controls, hold down the soft key button directly below the screen function you wish to change, turn the control knob counterclockwise and/or clockwise to find the desired setting, then release the button.

ANLG INP CALIB (Analog Input Calibration)

- INPUT

[analog], [digital] or [dig+J17]

This sets the analog inputs, the AES/EBU digital input, or the AES/EBU digital input + J.17 pre-emphasis as the audio source.

Note: Digital sources are only available with Model 2200-D.

If digital lock is lost, the unit automatically switches to analog input. The unit automatically returns to digital input after 1 second of lock.

Note that dig+j17 de-emphasizes the digital signal according to the CCITT J.17 standard.

- AI REF VU and AI REF PPM

Analog Input REFerence VU: in two ranges, [–18.0dBu to +21.0dBu] or [–40.0dBu to –1.0dBu], jumper-selectable, in 0.5dB steps.

Analog Input REFerence PPM: in two ranges, [–13.0dBu to +26.0dBu] or [–35.0dBu to +4.0dBu], jumper-selectable, in 0.5dB steps.

Note: AI REF VU and AI REF PPM have two ranges; high and low, dependent on the input sensitivity jumper settings (see page 2-3). As shipped, the 2200 uses the higher range.

Note: AI REF VU and AI REF PPM are the same control at the system level and simply display the data in two scales. Moving either one changes the other.

This step sets the center of the 2200’s gain reduction range to the level to which your studio operators peak their program material on the studio meters. This assures that the 2200’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, in the case of PPMs, the actual average or RMS level.

If you know the reference VU or PPM level that will be presented to the 2200, set the appropriate AI REF to this level, but do verify it with the steps shown directly below.

The complete procedure for calibrating AI REF level is given in Basic System Setup, step E, page 2-22. Follow the complete procedure to calibrate the 2200 to your standard studio level.

To verify the correct setting of AI REF level with program material:

Recall the CLASSICAL PROTECT preset.

Observe the Gain Reduction meter on a wide range of program material, voice and music. It should average between 5.0 and 15.0dB.

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

If the Gain Reduction meter averages less gain reduction (higher on the meter), or if the Gated indicator stays on when program material is present, go back to the I/O CALIB screen, and re-adjust the AI REF level to a lower level.

If the AGC Gain Reduction meter averages more gain reduction (lower on the meter), go back to the I/O CALIB screen, and re-adjust the AI REF level to a higher level.

When finished, reset AGC to out if required, (e.g., if that was its setting prior to verifying AI REF level).

Model 2200-D only: This control has no effect on the AES/EBU digital input.

- AI CLIP (Analog Input Peak/Clip Level)

[+5.0dBu to +27.0dBu] or [–17.0dBu to +5.0dBu], 0.5dB steps.

Note: AI CLIP has two ranges; high and low, dependent on the input sensitivity jumper settings (see page 2-3). As shipped, the 2200 uses the higher range.

This setting calibrates the level at which the 2200's A-D (Analog-to-Digital) converter clips to the absolute maximum peak level that your installation supplies to the 2200's analog input.

This setting maximizes the 2200's signal-to-noise ratio. If the AI CLIP level is set too low, the 2200's analog-to-digital converters will overload and distort on program peaks. If AI CLIP is set too high, the signal-to-noise ratio will suffer. Use care and attention in setting this adjustment.

The complete procedure for calibrating AI CLIP level is given in Basic System Setup, step 6, page 2-20. Follow the complete procedure to calibrate the 2200 to your standard studio level.

If you know the maximum peak level that will be presented to the 2200, set AI CLIP to about 2dB higher than this level (for safety).

To verify the correct setting of AI CLIP with program material:

Press L/R Channel Input button to view the Input meters.

Access the AI CLIP control in the System Setup AI INP CALIB screen.

Observe the L and R Input meters (the two meters on the right) on a wide range of program material, including live studio voice. The meters should never reach as high as 0dB, which is the level at which the input A-D converter clips. But on the highest peaks, the meters should indicate as high as –3dB.

Be sure to observe the meters on live voice, which tends to have higher level peaks than recorded music.

If necessary, re-adjust the AI CLIP level.

Model 2200-D only: This control has no effect on the AES/EBU digital input.

ANLG OUTP CALIB (Analog Output Calibration)

- AO 100% (Analog Output Level)

[–22.1dBu to +20.0dBu], 0.1dB steps

Adjusts the analog left/right output level. The level indication on the screen is the maximum peak output level that the processing will produce to modulate the transmitter to 100% peak modulation.

- AO PRE-E (Analog Output)

[flat] or [pre-emph]

Controls whether the analog left/right outputs produce a flat signal, or a pre-emphasized signal, following the pre-emphasis set with Stereo Encoder PROC PRE-E control.

DIG INP CALIB (Digital Input Calibration)

Note: Digital Input controls are only applicable to Model 2200-D. Digital input uses sample rate conversion. The internal process rate is always based on internal clock.

- INPUT

  [analog], [digital] or [dig+J17]

  This selects the digital or analog input of the 2200-D. When digital (or digital+J17) is selected, and the digital input (incoming clock) is lost, the 2200-D will switch to analog input. the unit automatically returns to digital input after 1 second of lock. Note that you can also manually switch between analog input and digital input.

  Note that dig+J17 applies J.17 de-emphasis to the incoming digital audio.

- DIG STAT

  [lock] or [no lock]

  DIG STAT is not user-adjustable. It indicates the status of the digital input, either locked (if the AES/EBU digital input is valid) or unlocked (if it is not valid). When digital input is unlocked and input was selected as digital, then the input has automatically switched to the analog input.

- DI REF VU and DI REF PPM

  Digital Input REFerence VU: [–30.0dBFS to –10dBFS], in 0.5dBFS steps.

  Digital Input REFerence PPM: [–22.0dBFS to –2.0dBFS], in 0.5dBFS steps.

  Note: DI REF VU and DI REF PPM are the same control at the system level and simply display the data in two scales. Moving either one changes the other.

  The incoming audio signal can be referenced from –30dB to –10dB of the maximum allowable digital word.

DIG OUTP CALIB (Digital Output Calibration)

Note: Digital Output controls are only applicable to Model 2200-D.

- DO 100% (Digital Output Level)

  [–20.0dBFS to 0.0dBFS], in 0.1dB steps

  This control provides up to 20dB of attenuation to the digital output level. This level indicates the digital output level corresponding to 100% modulation, in dB below digital full-scale.

- DO PRE-E

  [flat], [pre-emph], [J.17] or [J.17+pre-e]

  Within the audio processing, the audio signal is pre-emphasized to either 50µs or 75µs (as set with Stereo Encoder PROC PRE-E control). This control sets whether the digital AES/EBU output remains pre-emphasized, produces a flat signal, applies J.17 pre-emphasis, or applies a combination of J.17 and previously set pre-emphasis.

Important: The 2200-D has the ability to produce an AES/EBU output signal with the pre-emphasis removed. It is sometimes desirable to locate the 2200 at the studio site, with its output feeding a digital STL. However, digital STL modems using lossy bit-rate reduction (such as Marti MD-2, Moseley DSP 6000, and TFT DMM92) are not designed to carry pre-emphasized AES/EBU data. The AES/EBU inputs applied to these units must be flat. If this is your configuration, you must make the 2200-D's AES/EBU output flat.

When the AES/EBU digital output of the 2200-D is sent directly to the AES/EBU input of a digital exciter, such as the Harris "Digit," the 2200-D's output should remain pre-emphasized. Also, if you are using a digital link with no data rate reduction, the 2200-D's output should be pre-emphasized.

- DO (SAMPLING) RATE

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

This control sets the data rate of the OPTIMOD digital output to 32kHz, 44.1kHz or 48kHz.

- DO SYNC

[internal] or [external]

This control determines whether the digital output sample rate is locked to the digital input sample rate, or whether internal sync (clock) is provided for 32, 44.1 and 48kHz output sample rates.

External sync is provided to sync the output to the input rate. In external mode, the digital output rate is derived from and frequency-locked to the digital input rate. If the input is 32kHz, the output will be 32kHz. If the input is 44.1kHz, the output will be 44.1kHz. If the input is 48kHz, the output will be 48kHz. The unit automatically switches to internal sync and selected output sample rate if the input rate exceeds +/-4% of the selected output rate (DO RATE). The unit automatically returns to external sync after 1 second at a valid sample rate.

DO SYNC may be external while INPUT is analog; in this case, the analog input provides the audio source and the digital input provides sync only for the digital output.

Stereo Encoder (and processing Pre-Emphasis)

Stereo Encoder provides the user with control of the digital stereo encoder.

To access Stereo Encoder controls:

A ☐ Press System Setup button to access the System Setup screen.

```
| I/O   |  STEREO |  REMOTE  |  TEST  |
| CALIB | ENCODER |INTERFACE|        |
```

B ☐ From the System Setup screen, press STEREO ENCODER soft key button.

```
|PROC PRE-E|  MODE  |PILOTLVL|XTLK TEST|
|   75   us| stereo |   9.0 %| normal  |
```

To change any of these controls, hold down the soft key button directly below the screen function you wish to change, turn the control knob counterclockwise and/or clockwise to find the desired setting, then release the button.

● PROC PRE-E

[50µs] or [75µs]

This controls the pre-emphasis of the internal processing's high-frequency limiters, and the pre-emphasis of the stereo encoder's output. It does not control whether analog left/right outputs are flat or pre-emphasized; they are controlled by AO PRE-E in the System Setup ANLG OUTP screen.

Set this control to the pre-emphasis standard in your country:

75µs	NORTH, CENTRAL, SOUTH AMERICA
50µs	EUROPE, ASIA, AFRICA, PACIFICA
	EXCEPT
75µs	TAIWAN, KOREA, THAILAND

● MODE (modulation)

[stereo], [pilotoff], [mono-L],[mono-R], [mono-SUM]

This control sets the modulation type, as follows:

stereo switches the 2200’s stereo encoder on, with pilot on. The level of the pilot tone is adjustable with the PILOTLVL control (see below).

[pilotoff] switches the 2200’s 19kHz stereo pilot tone off.

mono-L (mono from left) switches the 2200’s stereo encoder off, using the left input as the program source.

mono-R (mono from right) switches the 2200’s stereo encoder off, using the right input as the program source.

mono-SUM (mono from sum) sums the audio prior to processing and switches the 2200’s stereo encoder off.

● PILOTLVL (Pilot Level)

[8.0 % to 10.0 %], 0.1 steps

This control sets the level of the 2200’s 19kHz stereo pilot tone in percent modulation. The pilot will go off automatically if a mono mode is selected.

● XTLK TEST (Stereo Generator Test)

[normal], [main>sub] or [sub>main]

To facilitate measurement of main channel-to-subchannel [m>s]and sub-channel-to-main channel [s>m] crosstalk, two special test modes are provided. These apply the right channel audio directly to the main channel (L+R) or subchannel (L–R) inputs of the stereo encoder.

This control only affects the stereo encoder output, not the analog or digital audio outputs.

With XTLK TEST set to m>s or s>m, the MODE is automatically switched to stereo, and MODE is then limited to stereo or pilot off. When the user returns XTLK TEST to operate, the unit reverts to the MODE setting that was active when the test was first entered.

Remote Interface

The 2200 features eight rear panel opto-isolated inputs that allow you to direct the 2200 to perform certain functions when a voltage (6-24V) is presented to the input. Functions are assigned in the System Setup Remote Interface control screen.

To access Remote Interface controls:

A ☐ Press System Setup button to access the System Setup screen.

```
| I/O    |  STEREO | REMOTE   |  TEST  |
| CALIB  | ENCODER |INTERFACE|         |
```

B ☐ From the System Setup screen, press the REMOTE INTERFACE soft key button.

```
|REM INTF 1|REM INTF 2|REM INTF 3| MORE|
|user pst 1|user pst 2|user pst 3|   >>|
```

To change any of these controls, hold down the soft key button directly below the screen function you wish to change, turn the control knob counterclockwise and/or clockwise to find the desired setting, then release the button.

The actions of the remote interface will update the main screen accordingly with the new on-air preset (e.g., preset name, tone or bypass). They will not change any other 2200 screen.

Remote Interface points can be assigned as follows:

[user pst x], where x stands for a numeral between 1 and 8: Switches a user preset on the air. Any of the eight user programming presets may be recalled by the control interface. A momentary pulse of voltage will switch this function.

[x], where x stands for a factory preset name: Switches a factory preset on the air. Any of the eight factory programming presets may be recalled by the control interface. A momentary pulse of voltage will switch this function.

[bypass]: Switches the 2200's into bypass mode. A momentary pulse of voltage will switch this function.

[tone]: Switches the 2200's test tone on. A momentary pulse of voltage will switch this function.

[exit test]: If a test preset is switched on the air, exit test reverts to the previous processing preset (from either tone or bypass modes, or from XTLK TEST).

[stereo]: Switches the 2200's stereo encoder on, and the pilot on (if it was off). A momentary pulse of voltage will switch this function.

[mono-L]: Switches the 2200's stereo encoder off, using the left input as the program source. A momentary pulse of voltage will switch this function.

[mono-R]: Switches the 2200's stereo encoder off, using the right input as the program source. A momentary pulse of voltage will switch this function.

[mono-SUM]: Switches the 2200's stereo encoder off, summing the audio prior to processing. A momentary pulse of voltage will switch this function.

[analog in]: Selects the analog inputs as the audio source. A momentary pulse of voltage will switch this function.

[digital in]: In Model 2200-D only; Selects the AES/EBU digital input as the audio source. A momentary pulse of voltage will switch this function

[dig+J17 in]: In model 2200-D only; Selects the AES/EBU digital input as the audio source and applies J.17 de-emphasis prior to the processing. A momentary pulse of voltage will switch this function

TEST Mode

The 2200's Test Mode allow the user to calibrate, set up and test the transmission chain.

To access Test Mode controls:

A ☐ Press System Setup button to access the System Setup screen.

I/O	STEREO	REMOTE	TEST
CALIB	ENCODER	INTERFACE	

B ☐ From the System Setup screen, press TEST soft key button.

MODE	TONE	BYPASS GAIN
operate	1000 Hz	–6 dB

To change any of these controls, hold down the soft key button directly below the screen function you wish to change, turn the control knob counterclockwise and/or clockwise to find the desired setting, then release the button.

- MODE

[operate], [tone] or [bypass]

This control activates Tone Mode, Bypass Mode, or normal operate mode.

Select operate to use 2200 processing.

Select Tone to put a tone on-air. Test tones apply 100% L+R modulation.

Select Bypass to bypass 2200 processing. Note that bypass includes 50µs or 75µs pre-emphasis.

- TONE

[30 Hz], [100 Hz], [400 Hz], [1000 Hz], [10000 Hz], [BESSEL] or [15000 Hz]

This control set the frequency of the test tone. The tone is activated in Tone Mode (see above). The default tone frequency is 100Hz. The bessel tone is 13.5868kHz. It causes +/-75kHz deviation of the FM carrier when the carrier frequency (as observed on a spectrum analyzer) nulls for the second time as the level of the tone is increased from zero modulation.

- BYPASS GAIN

[–18dB to +15dB], in steps of 1dB

This control sets the level of signal passed through the 2200 in Bypass Mode. The default bypass gain setting is 0dB. Increase or decrease the Bypass signal level, as desired.

Section 3
Operation

page contents

Caution

The installation and servicing instructions in this manual are for use by qualified personnel only. To avoid electric shock do not perform any servicing other than that contained in the Operating Instructions unless you are qualified to do so. Refer all servicing to qualified service personnel.

2200 Controls and Meters

2200 Front Panel

Screen Display labels the four soft key buttons and provides control setting information.

Screen Saver feature: The screen has a built-in screen saver that turns off the backlight after approximately one hour. The screen turns back on when any front panel control is touched. Note that buttons don't perform their normal function when the screen is blank. Similarly, the control knob's first turn is not read, until it stops for a second or so.

Contrast Button adjusts the optimum viewing angle of the screen display. Press this button to cycle through four contrast settings for the screen display.

Four Soft Key Buttons provide access to all 2200 functions and controls. The functions of the buttons change with each screen according to the labels at the bottom of each screen. Push a button:

To select options (always identified on the screen by all-capital-letter words surrounded by left and right vertical bars), press the button directly below the option.

To change a parameter setting (always identified by lower-case letters or numerals), hold down the button directly below the parameter setting, turn the control knob to scroll through choices, and release the button to set the parameter.

Control Knob is used for changing data in one of three methods.

To scroll through submenu choices: Presets (on Recall Preset screen), FULL CONTROL parameters (on Modify Processing FULL CONTROL screen) and 8 Remote Interfaces (on System Setup REMOTE INTERFACE screen).

To change a parameter setting, by ***simultaneously holding down a soft key***. The parameters being changed take effect immediately, except for the following system level controls: MODE (on test screen), MODE (on Stereo Encoder screen), INPUT, AO PRE-E, DO PRE-E, DO RATE, DO SYNC and XTLK TEST. The setting for these controls do not take effect until the soft key is released.

To adjust the LESS-MORE control. Adjusting this control changes the sound immediately.

Escape Button returns the user to the previous screen; pressing this button repeatedly will always return you to main screen, which shows the on-air preset name.

Recall Preset Button brings up a screen that displays the current on-air preset and next preset (which can be changed by turning the control knob). To put a different preset on-air, turn the control knob to find the preset desired, then press the RECALL NEXT soft key.

When the button's yellow LED is lit, the Recall Preset screen is displayed.

Modify Processing Button brings up a screen to modify parameters for the current on-air preset. For Two-Band presets: LESS-MORE, EQ and FULL CONTROL. For the Protection preset, DRIVE and 30HzHPF.

> When the button's yellow LED is lit, the Modify Processing screen (or one of its submenus) is displayed.

System Setup Button brings up a screen to modify system settings (such as I/O levels). There are four System Setup submenus: I/O CALIB, STEREO ENCODER, REMOTE INTERFACE, TEST.

> When the button's yellow LED is lit, the System Setup screen (or one of its submenus) is displayed.

HF Limiting LEDs light when the high-frequency content of audio is being limited by the very fast high-frequency limiters. These LEDs indicate when greater than 0.5dB HF limiting is occurring.

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

Composite Meter is a 10-segment bargraph showing the stereo encoder's composite output level before the composite level controls.

Function Button selects which of three functions are displayed in the Function meters: Enhance, L/R Input or L/R Output.

Function LEDs indicate which function is currently displayed by the Function meters: Enhance, L/R Input or L/R Output. Press the Function button to toggle between the three functions.

Function Meter indicates level of Enhance, L/R Channel Input or Output, as selected with Function button. The meters operate over a –27dB to 0dB range. Input meters are referenced to clip level. Output meters are referenced to 100% modulation level. HF Enhance meter shows the active amount of enhancement activity. Since the HF Enhancement is program-dependent, it will vary with source material and the HF parameter. Note: HF Enhance is displayed only on the left-hand meter, below "HF."

Stereo Encoder Screwdriver-Adjustable Controls

> Orban supplies a special green-handled flat-blade screwdriver (Xcelite R3323) to adjust the stereo encoder controls.

Comp 1 sets the output level of Composite Output 1.

Comp 2 sets the output level of Composite Output 2.

Introduction to Processing

Some Audio Processing Concepts

Loudness is increased by reducing the peak-to-average ratio of the audio. 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 clipping 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.

Distortion in Processing

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

The FM pre-emphasis curve introduces further complications. Pre-emphasis boosts the treble at 6dB/octave starting at 2.1kHz (for 75µs countries) or 3.2kHz (for 50µs countries). This reduces the headroom available at high frequencies, and makes it difficult to achieve a bright sound. This is because bright sound requires considerable high-frequency power to appear at the output of the receiver’s de-emphasis filter, and thus requires a *very large* amount of high-frequency power to be transmitted so that a sufficient amount will survive the de-emphasis process.

Without very artful processing, the pre-emphasis will radically increase the level of the peaks and force you to decrease the average level proportionally. Orban’s high-frequency limiting and distortion-canceling clipping systems greatly ease this trade-off, but cannot entirely eliminate it. Therefore, you can only increase brightness by reducing average modulation (loudness) — unless you accept the increased distortion caused by driving the final clippers harder.

Loudness, Brightness and Distortion.

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

Perhaps the most difficult part of adjusting a processor is determining the best trade-off for a given situation. We feel that it is usually wiser to give up ultimate loudness to achieve brightness and low distortion. A listener can compensate for loudness by simply adjusting the volume control. But there is *nothing* the listener can do to make an excessively-clipped signal sound clean again, or to undo the effects of excessive high-frequency limiting.

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

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-FM — from Bach to rock

OPTIMOD-FM can be adjusted so that the output sounds as close as possible to the input at all times (using the Protection Limiter Structure), or so that it sounds open but more uniform in frequency balance (and often more dramatic) than the input (using the Two-Band Structure). The loudness/brightness/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 understand the trade-offs involved in optimizing one parameter (such as loudness) at the expense of others (such as brightness, distortion, or excessive density).

Never lose sight of the fact that, while the listener can easily control loudness, he or she cannot undo excessive high-frequency limiting or 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.

About the Processing Structures

In the 2200, a *processing structure* is a program that operates as a complete audio processing system. Only one processing structure can be active at a time.

There are two processing structures in the 2200:

Two-Band

Protection/Limiting

Unlike an analog system, where creating a complete processing system involves physically wiring its various components together, the 2200 realizes all of its processing structures as a series of high-speed mathematical computations made by Digital Signal Processing (DSP) integrated circuit chips. So the 2200 can be changed from one structure to another by loading new software from high-speed semiconductor memory within the 2200.

Factory Programming Presets

Factory Presets are our "recommended settings" for various program formats. These presets were designed by our experienced engineers as good starting points for the program format. In many cases, the sound of the factory preset will suit your needs without the need for further adjustment.

There are eight factory presets:

Classical-Protect
2B General Purpose
Talk
Music-Light
Music-Medium
Music-Heavy
Music+Bass Medium
Music+Bass Heavy

Classical-Protect produces a very clean, open sound that is ideal for stations whose success depends on attracting and holding audiences for very long periods of time. It uses the Protection/Limiting structure. All other presets use the Two-Band structure.

2B General Purpose provides an average amount of processing;

Talk provides processing for Talk format stations that primarily feature news, call-in shows, interviews, and other voice material. TALK keeps the levels of announcers and guests consistent, and keeps a proper balance between voice and commercials.

Music-Light produces a very open, unprocessed sound. This is a sound that is easily listenable for many hours without fatiguing listeners.

Music-Medium provides processing that is between Music-Light and Music-Heavy. This is a good choice for many stations.

Music-Heavy provides aggressive processing for stations that want to maximize on-air loudness, and that do not assume that a listener will listen to the station for hours at a time.

Music+Bass Medium produces a very punchy, clean, open sound.

Music+Bass Heavy provides aggressive processing with additional bass punch.

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 LESS-MORE, EQ and FULL CONTROL.

To recall a preset:

Press Recall Preset Button. Turn the control knob to scroll through preset list and stop when you find a desired preset. Press RECALL NEXT soft key button.

It is normal for the audio to mute for about a second when switching between a preset based on the Two-Band structure and a preset based on the Protection/Limiting structure. This gives the 2200 time to download the appropriate code to its DSP chips.

Customizing the 2200's Two-Band Sound

The subjective setup controls on the 2200 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, brightness, and audible distortion.

When you start with one of our Two-Band factory presets, there are three levels of subjective adjustment available to you to let you customize the factory preset to your requirements:

LESS-MORE

EQ

FULL CONTROL.

LESS-MORE

After selecting a factory preset, LESS-MORE is the next level of adjustment.

As you go from less to more, the air sound will become louder, but (as with any processor) processing artifacts will increase. The single LESS-MORE control changes many different processing controls at the same time.

Many 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.

To adjust Less-More:

Press Modify Processing Button, then press LESS-MORE soft key button. Hold the LESS-MORE button down while turning the control knob to change LESS-MORE setting. When you find a setting you like, release the LESS-MORE button.

EQ

After LESS-MORE, EQ is the next level of adjustment. It gives you equalization control independent of the LESS-MORE control. EQ provides a 30Hz High-Pass Filter, Low Bass (boost) control and HF Enhancement. We give a detailed description of the EQ controls on page 3-11 (Two-Band Processing Control Details.)

To adjust EQ control

Press the Modify Processing Button, then press the EQ soft key button. From the EQ display, hold down the soft key button for the EQ control you want to change, turn the control knob to scroll through choices and release the button when you find a desired setting.

FULL CONTROL

With FULL CONTROL you can modify any subjective processing control to create a sound exactly to your taste. Use LESS-MORE to get as close as possible to your desired sound. Then use FULL CONTROL to make small changes to get the sound you want.

To adjust FULL CONTROL settings:

Press Modify Processing Button, then press the FULL CONTROL soft key button. From the FULL CONTROL display, hold down the soft key button for the control you want to change, turn the control knob to scroll through choices and release the button when you find a desired setting.

Saving Your Custom Settings

To save a custom setting:

Press Escape button until the on-air screen is displayed. Press SAVE CHANGES soft key button, then turn the control knob to choose one of the eight USER PRESET names. Press the SAVE CHANGES button to save custom setting.

Two-Band Processing Control Details

The Two-Band Structure is an improved version of Orban's classic 8100A OPTIMOD-FM, but with increased high frequency clarity and more features (Gated AGC, Bass Equalizer, High Frequency Enhancer).

Depending on how it is adjusted, it can produce an open, easy-to-listen sound that is similar to the source material, or it can produce a highly processed very loud sound that will never "get lost on the dial."

The Two-Band Structure's EQ controls

The Two-Band Structure has 3 equalization controls that work independently of the LESS-MORE control. The EQ controls are as follows:

30HzHPF (30Hz High Pass Filter) control determines if the 30Hz high-pass filter prior to the AGC is in or out of the signal path.

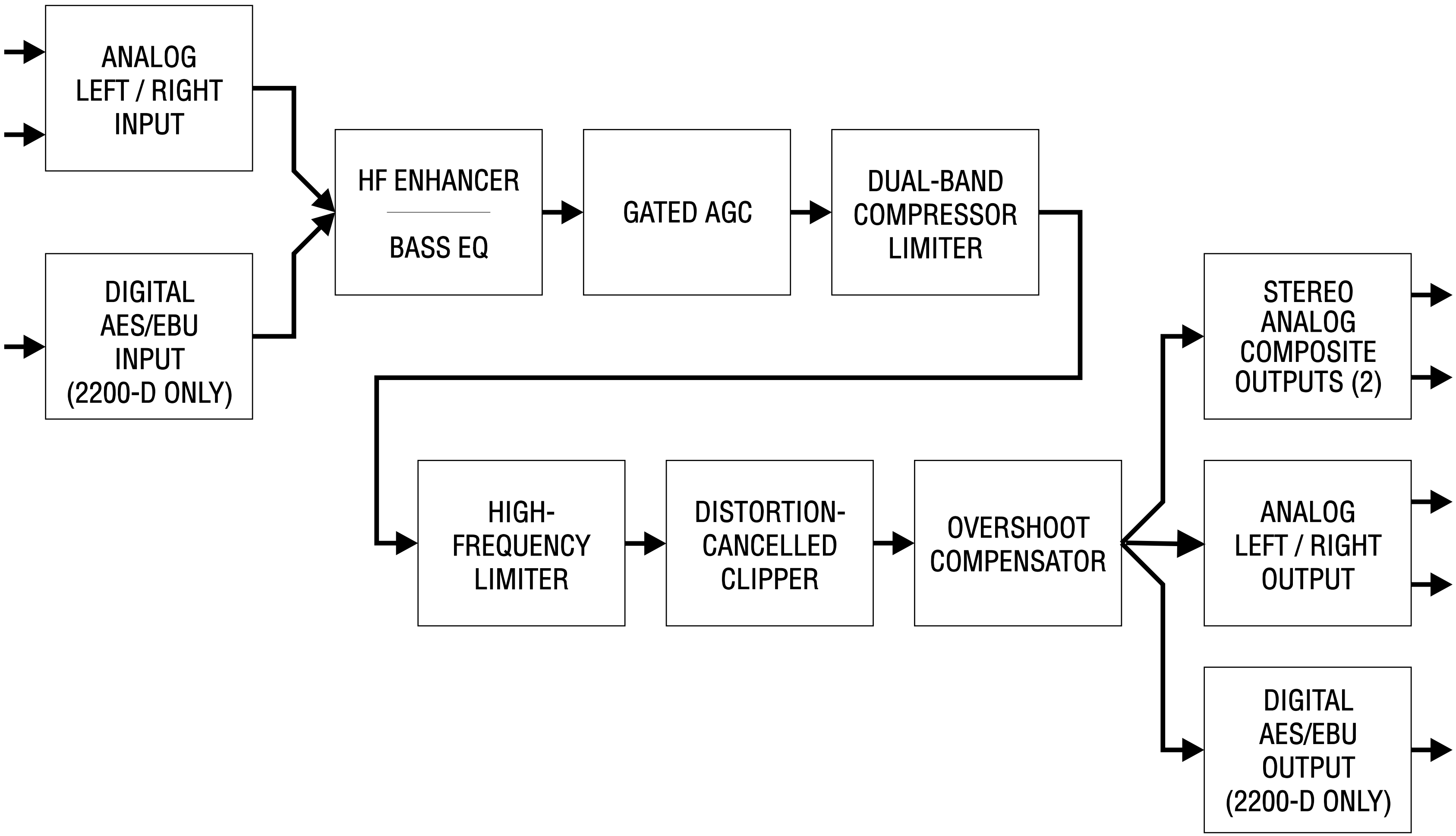
The 30Hz high-pass filter has an 18dB/octave slope, is down 0.5dB at 30Hz, and is located before the gain-riding AGC. It can be switched in or out-of-circuit by the 30HzHPF control.

This filter eliminates modulation-wasting subsonic energy from acoustic and turntable rumble, and removes most of the energy from pops caused by breath blasts into microphones. It prevents any such subsonic energy from modulating the audio processor's AGC and compressor control signals (which could cause unpleasant distortion), and prevents the automatic frequency control loops in FM exciters from introducing modulation distortion into the audio or even becoming entirely unlocked.

The cutoff frequency of this filter is so low that the only common musical instruments producing lower fundamental frequencies are the pipe organ and synthesizer. The bass energy in most pop music occurs above 40Hz. We recommend operating the system with the 30HzHPF control set in.

LOW BASS (Low Bass Boost) control is an equalization control designed to add punch and slam to rock and urban music. It provides a shelving boost from 0dB to +12dB in 1dB steps. The equalizer operates at 110Hz and below.

Because the Two-Band Structure often increases the brightness of program material when the HF ENHANCE control is used, 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!



TWO-BAND STRUCTURE
(SIMPLIFIED BLOCK DIAGRAM)

The moderate-slope (12dB/octave) shelving boost achieves a bass boost that is audible on smaller radios, but which can sound boomy on high-quality receivers. There are no easy choices here; you must choose the amount of boost you want by identifying your target audience and the receivers they are most likely to be using. In general, we recommend a +1 to +4dB boost for most formats.

HF ENHANCE (HF Enhancement) control sets the amount of high frequency energy added to program material. This enhancement is dynamically determined by the program material: Continuous analysis of program material intelligently and automatically determines the amount of equalization necessary at each moment to achieve detailed, defined program material that is never shrill or over-sibilant.

On mixed program material, HF ENHANCE usually produces the best sound if adjusted in the 0 to 5 range; the detail and definition of the program material is improved, yet the material does not sound excessively shrill. Always use your ears and judge how the processing affects the material.

Generally, if the material is very bright to begin with, you will hear little effect; if the program material is dull, you will hear very significant brightening. HF Enhancement works seamlessly with program material that contains previously enhanced tracks, like CD-quality music, because the 2200’s enhancement detects that such material already has considerable HF power and reduces its enhancement accordingly.

The Two-Band Structure’s Full Setup Controls

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

The AGC and the Two-Band gain reduction will eventually recover to their nominal levels. However, recovery is slow enough to be imperceptible. This avoids OPTIMOD-FM’s getting stuck with a large amount of gain reduction on a long, low-level musical passage immediately following a loud passage.

It is common to set the GATE THRS control to –40dB. Higher settings are primarily useful for radio drama, outside sports broadcasts, and other non-musical programming that contain ambiance, low-level crowd noise, and the like. Slightly higher settings may increase the musicality of the compression by slowing down recovery on moderate-level to low-level musical passages. When such passages cause the gate to cycle on and off, recovery time will be slowed down by the ratio of the “on time” to the “off time.” This effectively slows down the release time as the input gets quieter and quieter, thus preserving musical values in material with wide dynamic range (classical music, for example). The control can be set to off or adjusted over a range from –44dB to –15dB.

AGC (Automatic Gain Control) on/off control activates or defeats the slow AGC prior to the two-band compressor. If you are using an external compressor before the 2200 to protect an STL (like the Orban 8200ST OPTIMOD-Studio), set the AGC on the 2200 to off.

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). Total gain reduction 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 compressor control, discussed directly below. The range of the AGC DRIVE control is -10dB to +25dB.

REL TIME (Release Time) control determines how fast the two-band compressor releases (and therefore how fast loudness increases) when the level of the program material decreases.

It can be adjusted from 1dB/Second (slow) to 20dB/Second (fast). Settings toward 20dB/Second result in a more consistently loud sound on the air, while settings toward 1dB/Second allow a wider variation of dynamic range. The actual release time of the compressor is determined by both the setting of the REL TIME control and the dynamics and level of the program material. 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 REL TIME 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 REL TIME control) will no longer yield more loudness, and will simply degrade the punch and definition of the sound.

When the REL TIME control is set between 8 and 1dB/Second (the slowest settings), the amount of gain reduction is surprisingly non-critical. Since gating prevents noise from being brought up during short pauses, and pumping does not occur at high levels of gain reduction, 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. Therefore, when you operate the REL TIME 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 Structure.

With faster REL TIME 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 on the basis of listening tests how much gain reduction gives you the density 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 even though peaks are not actually increasing. If the gain-riding AGC is defeated (with the AGC 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-FM remarkably resistant to operator gain-riding errors.

2B DRIVE control adjusts signal level going into the two-band compressor, and therefore controls the dynamic range of the output audio by determining the amount of gain reduction in the two-band compressor. Depending on the setting of the REL TIME control (see above), the resulting sound texture can be open and transparent (low settings of 2B DRIVE), solid and dense (high settings of 2B DRIVE), or somewhere in between.

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 defeated), maintains a sense of dynamic range while still controlling levels effectively. Because OPTIMOD-FM's density gently increases between 0 and 10dB of compression, 10dB of compression sounds very natural, even on classical music.

BASS COUPL (Bass Coupling) control is used to set the balance between bass and the rest of the frequency spectrum.

The two-band compressor processes audio in a Master Band for all audio above approximately 200Hz, and a Bass Band for audio below approximately 200Hz. The BASS COUPL 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 COUPL 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 COUPL control until the Bass and Master Gain Reduction meters track as closely as possible.

With the Two-Band's REL TIME control set to 2dB/Second, setting the BASS COUPL 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 COUPL toward 100% (wideband) do not sound good. Instead, set the BASS COUPL 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 2200. 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.

HF LIMIT control determines how the processor avoids high-frequency overloads due to the pre-emphasis curve. When set toward −4.0dB (soft), the highs are controlled mostly by limiting (a form of dynamic filtering), which tends to soften highs — this could improve the sound of marginally distorted program material. When set toward +2.0dB (hard), the highs are controlled mostly by clipping, which could potentially distort highs.

Control of highs by limiting tends to slightly dull the sound. Control of highs by clipping doesn't reduce brightness, but the resulting sound can tend towards grittiness and smearing.

> Because the OPTIMOD-FM distortion-canceling clipper does not produce significant distortion at low frequencies, the HF LIMIT control will have a different effect on clipping distortion than you might expect. Outright break-up (principally sibilance splatter) will not occur — you must listen to the upper midrange and the highs to hear the effect of the clipper. Program material containing highly equalized hi-hat cymbals will clearly demonstrate the effect of adjusting the control.
>
> When the CLIPPING control is set to 0.0dB or below and the REL TIME control is set slower than 8dB/Second, it is possible to set the HF LIMIT control to +2.0dB without producing objectionable distortion (provided that the program material is very clean). If the CLIPPING control is set above 0.0 and/or faster release times are used (such that greater level and density is produced), it is usually necessary to readjust the HF LIMIT control closer to –2.0dB (soft) to avoid objectionable distortion. Fortunately, the high-frequency limiter knows that greater density and level have been produced when these other controls are set this way, and most of the necessary increases in high-frequency limiting will occur automatically. In fact, you will clearly hear a loss of highs when you adjust any control to produce greater loudness and density — this is an automatic response to the loudness/brightness/distortion trade-off inherent to all broadcast processing.
>
> We recommend that you examine the factory settings used by the "LESS-MORE" curves (adjust LESS-MORE to a given setting, then enter FULL CONTROL to look at the settings). This will help you learn about the trade-offs.

CLIPPING control adjusts signal level going into the distortion-canceling clippers and therefore determines the amount of peak limiting done by clipping. Range is –4.0dB to +2.0dB. This control and the FINAL CLIP control govern the trade-off between loudness and distortion.

OPTIMOD-FM controls fast peaks by distortion-canceled clipping. The CLIPPING control adjusts the level of the audio driving the distortion-canceled clippers, and therefore adjusts the peak-to-average ratio. The loudness/distortion trade-off is primarily determined by the CLIPPING control.

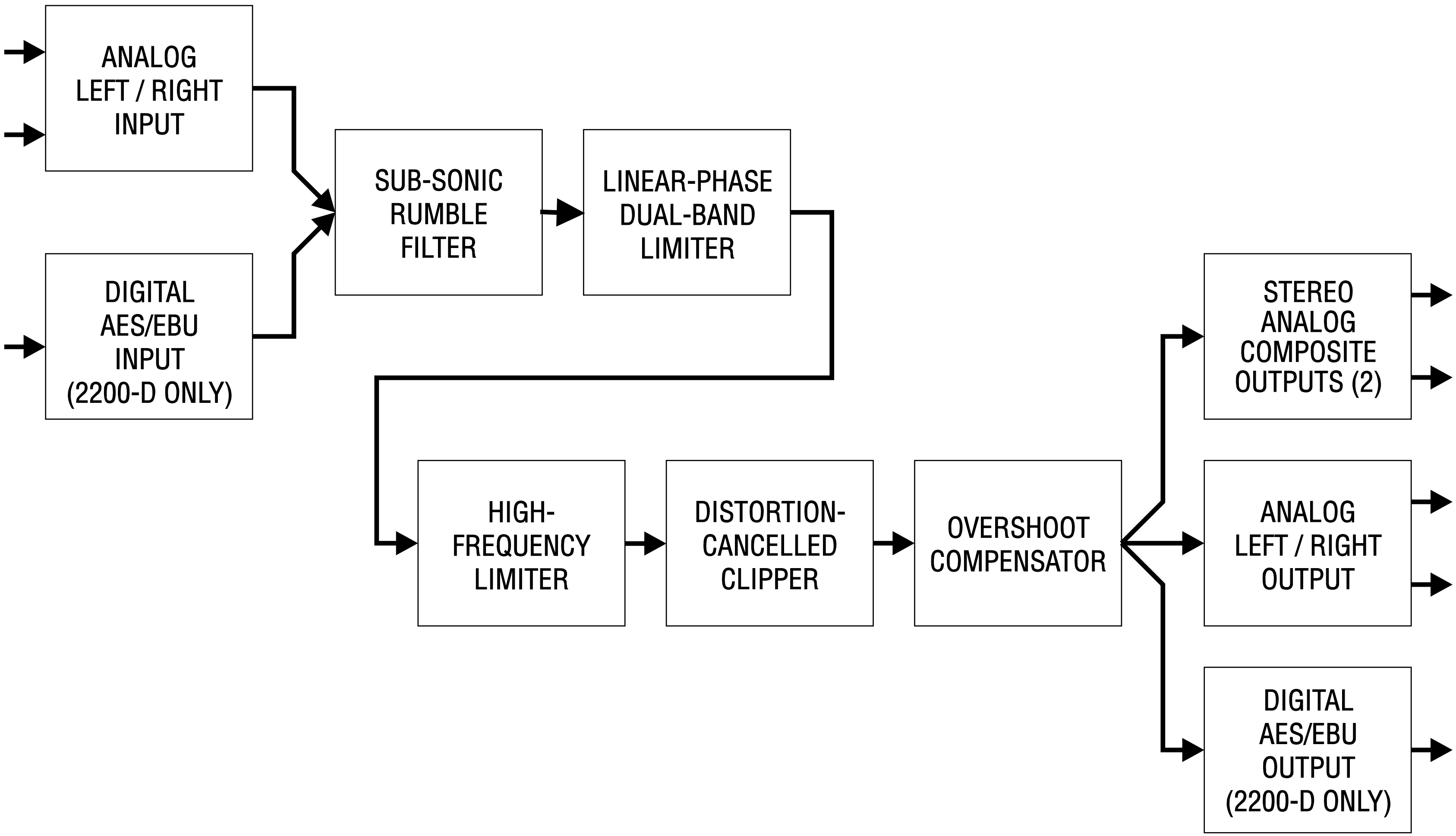
Turning up the CLIPPING control drives the distortion-canceled clippers harder, reducing the peak-to-average ratio, and increasing the loudness on the air. When the amount of clipping is increased, the audible distortion caused by clipping is increased. Lower settings reduce loudness, of course, but result in a cleaner sound and better high-frequency response.

> In our opinion, when the REL TIME control is set between 1 and 8dB/Second, the best setting for the CLIPPING control is between –1.0 and 0.0. If the program material is clean, this setting produces an output that sounds undistorted even on high-quality receivers.
>
> If you use faster settings of the REL TIME control, or if program material is not always clean, use lower settings of the CLIPPING control. Ultimately, your ears must judge how much distortion is acceptable. But audition difficult program material like live voice and piano before you make your final decision.

If you are optimizing for live voice, you will probably want to reduce the setting to the –3 to –4 range to eliminate any audible clipping.

FINAL CLIP (Drive) control determines the level driving the final clipper that performs protection peak limiting. This clipper follows the distortion-canceling clipper system, and is not itself distortion-canceling.

The effect of adjusting this control is very critical — changes of 0.1dB make clearly audible differences in the amount of distortion produced by the processing. In most cases, we recommend that the user not adjust this control and use the factory preset settings instead; the control has only been made available for experienced, sophisticated users who need to achieve the absolute maximum on-air loudness and who are willing to take the time necessary to listen to many different kinds of program material to verify that nothing falls apart after the clipper drive has been increased. The effect of adjusting this control is very similar to the effect of changing the amount of clipping in a composite clipper, except that in the 2200 (unlike a composite clipper), the SCA region of the baseband spectrum is always perfectly protected from interference.



PROTECTION LIMITER STRUCTURE
(SIMPLIFIED BLOCK DIAGRAM)

Customizing The Protection Limiter Structure Sound

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. While the Protection Limiter Structure can readily reduce the dynamic range, it is designed to do so without increasing program density, loudness, or the consistency of sound from different sources. Its primary function is to protect the transmitter from over-deviation while preserving the spectral and textured quality of the source material.

There are virtually no user controls — the parameters of the structure have been chosen to make it audibly *undetectable*.

Protection/Limiting Control Details

There are two parameters for the Classical-Protect preset:

DRIVE adjusts signal level going into the compressor, and therefore controls the dynamic range of the output audio by determining the amount of gain reduction in the compressor. The range is 0-25dB.

30Hz HPF determines if the 30Hz high-pass filter prior to the limiter is in or out of the signal path.

The 30Hz high-pass filter has an 18dB/octave slope and is down 0.5dB at 30Hz. It eliminates modulation-wasting subsonic energy from acoustic and turntable rumble, and removes most of the energy from *pops* caused by breath blasts into microphones. It prevents any such subsonic energy from modulating the audio processor's AGC and compressor control signals (which could cause unpleasant distortion), and prevents the automatic frequency control loops in FM exciters from introducing modulation distortion into the audio or even becoming entirely unlocked.

The cutoff frequency of this filter is so low that the only common musical instruments producing lower fundamental frequencies are the pipe organ and synthesizer. The bass energy in most pop music occurs above 40Hz. We recommend operating the system with the 30Hz HPF control set in.

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Section 4
Maintenance

page contents

CAUTION

The installation and servicing instructions in this manual are for use by qualified personnel only. To avoid electric shock do not perform any servicing other than that contained in the Operating Instructions unless you are qualified to do so. Refer all servicing to qualified service personnel.

Routine Maintenance

The 2200 OPTIMOD-FM Audio Processor uses highly stable analog and digital circuitry throughout. Recommended routine maintenance is minimal.

1. Periodically check audio level and gain reduction meter readings.

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

2. Listen to the 2200's output.

A good ear will pick up many faults. Familiarize yourself with the "sound" of the 2200 as you have set it up, and be sensitive to changes or deteriorations. But if problems arise, please don't jump to the conclusion that the 2200 is at fault. The troubleshooting information in Section 5 will help you determine if the problem is with OPTIMOD-FM or is somewhere else in the station's equipment.

3. Periodically check for corrosion.

Particularly in humid or salt-spray environments, check for corrosion at the input and output connectors and at those places where the 2200 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 2200 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 twenty screws holding the top cover in place, and lift the top cover off.

Use a #1 Phillips screwdriver.

2. Removing the Display Assembly.

Note: You will need to complete this step if you intend to remove the main board.

A ☐ Gently pull off the front panel control knob.

B ☐ Remove backlight power connector

C ☐ Detach the four ribbon cables that connect the display board to the main board and power supply board.

a) First, identify the four cables:

A larger white flat cable connecting to jumper J200 on the main board;

A gray ribbon cable connecting to a DIP header at jumper J203 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, you must first remove the retainer clip from the DIP header at J203.)

D ☐ Remove the four nuts that connect the display board to the main chassis.

Use a ¼“ long shafted nut driver or a flexible shaft nut driver.

E ☐ Grasp the edges of the front panel, pull slightly forward, and, while holding the panel vertical, guide it forward.

3. Removing the Main Board.

A ☐ If you have not done so yet, remove the top cover (Step 1) and remove the display assembly (Step 2).

B ☐ Remove the display board's metal shielding; this was left in place when the display board was detached.

C ☐ Remove the two hex nuts holding each composite output connector to the chassis, using a 3⁄16" hex nut driver.

D ☐ Remove the two hex nuts holding the Remote Interface connector to the chassis, using a 3⁄16" hex nut driver.

E ☐ Detach all 4 XLR connectors (or 6 with Model 2200-D), using a jeweler's screw-driver; engage the locking mechanism 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 four #1 Phillips screws 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.

4. Removing the Power Supply Board.

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

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 twelve #1 Phillips screws holding the heat sink to the side of the chassis — note that some have nuts and some are tapped — and remove heat sink.

D ☐ Remove the two Phillips screws that hold the IEC connector to the chassis.

E ☐ Remove the four bolts that hold the transformer; retain the four star washers and eight plastic insulators.

F ☐ Remove the three Phillips screws holding the power supply board to the main chassis.

G ☐ Remove the nut and star washer from the ground wire with a 1⁄4" nut driver.

H ☐ Carefully lift the power supply board up.

5. Reattaching the Power Supply Board.

A ☐ Replace the two Phillips screws that hold the IEC connector.

B ☐ Replace the twelve #1 Phillips screws that hold the heat sink to the side of the chassis.

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

D ☐ Replace the ground wire nut.

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

F ☐ Replace the four bolts that hold the transformer; use the starwashers and plastic insulators.

G ☐ If the main board is installed, reattach the two ribbon cables that connect the main board to the power supply.

6. Replacing the Main Board.

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

B ☐ Reattach all 4 XLR connectors (or 6 with Model 2200-D), using a jeweler's screwdriver.

C ☐ Replace the two hex nuts that hold the Remote Interface connector to the chassis, using a $^{3}/_{16}$" hex nut driver.

D ☐ Replace the two hex nuts that hold each Composite Output connector to the chassis, using a $^{9}/_{16}$" hex nut driver.

E ☐ Replace the four #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.

7. Replacing the Display Board.

A ☐ Reattach the display board’s metal shielding to the main chassis.

B ☐ Set the display assembly in place so that it aligns with its mounting holes.

C ☐ Replace the four nuts that connect the display board to the main chassis.

Use a ¼“ long shafted nut driver or a flexible shaft nut driver.

D ☐ Replace the four ribbon cables that connect the display board to the main board and power supply board.

E ☐ Replace the front panel control knob.

8. 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.)

In-System Testing ("Proof of Performance")

The FCC (Federal Communications Commission — U.S.A.) no longer requires periodic Proof of Performance measurements for FM stations. However, many stations will still wish to make periodic equipment performance measurements to ensure that their transmission system is working correctly and that they comply with all government regulations. The text below provides the general information that is needed to perform measurements verifying the performance of a transmission system including the 2200. Instructions for bench-top verification of 2200 performance *outside of the transmission system* are found below in **Field Audit of Performance** starting on page 4-16.

These instructions are written with the assumption that the analog inputs and outputs are used. If the 2200-D's digital I/O is used instead, follow the procedures below by analogy — you will have to supply digitized test tones.

The **NAB Broadcast and Audio System Test CD** provides a good source of digitally-generated test tones on compact disc. They supply tones at a 44.1kHz sampling rate that can be applied to the 2200-D's AES/EBU input. This test CD is available from:

NAB Services
1771 N Street N.W.
Washington, D.C. 20036, U.S.A.
order: (1) (800) 368-5644

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.)

- Precision FM modulation monitor or demodulator

  (Belar FMMA-1; TFT 844, or similar.)

- Precision stereo monitor or demodulator

  (Belar FMSA-1; TFT 844, or similar.)

- **optionally**, baseband spectrum analyzer, 0-100kHz

  (Tektronix 5L4N plug-in with 5111 bistable storage mainframe or similar.)

Monophonic Performance Verification

Monophonic performance verification is straightforward. Begin with the 2200 in stereo Mode, recall Bypass Test preset, and drive the Left OPTIMOD-FM Analog Input with the test signal.

1. Prepare the unit.

A ☐ Use the front panel controls to set the 2200's software controls to their default settings, as follows:

Preset

2B GENERAL PURPOSE

I/O CALIB (ANLG INP CALIB)

INPUT	analog
AI REF VU	+4.0dBu
AI REF PPM	+12.0dBu
AI CLIP	+20.0dBu

I/O CALIB (DIG INP CALIB); *2200-D only*

INPUT	analog
DIG STAT	no lock (depends if digital signal is present
DI REF VU	–30.0dBFS
DI REF PPM	–22.0dBFS

IO CALIB (ANLG OUTP CALIB)

AO 100%	+10.0dBu
AO PRE-E	flat

IO CALIB (DIG OUTP CALIB); *2200-D only*

DO 100%	–2.8dBFS
DO PRE-E	flat
DO RATE	32kHz
DO SYNC	internal

STEREO ENCODER

PROC PRE-E	50µs or 75µs, as appropriate to your country
MODE	stereo
PILOTLVL	9.0%
XTLK TEST	normal

TEST

MODE	operate
TONE	400Hz
BYPASS GAIN	0dB

B ☐ Press System Setup button to access System Setup menu.

C ☐ Press TEST soft key to access Test menu.

D ☐ Recall TONE preset: Press MODE soft key, use the control knob to scroll to tone, then release the MODE soft key.

E ☐ Adjust COMPOSITE 1 OUTPUT or COMPOSITE 2 OUTPUT for a reading of 100% on the modulation meter.

F ☐ Press System Setup button to re-access System 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 clipping, but leaves the 15kHz low-pass filter in the signal path.

I ☐ Inject both channels of the 2200 with a 50Hz signal, and adjust the signal generator's output level to obtain 100% composite modulation. (This level is the reference level.)

J ☐ Inject Left Channel at 50Hz with the reference level input (as established in the previous step).

Drive the left channel with the reference level input and ground the right channel (by tying pins #2 and #3 together).

K ☐ Press System Setup button to re-access System Setup menu.

L ☐ Press STEREO ENCODER soft key to access Stereo Encoder menu.

M ☐ Set 2200 for mono-L operation: Press MODE soft key, use the control knob to scroll to mono-L, then release the MODE soft key.

N ☐ Verify 100% modulation level on the left and right modulation meters.

O ☐ Set 2200 for mono-R operation: Press MODE soft key, use the control knob to scroll to mono-R, then release the MODE soft key.

P ☐ Verify no level on modulation meters.

Q ☐ Inject Right Channel at 50Hz with the reference level input (as established in the previous step).

Drive the right channel with the reference level input and ground the left channel (by tying pins #2 and #3 together).

R ☐ Press System Setup button to re-access System Setup menu.

S ☐ Press STEREO ENCODER soft key to access Stereo Encoder menu.

T ☐ Set 2200 for mono-R operation: Press MODE soft key, use the control knob to scroll to mono-R, then release the MODE soft key.

U ☐ Verify 100% modulation level on the left and right modulation meters.

V ☐ Set 2200 for mono-L operation: Press MODE soft key, use the control knob to scroll to mono-L, then release the MODE soft key.

W ☐ Verify no level on modulation meters.

Stereo Performance Verification

Many stations may wish to verify that they meet the requirements of the old part 73.322 of the FCC Rules (which was deleted in 1983, and which referred to stereo performance). Part 73.322 referred to the performance of the *transmitter only* (starting with stereo encoder input terminals), and measurements may be made by connecting the test oscillator directly to the OPTIMOD-FM main audio inputs. Following is an outline of the appropriate measurements and how to perform them.

1. Prepare the unit.

A ☐ Use the front panel controls to set the 2200's software controls to their default settings, as follows:

Preset

2B GENERAL PURPOSE

I/O CALIB (ANLG INP CALIB)

INPUT	analog
AI REF VU	+4.0dBu
AI REF PPM	+12.0dBu
AI CLIP	+20.0dBu

I/O CALIB (DIG INP CALIB)*; 2200-D only*

INPUT	analog
DIG STAT	no lock (depends if digital signal is present)
DI REF VU	–30.0dBFS
DI REF PPM	–22.0dBFS

IO CALIB (ANLG OUTP CALIB)

AO 100%	+10.0dBu
AO PRE-E	flat

IO CALIB (DIG OUTP CALIB)*; 2200-D only*

DO 100%	–2.8dBFS
DO PRE-E	flat
DO RATE	32kHz
DO SYNC	internal

STEREO ENCODER

PROC PRE-E	50µs or 75µs, as appropriate to your country
MODE	stereo
PILOTLVL	9.0%
XTLK TEST	normal

TEST

MODE	operate
TONE	400Hz
BYPASS GAIN	0dB

B ☐ Press System Setup button to access System Setup menu.

C ☐ Press TEST soft key to access Test menu.

D ☐ Recall TONE preset: Press MODE soft key, use the control knob to scroll to tone, then release the MODE soft key.

E ☐ Adjust COMPOSITE 1 OUTPUT or COMPOSITE 2 OUTPUT for a reading of 100% on the modulation meters.

F ☐ Press System Setup button to re-access System 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 clipping, but leaves the 15kHz low-pass filter in the signal path.

I ☐ Inject both channels of the 2200 with a 50Hz signal, and adjust its output level to obtain 100% composite modulation. (This level is the reference level.)

2. Test the main channel.

A ☐ Connect the oscillator to the left and right OPTIMOD-FM analog inputs in-polarity (“in-phase”).

B ☐ Set the oscillator to 50Hz and its level to the reference level (established in step 1-I, above).

C ☐ Observe the L–R meter on your stereo monitor.

> If L–R fails to null below –40dB, suspect a differential phase error between the left and right channels with•j the 2200. Such an error will also cause L+R and L–R to have poor frequency response, even if the left and right channels have accurate frequency response. Such an error could be caused by certain failures in the analog input stage.

> If you are doing the measurement from a remote location and driving the oscillator into a transmission link prior to the 2200’s input terminals, suspect a differential phase error between the left and right channels of the transmission link. If L–R fails to null below –20dB, this indicates that the phase error is large enough to potentially cause audible errors in the frequency response of the L+R signal.

D ☐ Using the L+R meter and output of your stereo monitor, measure the frequency response, total harmonic distortion, and noise characteristics of the main channel.

> As a minimum, measure harmonic distortion at 50, 100, 400, 1000, 5000, 10000, and 15000Hz, at 25%, 50%, and 100% modulation. If you have an automatic sweeping distortion test set, this can provide substantially more detailed information about system performance than does the spot-frequency tests because the distortion is measured at many more frequencies. However, bear in mind that the oscillator output level of any such instrument must be conditioned to follow the inverse of the FM pre-emphasis curve to hold percentage modulation constant and to prevent over-modulation at high frequencies. (The **NAB Broadcast and Audio Systems Test CD** has a series of tones whose levels precisely follow the 75µs and 50µs de-emphasis curves, and which can be applied to a pre-emphasized system

without need to readjust levels to hold modulation approximately constant.)

The old FCC Rules were ambiguous regarding the bandwidth of the stereophonic distortion measurements. Strict interpretation requires measurement of all distortion products up to 30kHz. The only way this can be done is by using a spectrum analyzer to examine the demodulated baseband, and by calculating a R.S.S. (root-sum-square) sum of all harmonics to 30kHz with appropriate correction for de-emphasis. However, all stereo monitors introduce a sharp-cutoff lowpass filter at 15kHz, and practical considerations thus limit stations without a spectrum analyzer to measuring only distortion products extending to 15kHz.

If the monitor's 15kHz lowpass filter is inadequate, leakage of the pilot into the monitor output may influence both THD and noise measurements. If this is the case, an external 19kHz notch filter may have to be used before the noise and distortion meter.

3. Test the stereophonic subchannel.

A ☐ Reverse the polarity of the right channel input to the 2200.

B ☐ Observe the L+R meter on your stereo monitor.

You should see the same amount of crosstalk as seen in the subchannel in step 2-C.

C ☐ Measure frequency response, total harmonic distortion, and noise for the stereo subchannel using the same techniques that you used for the main channel, but using the L–R meter and output of your stereo monitor.

Once again, only a spectrum analyzer can measure harmonic distortion to 30kHz (in this case, 38kHz±30kHz), and practical considerations usually limit the bandwidth of the measurement to 15kHz.

Measuring L–R noise is particularly problematical because most stereo monitors have no provision for applying de-emphasis to the L–R meter. Provided that the noise is uncorrelated (i.e., is dominated by hiss, rather than hum or discrete tones), then you can calculate the L–R noise by the formula:

$$s = 10 \log (10^{c/10} - 10^{m/10})$$

where

s is the L–R noise in dB below 100% modulation;
c is the left or right channel noise in dB below 100% modulation (assuming left and right noise measurements are almost equal); and,
m is the L+R noise in dB below 100% modulation.

4. Measure separation.

Careful reading of the old FCC Rule 73.322 reveals that there are no explicit requirements for frequency response, harmonic distortion, or noise performance of left or right channels. The only requirement specifically applicable to left and right channels is that separation must exceed 29.7dB, 50 to 15,000Hz, left-into-right and right-into-left.

A ☐ Connect the oscillator to the left OPTIMOD-FM analog input.

B ☐ Short out the right OPTIMOD-FM analog input by connecting pin #2 of the input XLR connector to pin #3.

If you fail to do this, the right input can pick up stray crosstalk from the oscillator that will falsify the separation measurement.

C ☐ As a minimum, measure left-into-right separation at 50, 100, 400, 1000, 5000, 10000, and 15000Hz at 100% modulation.

Remember to reduce the oscillator level at high frequencies to compensate for the FM pre-emphasis curve.

Because of the instability of many stereo monitors, the monitor should always be aligned according to the manufacturer's instructions before separation measurements are performed.

D ☐ Disconnect the oscillator from the left OPTIMOD-FM analog input, and connect it to the right OPTIMOD-FM analog input.

E ☐ Short out the left OPTIMOD-FM analog input by connecting pin #2 of the input XLR connector to pin #3.

F ☐ As a minimum, measure right-into-left separation at 50, 100, 400, 1000, 5000, 10000, and 15000Hz at 100% modulation.

5. Measure main-channel-to-subchannel and subchannel-to-main-channel crosstalk.

This step measures the crosstalk in the *transmitter* by using the 2200's crosstalk test (XTLK TEST) to eliminate trivial linear crosstalk due to slight phase differences between the left and right channels.

OPTIMOD-FM's crosstalk test facilitates measurement of main-channel-to-subchannel and subchannel-to-main-channel crosstalk. The test applies the output of the tone generator directly to either the main channel or subchannel stereo encoder input, and scales internal gains appropriately in the stereo encoder to keep total composite modulation constant.

A ☐ Verify bypass is still active: The Test menu should be displayed with the first parameter, MODE, set to bypass. (If bypass is not active, repeat steps 1-F thru 1-H, page 4-11.)

B ☐ Press System Setup button to re-access System Setup menu.

C ☐ Press STEREO ENCODER soft key to access Stereo Encoder menu.

D ☐ Set XTLK TEST to main>sub: Press XTLK TEST soft key, use the control knob to scroll to main>sub, then release the XTLK TEST soft key.

E ☐ Press System Setup button to access System Setup menu.

F □ Press TEST soft key to access Test menu.

G □ Recall TONE preset: Press MODE soft key, use the control knob to scroll to tone, then release the MODE soft key.

H □ Set the tone to the desired frequency.

Do the test at 50, 100, 400, 1000, 5000, 10000, and 15000Hz.

I □ Measure the signal level appearing in the stereophonic subchannel (L–R) on your stereo monitor. This is the main-channel to subchannel crosstalk.

J □ Set XTLK TEST to sub>main: Press XTLK TEST soft key, use the control knob to scroll to sub>main, then release XTLK TEST soft key.

K □ Measure the signal level appearing in the stereophonic main channel (L+R) on your stereo monitor. This is the subchannel to main-channel crosstalk.

Because crosstalk measurements on stereo monitors are usually derived from stable passive filters, these measurements are usually far more stable and reliable than separation measurements.

You can also measure the crosstalk levels on a spectrum analyzer connected to the demodulated composite output of the modulation monitor. This can be revealing, because the spectrum analyzer shows the difference between linear crosstalk and non-linear crosstalk. Linear crosstalk appears in the main channel at the same frequency as the oscillator, and in the subchannel at 38kHz ± [the oscillator frequency]. Non-linear crosstalk is crosstalk appearing at other frequencies than the linear crosstalk. Linear crosstalk is innocuous unless its level is very high (less than 20dB below 100% modulation), while non-linear crosstalk is distortion and will be demodulated as such by the receiver.

L □ Repeat steps 5-B through 5-K for each frequency at which crosstalk is to be measured.

6. Measure 38kHz Subcarrier Suppression.

A □ 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 clipping, but leaves the 15kHz low-pass filter in the signal path.

B □ Be sure that the oscillator is still connected to the 2200’s left analog input.

C □ Set the oscillator frequency to 7.5kHz.

D □ Set the oscillator output level to produce 100% composite modulation.

E □ Set XTLK TEST to sub>main: Press XTLK TEST soft key, use the control knob to scroll to sub>main, then release the XTLK TEST soft key.

F □ Measure the 38kHz subcarrier level on your stereo monitor.

You can also measure the 38kHz subcarrier level on a spectrum analyzer connected to the composite output of the modulation monitor.

7. Measure pilot tone frequency.

A ☐ Set XTLK TEST to normal: Press XTLK TEST soft key, use control knob to scroll to normal, then release the XTLK TEST soft key.

B ☐ Suppress the oscillator.

C ☐ Connect a frequency counter to the 2200's composite output.

D ☐ Make sure that the pilot tone is turned on, and, measure the pilot tone frequency on the counter.

It should be 19,000Hz ±1Hz.

8. Measure pilot tone injection.

This is most easily measured on your stereo monitor. All monitors have the ability to directly indicate the pilot tone injection, which should be between 8% and 10% modulation.

If you do not have a stereo monitor, you can measure the pilot tone injection with a spectrum analyzer connected to the 2200's composite output. The pilot tone should be 19kHz at –21dB below 100% modulation (for 9% injection).

The 2200 itself has a pilot level control. To reach it, press System Setup button, then press STEREO ENCODER soft key. To adjust the pilot level, press PILOTLVL soft key, turn the control knob until the modulation meter reads 9%, and then release the PILOTLVL soft key.

9. Test Over: Return to normal operation.

A ☐ To replace bypass with the previous on-air preset, simply press the Recall Preset button once.

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.)

- Two 620Ω ±5% resistors.

This procedure is useful for detecting and diagnosing problems with the 2200'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 2200 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 2200 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.

There are some countries where regulatory authorities require peak output levels to not exceed a specified level when driving post office lines. Germany, for example, limits the level to 4.5Vpeak (+12.3dBu). The Analog Output calibration screen (accessed via the System Setup button, then I/O CALIB and ANLG OUTP soft keys) allows you to preset the peak output level (AO 100%) produced by the processing in its normal operating mode. The absolute maximum output level in any test mode is 3dB higher than this. So, for example, German users should never preset the peak output level higher than +9.3dBu if they are driving post lines.

There are no tests in this procedure that require output levels above +10dBu. Users in countries with a +12.3dBu limit should therefore have no difficulty completing the tests.

(The maximum possible output level from the analog outputs is +20dBu.

1. Prepare the unit.

A ☐ Use the front panel controls to set the 2200's software controls to the following settings:

I/O CALIB (ANLG INP CALIB)

INPUT	analog
AI REF VU	+4.0dBu
AI REF PPM	+12.0dBu
AI CLIP	+20.0dBu

I/O CALIB (DIG INP CALIB); *2200-D only*

INPUT	analog
DIG STAT	no lock (depends if digital signal is present)
DI REF VU	–30.0dBFS
DI REF PPM	–22.0dBFS

IO CALIB (ANLG OUTP CALIB)

AO 100%	+10.0dBu
AO PRE-E	flat

IO CALIB (DIG OUTP CALIB); *2200-D only*

DO 100%	–2.8dBFS
DO PRE-E	flat
DO RATE	32kHz
DO SYNC	internal

STEREO ENCODER

PROC PRE-E	50µs or 75µs, as appropriate to your country
MODE	stereo
PILOTLVL	9.0%
XTLK TEST	normal

TEST

MODE	operate
TONE	400Hz
BYPASS GAIN	0dB

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

C ☐ Press System Setup button to access System 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 clipping, but retains 15kHz low pass limiting filters (both analog and DSP) in-line.

F ☐ Connect one 620Ω ±5% resistor between pin #2 and pin #3 of the LEFT ANALOG OUTPUT XLR connector, and one 620Ω ±5% resistor between pin #2 and pin #3 of the RIGHT ANALOG OUTPUT XLR connector.

G ☐ Connect the audio voltmeter between pin #2 and pin #3 of the LEFT ANALOG OUTPUT XLR connector.

H ☐ Connect the sine-wave oscillator to pins #2 and #3 of the 2200's LEFT ANALOG INPUT XLR connector. Set the oscillator to 50Hz, and adjust its output level to produce a level of 2.45Vrms (+10dBu) at the 2200's LEFT ANALOG OUTPUT XLR connector.

This level corresponds to 100% modulation.

I ☐ Disconnect the oscillator from the LEFT ANALOG INPUT XLR connector, and connect it to the RIGHT ANALOG INPUT XLR connector.

J ☐ Disconnect the audio voltmeter from the LEFT ANALOG OUTPUT XLR connector, and connect it between pin #2 and pin #3 of the RIGHT ANALOG OUTPUT XLR connector.

K ☐ Verify that the level at the RIGHT ANALOG OUTPUT XLR connector is 2.45Vrms (+10dBu).

L ☐ Disconnect the oscillator and audio voltmeter from the 2200.

Test Power Supplies (optional)

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

A ☐ Measure the +5 volt supply with the DVM. Verify the presence of +5 volts (±0.05V).

The +5 volt digital supply appears between TP6 and ground test point TP3 on the Power Supply Board.

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

The ripple and noise should not exceed 100mVp-p.

2. 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.

3. 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.

4. Check frequency response of Analog I/O.

If a tracking or sweep generator and spectrum analyzer are not available, the frequency response can be measured with an audio oscillator and N/D test set. If you will be doing this, ignore the rest of step 4, and instead: Connect the oscillator as in step 1-H, but reduce its output level by 20dB (to avoid overloading the 2200 at high frequencies). Connect the N/D test set to pins #2 and #3 of the 2200's LEFT ANALOG OUTPUT XLR connector. Measure the frequency response with the oscillator set to 1kHz, then verify that response at 50Hz, 100Hz, 400Hz, 5kHz, 10kHz, and 15kHz is within ±0.5dB of that measured at 1kHz. At 15kHz, the response should be within ±0.75dB. Repeat for the right channel.

A ☐ Connect the output of a tracking or sweep generator to pins #2 and #3 of the L ANALOG INPUT XLR connector. Set the generator for a 20 - 20,000Hz logarithmic sweep.

B ☐ Connect the input of a spectrum analyzer or oscilloscope to pins #2 and #3 of the 2200's LEFT ANALOG OUTPUT XLR connector.

C ☐ Adjust the output level of the tracking or sweep generator to obtain approximately 125mVrms (−15.91dBu) or less at the 2200's output (to avoid clipping the 2200 at high frequencies because of pre-emphasis).

D ☐ Verify that the swept output is flat >−0.75dB at 15kHz.

E ☐ Disconnect tracking or sweep generator from the LEFT ANALOG INPUT XLR connector, and connect it to the RIGHT ANALOG INPUT XLR connector.

F ☐ Disconnect the spectrum analyzer or oscilloscope from the LEFT ANALOG OUTPUT XLR connector and connect it to the RIGHT ANALOG OUTPUT XLR connector.

G ☐ Verify that the swept output is flat >–0.75dB at 15kHz.

H ☐ Disconnect the tracking or sweep generator and the spectrum analyzer or oscilloscope from the 2200.

5. Check noise and distortion performance of Analog I/O.

A ☐ Set AO PRE-E to pre emph: From System Setup IO Calib screen, press ANLG OUTP CALIB soft key, then press AO PRE-E soft key, use the control knob to scroll to pre-e, then release the AO PRE-E soft key.

B ☐ Connect a THD analyzer to the LEFT ANALOG OUTPUT XLR connector. Set the THD analyzer's bandwidth to 22kHz.

C ☐ Connect the oscillator to the LEFT ANALOG INPUT XLR connector.

D ☐ Verify that at 50Hz, 100Hz, 400Hz, 1kHz, 5kHz and 10kHz, THD does not exceed 0.1%. At 15kHz THD should be <0.5%.

For each frequency, adjust the output level to produce 2.45Vrms (+10dBu) at the 2200's LEFT ANALOG OUTPUT XLR connector.

In many cases, measured results will be constrained entirely by the quality of the oscillator, distortion analyzer, and/or by the presence of RF fields.

E ☐ Disconnect the THD analyzer from the LEFT ANALOG OUTPUT XLR connector, and connect to the RIGHT ANALOG OUTPUT XLR connector.

F ☐ Disconnect the oscillator from the LEFT ANALOG INPUT XLR connector and connect to the RIGHT ANALOG INPUT XLR connector.

G ☐ Repeat steps 5-C through 5-D for the right channel.

H ☐ Disconnect the oscillator and THD analyzer from the 2200.

I ☐ Short the 2200's left and right inputs by connecting pins #2 and #3 of the LEFT ANALOG INPUT XLR connector together, and by connecting pins #2 and #3 of the RIGHT ANALOG INPUT XLR connector together.

J ☐ Verify that the noise at the LEFT ANALOG OUTPUT XLR connector and the RIGHT ANALOG OUTPUT XLR connector is below –70dBu (80dB below 100% modulation).

Note that hum or buzz due to test equipment grounding problems and/or high-RF fields may result in falsely high readings. Such problems should become immediately apparent if the output of the THD analyzer is monitored with an oscilloscope.

K ☐ Remove the shorting jumpers from the 2200's inputs.

6. Check frequency response of Digital I/O (2200-D only).

A ☐ Use the front panel controls to set the 2200's software controls, as follows:

Preset

2B GENERAL PURPOSE

I/O CALIB (ANLG INP CALIB)

INPUT	digital
AI REF VU	+4.0dBu
AI REF PPM	+12.0dBu
AI CLIP	+20.0dBu

I/O CALIB (DIG INP CALIB)

INPUT	digital
DIG STAT	no lock (depends if digital signal is present)
DI REF VU	–30.0dBFS
DI REF PPM	–22.0dBFS

IO CALIB (ANLG OUTP CALIB)

AO 100%	+10.0dBu
AO PRE-E	flat

IO CALIB (DIG OUTP CALIB)

DO 100%	–2.8dBFS
DO PRE-E	flat
DO RATE	32kHz
DO SYNC	internal

STEREO ENCODER

PROC PRE-E	50µs or 75µs, as appropriate to your country
MODE	stereo
PILOTLVL	9.0%
XTLK TEST	normal

TEST

MODE	bypass
TONE	400Hz
BYPASS GAIN	0dB

B ☐ Connect the digital source generator to the AES/EBU input of the 2200-D.

C ☐ Inject the digital input with a level of -20dBFS at 1kHz (32kHz Sample Rate; 24 bits).

Measure the digital output levels and use the levels measured as the reference level for the following test.

D ☐ Verify the frequency response is ±0.5dB, falling off to >–1dB at 15kHz.

E ☐ Disconnect the digital source generator from the 2200.

7. Check noise and distortion performance of Digital I/O. (2200-D only)

A ☐ Use the front panel controls to set the 2200's software controls, as follows:

I/O CALIB (ANLG INP CALIB)

INPUT	digital
AI REF VU	+4.0dBu
AI REF PPM	+12.0dBu
AI CLIP	+20.0dBu

I/O CALIB (DIG INP CALIB)

INPUT	digital
DIG STAT	no lock (depends if digital signal is present)
DI REF VU	–30.0dBFS
DI REF PPM	–22.0dBFS

IO CALIB (ANLG OUTP CALIB)

AO 100%	+10.0dBu
AO PRE-E	flat

IO CALIB (DIG OUTP CALIB)

DO 100%	–2.8dBFS
DO PRE-E	flat
DO RATE	32kHz
DO SYNC	internal

STEREO ENCODER

PROC PRE-E	50µs or 75µs, as appropriate to your country
MODE	stereo
PILOTLVL	9.0%
XTLK TEST	normal

TEST

MODE	bypass
TONE	400Hz
BYPASS GAIN	0dB

B ☐ Connect the digital source generator to the AES/EBU input of the 2200-D.

C ☐ Inject the digital input with a digital signal with a sample rate of 32kHz (24 bits).

D ☐ Verify that at 50Hz, 100Hz, 400Hz, 1kHz, 5kHz and 10kHz, THD does not exceed 0.01%. At 15kHz THD should be <0.1%.

At each frequency, adjust the input level for an output level of approximately –2.8dBFS.

In many cases, measured results will be constrained entirely by the quality of the oscillator, distortion analyzer, and/or by the presence of RF fields.

E ☐ Disconnect the digital source generator from the 2200.

Test Stereo Baseband Encoder

1. Prepare for test.

A ☐ Apply AC power to the 2200.

B ☐ Verify 2200 software controls are set to their default settings. (Refer to page 4-8.)

C ☐ Press System Setup button to re-access System Setup menu.

D ☐ Press TEST soft key to access Test menu.

E ☐ Set TONE to 5000Hz: Press TONE soft key, use control knob to scroll to 5000Hz, then release the TONE soft key.

F ☐ Recall tone preset: Press MODE soft key, use the control knob to scroll to tone, then release the MODE soft key.

This test tone applies a digitally-generated 5000Hz sinewave at exactly 100% modulation to the 2200's D/A converters (91% in the case of the composite D/A)

G ☐ Press System Setup button to access System Setup menu.

H ☐ Press STEREO ENCODER soft key to access Stereo Encoder menu.

I ☐ Set MODE to pilotoff: Press MODE soft key, use control knob to scroll to pilotoff, then release the MODE soft key.

2. Test reconstruction filter gain.

A ☐ Connect the audio voltmeter/THD analyzer between ground (TP4) and TP501 on the main board.

B ☐ Verify that the audio voltmeter indicates +10.7dBu ±0.5dBu.

3. Measure 38kHz null.

A ☐ Press System Setup button to access System Setup menu.

B ☐ Press STEREO ENCODER soft key to access Stereo Encoder menu.

C ☐ Set MODE to stereo: Press MODE soft key, use control knob to scroll to stereo, then release the MODE soft key.

D ☐ Set XTLK TEST to normal: Press XTLK TEST soft key, use control knob to scroll to normal, then release the XTLK TEST soft key.

E ☐ Connect the spectrum analyzer to the 2200's COMPOSITE 1 OUTPUT. Adjust its span to 10kHz/div and its start frequency to 0kHz. Adjust its vertical scale to 10dB/division. Adjust its sensitivity so that the 5kHz spur is at the top of the screen.

The top of the screen now corresponds to 100% stereo modulation (±75kHz deviation).

F ☐ Using the spectrum analyzer, verify that the 38kHz component is <−70dB.

4. Measure Subchannel-to-Main Channel crosstalk.

A ☐ Press System Setup button to access System Setup menu.

B ☐ Press STEREO ENCODER soft key to access Stereo Encoder menu.

C ☐ Set XTLK TEST to sub>main: Press XTLK TEST soft key, use control knob to scroll to sub>main, then release the XTLK TEST soft key.

D ☐ Set the spectrum analyzer start frequency to 0kHz.

E ☐ Verify that the 5kHz spur surrounding 38kHz (33kHz and 43kHz) are below the top of the screen.

5. Measure Main-Channel-to Subchannel crosstalk.

A ☐ Press System Setup button to access System Setup menu.

B ☐ Press STEREO ENCODER soft key to access Stereo Encoder menu.

C ☐ Set XTLK TEST to main>sub: Press XTLK TEST soft key, use control knob to scroll to main>sub, then release the XTLK TEST soft key.

D ☐ Adjust the spectrum analyzer for a 32kHz to 44kHz frequency span.

E ☐ Verify that the 5kHz sidebands surrounding 38kHz (33kHz and 43kHz) are below the top of the screen.

6. Test pilot tone.

A ☐ Verify 2200 software controls are set to their default settings. (Refer to page 4-8.)

B ☐ Press System Setup button to re-access System Setup menu.

C ☐ Press TEST soft key to access Test menu.

D ☐ Set MODE to bypass: Press MODE soft key, use control knob to scroll to bypass, then release the MODE soft key.

E ☐ Verify that the 19kHz pilot is 21dB below the top of the screen (for 9% injection).

F ☐ Monitor the composite output with the frequency meter and verify that the pilot frequency is 19,000Hz (±1Hz).

G ☐ Monitor the composite output with the THD analyzer and verify that the THD of the pilot is below 0.1% with 80kHz lowpass filter.

7. Verify DC offset null.

A ☐ Verify 2200 software controls are set to their default settings. (Refer to page 4-8.)

B ☐ Press System Setup button to re-access System Setup menu.

C ☐ Press STEREO ENCODER soft key to access Stereo Encoder menu.

D ☐ Set MODE to pilotoff: Press MODE soft key, use control knob to scroll to pilotoff, then release the MODE soft key.

E ☐ Connect COMPOSITE 1 OUTPUT to a DC voltmeter.

F ☐ Verify that the observed DC output voltage is 0.00V (±30mV).

DC offset null is maintained with servo I501-A.

G ☐ Repeat for COMPOSITE 2 OUTPUT.

8. Check high frequency separation.

A ☐ Verify 2200 software controls are set to their default settings. (Refer to page 4-8.)

B ☐ Press System Setup button to re-access System Setup menu.

C ☐ Press TEST soft key to access Test menu.

D ☐ Set TONE to 15000Hz: Press TONE soft key, use control knob to scroll to 15000Hz, then release the TONE soft key.

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

This test tone applies a digitally-generated 15000Hz sinewave at exactly 100% modulation to the 2200's D/A converters.

F ☐ Observe the COMPOSITE 1 OUTPUT with the scope. Trigger the scope externally from the LEFT ANALOG OUTPUT. Set the scope sensitivity to 0.5V/div, and input coupling to "DC." Set the horizontal timebase to 0.2ms/div.

G ☐ Adjust VR500 (COMPOSITE 1) until the COMPOSITE 1 OUTPUT level is 4Vp-p.

H ☐ Press System Setup button to access System Setup menu.

I ☐ Press TEST soft key to access Test menu.

J ☐ Set MODE to bypass: Press MODE soft key, use control knob to scroll to bypass, then release the MODE soft key.

K ☐ Inject both analog inputs with a level at 50Hz to produce a level of 100% on the modulation monitor. (Use this as the input reference level.)

L ☐ Use the table below to determine the level at the various frequencies to drive the input channel.

Drive the Left Channel to measure separation left into right.

Drive the Right Channel to measure separation right into left.

Note: Separation >60dB at all frequencies

Frequency	Input Level (50µs)	Input Level (75µs)
50Hz	0.0dB	0.0dB
100Hz	0.0dB	–0.01dB
400Hz	–0.07dB	–0.15dB
1kHz	–0.41dB	–0.87dB
5kHz	–5.40dB	–8.16dB
10kHz	–10.36dB	–13.66dB
15kHz	–13.66dB	–17.07dB

M ☐ Verify that the baseline is flat. (To verify on scope, turn pilot off)

Variation from horizontal will typically be undetectable by eye. It must be less than ½ of a minor division on the scope graticule.

DO NOT USE AN ATTENUATOR PROBE. Such probes typically have enough phase error to completely invalidate any separation measurements. Note also that some scopes have enough phase error in their vertical amplifiers to make separation measurements inaccurate. If separation appears inadequate in this test, check it with another scope before assuming that the 2200 is faulty.

Note: If you have a Belar FMSA-1 stereo monitor, you may use it to measure separation. Be sure to turn the 2200 pilot on if you do this. Orban has determined that this instrument is sufficiently accurate to make separation measurements correctly.

Figure 4-1: Separation Scope Trace

9. Return OPTIMOD-FM to service.

A ☐ Remove the 620Ω resistors connected across the output in step 1-F.

B ☐ Recall your normal operating preset.

Field Alignment

The only circuits requiring calibration are the L/R Analog Output stages. Because the calibration procedure compensates only for the accumulated tolerances of time/temperature-stable components used in the circuitry, calibration is usually done once at time of manufacture, and is very unlikely to be required again over the life of the equipment. These field alignment instructions are therefore included primarily for reference — *routine alignment is neither necessary nor desirable* due to the high stability of the circuitry.

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 1, 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, ≥120kHz range

  Tektronix 5L4N plug-in with 5111 bistable storage mainframe, or similar.

  Audio Precision System One, or similar.

- 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.

CAUTION

If calibration is necessary, we *strongly recommend* that the unit in question be returned to the factory for calibration by our experienced technicians. They have access to special test fixtures and a supply of exact-replacement spare parts. Only in an emergency should you attempt to align and calibrate the 2200 in the field.

Follow these instructions in order, without skipping steps.

Refer to the drawings in Section 6 for locations of components and test points.

Prepare the Unit

1) Set the GROUND LIFT switch to the earth ground symbol setting (down setting), so that ground is connected.
2) Remove the 2200 from its rack and place it on a test bench *away from RF fields*.
3) Remove the 2200's top cover.
4) Apply AC power to the 2200.

   Allow the 2200 to finish its diagnostics routine before proceeding.

System Default Settings

1) Use the front panel controls to set the 2200's software controls to their default settings, as follows:

   Preset

   2B GENERAL PURPOSE

   I/O CALIB (ANLG INP CALIB)

INPUT	analog
AI REF VU	+4.0dBu
AI REF PPM	+12.0dBu
AI CLIP	+20.0dBu

   I/O CALIB (DIG INP CALIB)*; 2200-D only*

INPUT	analog
DIG STAT	no lock (depends if digital signal is present)
DI REF VU	–30.0dBFS
DI REF PPM	–22.0dBFS

   IO CALIB (ANLG OUTP CALIB)

AO 100%	+10.0dBu
AO PRE-E	flat

   IO CALIB (DIG OUTP CALIB)*; 2200-D only*

DO 100%	–2.8dBFS
DO PRE-E	flat
DO RATE	32kHz
DO SYNC	internal

   STEREO ENCODER

PROC PRE-E	50µs or 75µs, as appropriate to your country
MODE	stereo
PILOTLVL	9.0%
XTLK TEST	normal

   TEST

MODE	operate
TONE	400Hz
BYPASS GAIN	0dB

Calibrate and Test Analog Input/Output Circuitry

1. Prepare for test.

A ☐ Verify L/R analog outputs are loaded with 600Ω resistors.

B ☐ Apply power to the 2200.

C ☐ Press System Setup button to access System Setup menu.

D ☐ Press TEST soft key to access Test menu.

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

This test tone applies a digitally-generated 30Hz sinewave at exactly 100% modulation to the 2200's D/A converters.

F ☐ Set TONE to 30Hz: Press TONE soft key, use control knob to scroll to 30Hz, then release the TONE soft key.

G ☐ Adjust the output trim potentiometers VR400 (Left Channel) and VR401 (Right Channel) for a balanced output reading of +10dBu on both channels.

The reading of +10dBu corresponds to the AO 100% level setting of +10dBu.

Return OPTIMOD-FM to Service

1) Disconnect all test instruments from the 2200.

2) Replace top cover.

See page 4-3 for instructions.

3) Return the 2200 to its rack and reconnect it.

4) After the 2200 has been powered from the AC line, recall the desired operating preset, either locally or by remote control.

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Section 5
Troubleshooting

page contents

CAUTION

The installation and servicing instructions in this manual are for use by qualified personnel only. To avoid electric shock do not perform any servicing other than that contained in the Operating Instructions unless you are qualified to do so. Refer all servicing to qualified service personnel.

Problems and Possible Causes

Always verify that the problem is not the source material being fed to the 2200, or in other parts of the system.

RFI, Hum, Clicks, Or Buzzes

A grounding problem is likely. Review the information on grounding on page 2-12.

The 2200 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 almost always operate adjacent to high-powered transmitters without difficulty. In the most unusual circumstances, it may be necessary to reposition the unit to reduce RF interference, and/or to reposition its input and output cables to reduce RF pickup on their shields.

Particularly if you are using a long run of coaxial cable between the 2200 and the exciter, a ground loop or other problem may inject noise into the exciter's composite input — particularly if the exciter's input is unbalanced. This problem can almost always be cured by the Orban CIT25 Composite Isolation Transformer (see page 2-12).

Poor Peak Modulation Control

The 2200 ordinarily controls peak modulation to an accuracy of ±3%. This accuracy will be destroyed if the signal path following the 2200 has poor transient response. Almost any link can cause problems. Even the FM exciter can have insufficient flatness of response and phase linearity (particularly at low frequencies) to disturb peak levels. Section 1 of this manual contains a complete discussion of the various things that can go wrong.

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

Audible Distortion On-Air

Make sure that the problem can be observed on more than one receiver and at several locations. Multipath distortion at the monitoring site can be mistaken for real distortion.

Verify that the source material at the 2200's audio inputs is clean. Heavy processing can exaggerate even slightly distorted material, pushing it over the edge into unacceptability. Refer to Orban's publication *Audio Quality in the FM Plant* (included with your 2200) for hints on how to achieve the cleanest source quality.

The subjective adjustments available to the user have enough range to cause audible distortion at their extreme settings. Advancing the CLIPPING and/or FINAL CLIP controls too far will inevitably cause distortion. Setting the LESS-MORE control beyond “9” will cause audible distortion of some program material with all but the Protection Limiter structure.

If you are using the 2200 or the analog inputs of a 2200-D, 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 2200’s A/D converter will clip and distort. (See page 2-20).

If you are using an external processor ahead of the 2200, be sure that it is not causing problems. For example, if a “stereo enhancer” is used, be sure that it does not significantly increase the average level of the stereo difference channel (L–R). This will almost certainly exaggerate multipath distortion.

> The Orban 222A Stereo Enhancer is fully compatible with the 2200 and will not cause this problem.

Amplitude modulation of the carrier that is synchronous with the program (“synchronous AM”) can cause subtle distortion, and can exaggerate existing multipath distortion. Synchronous AM should be better than 35dB below 100% modulation as measured on a synchronous AM detector with standard FM de-emphasis (50µs or 75µs).

> The “incidental AM” position on most modulation or stereo monitors is insufficiently wideband to provide an accurate reading of synchronous AM — such metering was designed to indicate non-synchronous AM like hum and noise.

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 2200 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 THRESH (Gate Threshold) control, preventing noise below this level from being further increased.

If you are using the 2200’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 2200 to digitize the input. (This ratio is slightly better than 90dB.) 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 2200 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 2200’s noise floor.

The 2200’s optional 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 2200’s digital input will thus rarely limit the unit’s noise performance even with very high amounts of compression.

If a studio-to-transmitter link (STL) is used to pass unprocessed audio to the 2200, the STL’s noise level can severely limit the overall noise performance of the system because compression in the 2200 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. Section 1 of this manual has a more detailed discussion. Composite STL systems with marginal paths or co-channel interference can cause noise (hiss) which will be most apparent when listening in stereo.

Whistle on Air, Perhaps Only in Stereo Reception

This could be caused by a number of problems, any of which could present a spurious tone (perhaps supersonic) to the input of the stereo encoder. In any such case, the first thing to do is to examine the left and right analog outputs with a spectrum analyzer to see if any spurious tones are visible.

If the whistle is 6kHz, there could be a beat between the 2200's 32kHz sampling frequency and the 38kHz stereo subcarrier. In this case, the most probable cause is a failure of the 2200's digital-to-analog converter system, including a failure of the reconstruction filter following the converter.

A whistle at another frequency might be associated with power supply oscillation, oscillation of any opamp in the digital-to-analog converter system, or composite STL problems.

Shrill, Harsh Sound

This problem can be caused by excessively high settings of the HF ENHANCE control.

If you are driving an external stereo encoder with built-in pre-emphasis, you must set the 2200's AO PRE-E to flat to prevent double pre-emphasis, which will cause very shrill sound.

You will *always* achieve better peak control by defeating the pre-emphasis and input filters of an external stereo encoder, permitting the 2200 to perform these functions without overshoot. Section 1 of this manual contains a detailed explanation of these, and other, system design considerations.

Dull Sound

If driving an external stereo generator, dull-sounding source material can sound dull on the air.

If the 2200's output is set to flat there will be no pre-emphasis unless it is supplied elsewhere in the system. This will cause very dull sound.

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

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

The 2200 can generate its own test tones, and these can be triggered from the 2200's optically-isolated remote control terminals, or by recalling a tone preset from the front panel, Page 1-10 provides a further explanation. Recall bypass Test Tone to pass line-up tones transparently.

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 exciter after the 2200.

Interference from Stereo Into Subcarriers

A properly-operating 2200 generates an immaculately clean baseband, with program-correlated noise below −85dB above 57kHz. If the 2200 and the rest of the transmission system is operating correctly, subcarriers should experience no interference.

Interference from the stereo into a subcarrier is best diagnosed with a spectrum analyzer. First examine the spectrum of the 2200's composite output. If the 2200 is operating properly, program-correlated noise should be below −85dB above 57kHz.

Even the slightest amount of composite clipping will degrade this protection dramatically. Such composite clipping may be intentional (in a composite clipper), or unintentional (you could be over-driving a composite link between the 2200's composite output and the exciter's input).

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

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

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

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

19kHz Frequency Out-of-Tolerance

First, verify that a problem really exists by using a second frequency-measuring device and/or verifying the problem with your monitoring service. If the problem is real, you must replace crystal Y602; there is no frequency trim available.

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

This is often caused by relative phase shifts between the left and right channels prior to the 2200's input. This will cause innocuous linear crosstalk between the stereo main and subchannels (and *vice versa*). Refer to step 5-K on page 4-13.

General Dissatisfaction With Subjective Sound Quality

The 2200 is a complex processor which can be adjusted for many different tastes. For most users, the gamut offered by the LESS-MORE control is sufficient to find a satisfactory "sound." However, some users will not be satisfied until they have accessed other Modify Processing controls and have adjusted the subjective setup controls in detail to their satisfaction. Such users *must* fully understand the material in Section 3 of this manual to achieve the best results from this exercise.

By comparison to competitive processors, the 2200 offers a uniquely favorable set of trade-offs between loudness, brightness, distortion, and buildup of program density. If your radio station does not seem to be competitive with others in your market, the cause is usually problems with the source material, overshoot in the transmission link (including the FM exciter) following the 2200, or an inaccurate modulation monitor that is causing you to under-modulate the carrier. A station may suffer from any combination of these problems, and they can have a remarkable effect upon the overall competitiveness of a station's sound.

If you are competing with a station that has Orban's high-end 8200 Audio Processor, you will not be able to get the source-to-source consistency and low voice distortion offered by the 8200's Five-Band structure. However, you will still have a loud big-sounding signal. Stations that need the ultimate in state-of-the-art processing should upgrade to the Orban 8200.

Section 1 of this manual provides a thorough discussion of system engineering considerations, particularly with regard to minimizing overshoot and noise. Orban's publication *Audio Quality in the FM Plant* (included with the 2200) provides many suggestions for maximizing source quality.

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 2200'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 2200 — this is printed on the rear panel of the unit.

Telephone: (1) 510/351-3500

or Fax: (1) 510/351-0500

or Write: Customer Service
Orban
1525 Alvarado Street
San Leandro, CA 94577 USA

E-Mail: custserv@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 and operation procedures. If you are still unable to solve a problem, contact our Customer Service for consultation. Often, a problem is relatively simple and can be quickly fixed after telephone consultation.

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

Please refer to the terms of your Limited One-Year Standard Warranty, which extends to the first end user. After expiration of the warranty, a reasonable charge will be made for parts, labor, and packing if you choose to use the factory service facility. Returned units will be returned C.O.D. if the unit is not under warranty. Orban will pay return shipping if the unit is still under warranty. In all cases, 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 no smaller than 1.75″ (H) x 14.25″ (D) x 19″ (W) — 4.5 cm (H) x 36.2 cm (D) x 48.3 cm (W), with a minimum bursting test rating of 200 pounds (91 kg). Place the chassis in a plastic bag (or wrap it in plastic) to protect the finish, then pack it in the carton with at least 1.5 inches (4 cm) of cushioning on all sides of the unit. “Bubble” packing sheets, thick fiber blankets, and the like are acceptable cushioning materials; foam “popcorn” and crumpled newspaper are not. Wrap cushioning materials tightly around the unit and tape them in place to prevent the unit from shifting out of its packing. Close the carton without sealing it and shake it vigorously. If you can hear or feel the unit move, use more packing. Seal the carton with 3-inch (8 cm) reinforced fiberglass or polyester sealing tape, top and bottom in an “H” pattern. Narrower or parcel-post type tapes will not withstand the stresses applied to commercial shipments.

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

DELICATE INSTRUMENT, FRAGILE!

Insure the package properly. Ship prepaid, not collect. Do not ship parcel post.

Your **Return Authorization Number** must be shown on the label, or the package will *not* be accepted.

Section 6
Technical Data

page contents

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 2200-D.

Installation

Analog Audio Input

Configuration: Left and right.

Impedance: Electronically balanced 600Ω or >10kΩ load impedance, jumper-selectable.

Dynamic Range: >90dB.

Common Mode Rejection: ≥70dB at 50-60Hz. ≥45dB at 60Hz-15kHz.

Sensitivity: –20dBu to +20dBu to produce 10dB gain reduction at 1kHz, software- and jumper-adjustable.

Maximum Input Level: +27dBu.

Connector: XLR-type, female, EMI-suppressed. Pin 1 Chassis Ground, Pins 2 (+) and 3 electronically balanced, floating and symmetrical.

A/D Conversion: 18-bit.

Filtering: RFI-filtered, with high-pass filter at 0.15Hz.

Analog Audio Output

Configuration: Left and right. Flat or pre-emphasized (at 50µs or 75µs), software-selectable.

Source Impedance: 30Ω, ±5%, electronically balanced and floating.

Load Impedance: 600Ω or greater, balanced or unbalanced. Termination not required.

Output Level: Adjustable from –20dBu to +20dBu into 600Ω or greater load, software-adjustable.

Output Noise Level: ≤–90.0dB (Bypass mode, de-emphasized, 20Hz-15kHz bandwidth, referenced to 100% modulation).

Crosstalk: ≤–70dB, 20Hz-15kHz.

Distortion: ≤0.05% THD (Bypass mode, de-emphasized, 20Hz-15kHz bandwidth).

Connector: XLR-type, male, EMI-suppressed. Pin 1 Chassis Ground, Pins 2 (+) and 3 electronically balanced, floating and symmetrical.

Filtering: RFI-filtered.

Digital Audio Input (2200-D Only)

Configuration: Two-channel per AES/EBU-standard. 20-bit resolution.

Sampling rate: 25-55kHz, automatically-selected.

Connector: XLR-type, female, EMI-suppressed. Pin 1 Chassis Ground, Pins 2 and 3 transformer balanced and floating.

Input Reference Level: Adjustable from 0dBFS to –20dBFS, software-controlled.

Digital Audio Output (2200-D Only)

Configuration: Two-channel AES/EBU-standard. 18-bit resolution. Software-controllable for flat, pre-emphasized to the selected processing pre-emphasis, J.17 pre-emphasized, or pre-emphasized to the selected processing pre-emphasis plus J.17 pre-emphasis.

Sampling rate: 32kHz, 44.1kHz, or 48kHz, software-selected.

Connector: XLR-type, male, EMI-suppressed. Pin 1 Chassis Ground, Pins 2 and 3 transformer balanced and floating.

Status Bits: AES/EBU status bits are implemented to control pre-emphasis in the Orban 8208 digital Stereo Encoder.

Output Level Adjustment Output at 100% modulation, adjustable from 0dBFS to –22.8dBFS, software-controlled.

Composite Baseband Outputs

Configuration: Two (2) outputs, each with an independent output level control, output amplifier and connector.

Source Impedance: 0Ω voltage source or 75Ω (jumper-selectable), single ended, floating over chassis ground.

Load Impedance: 37Ω or greater. Termination not required.

Level (0Ω Source Impedance, 75Ω or higher Load Impedance): Adjustable 0.4Vp-p to 8.8Vp-p with front panel multi-turn output level controls, one per output.

Pilot Level: Adjustable from 8% to 10%, software-controlled.

Pilot Stability: 19kHz, ±0.5Hz (10° to 40° C).

D/A Conversion: 18-bit.

Signal-to-Noise Ratio: ≥85dB (Bypass mode, demodulataed, de-emphasized, 20Hz-80kHz bandwidth, referenced to 100% modulation, unweighted).

Distortion: ≤0.05% THD (Bypass mode, demodulated, de-emphasized, 20Hz-15kHz bandwidth, referenced to 100% modulation, unweighted).

Stereo Separation: At 100% modulation = 3.5Vp-p, >60dB, 30Hz-15kHz, >65dB typical at 1kHz; at 100% modulation = 1.0Vp-p, >50dB, 30Hz-15kHz.

Crosstalk (Linear): ≤–80dB, main channel to sub-channel or sub-channel to main channel) referenced to 100% modulation).

Crosstalk (Non-Linear): ≤–80dB, main channel to sub-channel or sub-channel to main channel) referenced to 100% modulation).

38kHz Suppression: ≥70dB; 75dB typical (referenced to 100% modulation).

76kHz and Sideband Suppression: ≥70dB; 80dB typical (referenced to 100% modulation).

Connector: BNC, floating over chassis ground. EMI-suppressed.

Maximum Load Capacitance: 0.047µF (0Ω source impedance).

Maximum Recommended Cable Length (0Ω Source Impedance): 100ft/30m RG-58A/U.

Filtering: RFI-filtered.

Remote Control Interface

Configuration: Eight opto-isolated inputs, user-programmable to select any eight of: User Presets, Factory Presets, Bypass, Tone, Exit Test (returns from Bypass or Tone), Stereo, Mono from Left, Mono from Right, Mono from Sum, Input Analog, Input Digital.

Control: Momentary or continuous low side contact closure. 10mA minimum sink current; 9VDC, 50mA rating.

Power Supply: Current-Limited 9VDC provided to facilitate use with contact closure.

Connector: DB-25, EMI-suppressed.

Filtering: RFI-Filtered.

Power

Voltage: 90-120VAC, 100-132VAC or 200-264VAC, 50-60Hz; 40VA.

Connector: IEC; detachable 3-wire power cord supplied. AC is EMI-suppressed.

Ground: Circuit ground is independent of chassis ground; can be isolated or connected with a rear panel switch.

Safety Standards: UL, CE, CSA.

Environmental

Operating Temperature Range: 32° to 122°F/0° to 50°C at nominal operating voltages.

Humidity: 0-95% RH, non-condensing.

Dimensions (W x D x H): 19“ x 14.25” x 1.75“/48.3cm x 36.2cm x 4.5cm. 1 rack unit high.

Weight: 12 lbs/5.4kg.

Shipping Weight: 15 lbs/6.8kg.

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 2200/2200-D. It starts with an overview of the 2200/2200-D 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 6-35 illustrates the following overview of 2200/2200-D circuit sections.

The 16.384MHz 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 2200/2200-D system.

The User Control Interface and LED Display Circuits section includes the connector, RF-filtering, and circuitry for the remote control inputs. It also includes circuitry for the front panel pushbutton switches, LED control status indicators, and LED Meters. The LED Meters measure various 2200/2200-D signal levels and display the results on six front panel 10-segment LED meters.

The L/R Input Circuits include the connectors and RF-filtering for the left and right audio inputs and the digital audio input, and the circuitry to interface these inputs to the digital processing.

The L/R Output Circuits include the connectors and RF-filtering for the left and right audio outputs and the digital audio output, and the circuitry to interface the digital processing to these outputs.

The Composite Output Circuits include the connectors and RF-filtering for the two composite outputs, and the circuitry to interface the digitally processed, stereo encoded signal to these outputs.

The DSP Circuits implement the bypass, test tone, audio processing, and stereo encoding functions using digital signal processing.

The Power Supply provides power for all 2200/2200-D circuit sections.

16.384MHz Oscillator and System Clocking

A synchronous clocking scheme is used on the 2200/2200-D to eliminate any asynchronous clocks operating in the sensitive regions of the L/R input A/D converter. A single 16.384MHz 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 and the 11.2896MHz free running crystal clock oscillator providing the

44.1kHz AES/EBU output sample rate (this does not fall within a sensitive region of the A/D). Synchronous counters are used to divide the 16.384MHz 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 and a 6.144MHz clock for providing the 48kHz AES/EBU output sample rate clock in addition to providing the AES/EBU input receiver with the ability to measure the input sample rate.

Component-Level Description:

The 16.384MHz digital output from crystal oscillator Y602 feeds the master clock (MCLK) inputs of both the input and the output SRC chips IC603 and IC615. The 16.384MHz clock also feeds flip-flop IC604, which divides by two to produce an 8.192MHz clock. The 8.192MHz clock feeds digital multiplexer chip IC610, which routes the 8.192MHz to AES/EBU digital audio transmitter chip IC616 when an internally generated 32kHz output sample rate is selected. The 8.192MHz 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 8.192MHz clock. The 8.192MHz clock is inverted by buffers IC605-A, -B to provide clocks 8.192MHZA* and 8.192MHZB* that have falling edges coincident with the transitions of the lower frequency clocks. 8.192MHZA* feeds the bit clock of the inter-DSP communication links following buffers IC710-B, -D. 8.192MHZB* feeds the A/D input clock (256 x sample rate), the L/R output D/A master clock, and the input bit clock on both the L/R output D/A and the composite D/A.

The 2.048MHz 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 2.048MHz signal, providing a clean 2.048MHz 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 2200/2200-D. 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-nine counter is placed between the VCO output and the phase detector comparator input. This places the VCO output at 18.432MHz. The divide-by-nine is implemented by the PLA IC618 between pins 2 and 15. A 6.144MHz clock is derived at the counter's divide-by-three point and is provided at pin 17 of the PLA. The PLA provides a buffered 18.432MHz output at pin 14 which feeds Z-180 microprocessor IC100.

IC614-A, -D provide buffered clocks 2.048MHZA and 2.048MHZB for driving the EXTAL inputs (pin 27) of the DSP chips. Each buffer drives four DSP chips.

The 256kHz clock output of IC613 (pin 15) is required for the DSP-to-composite D/A interface. The 128kHz clock (pin 14) is used for the inter-DSP word clock. The 128kHz, 64kHz and 32kHz clocks are all used in the LCD backlight drive circuit. The 32kHz clock is also used for the input word clock of both the output sample-rate converter (SRC) and the L/R output D/A. The 32kHz 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 32kHz 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 32kHz, phase synchronous. This ensures that the process-to-output buffer transfer internal to the DSP doesn't overlap the output buffer-to-peripheral transfer. The 8.192MHZB* clock that feeds the A/D input clock (IC312 pin 19) is internally divided down to produce a 32kHz word clock at IC312 pin 13 and a 2.048MHz bit clock at pin 14. These clocks are used to control the A/D-to-DSP serial interface and the input SRC-to-DSP serial interface.

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 and the remote contact closures. 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 2200/2200-D DSP and Control software.

1. Microprocessor and Power Monitoring Circuit

A Z-180 microprocessor executes software code required to control the functionality of the 2200/2200-D. The EXTAL port 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 2200/2200-D 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 2200/2200-D digital electronics. When the +5V line is above the minimum operating voltage of +4.75V, R103 will pull 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 RESET* low which puts the Z-180 into the reset condition, thereby preventing the Z-180 and the 2200/2200-D 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 2200/2200-D is powered down is stored in a EEPROM. The EEPROM does not lose data when the 2200/2200-D 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 2200/2200-D 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

2200/2200-D 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 IC202, IC205, IC207, IC303, IC305, IC609, IC611, IC708, and IC709, and to allocate the Z-180 data bus at appropriate times to various peripherals via tri-state data buffers IC120, IC204, and IC601. IC120 buffers or tri-states status information from the remote contact closure circuitry onto the Z-180 data bus. IC204 buffers or tri-states information from the user control interface onto the Z-180 data bus. IC601 buffers or tri-states status information from AES/EBU Receiver IC600 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 2200/2200-D 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

A remote interface connector and circuitry enables remote control of certain operating modes; the 2200 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 2200/2200-D will respond to the most recent control operation whether it came from the front panel, or remote interface.

Component-Level Description:

J101 is a 25-pin D-connector that connects the remote control input signals. The connector 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 2200/2200-D. 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 REMOTE* signal provided to IC120 pin 19 is brought low, IC120 places remote signals on the Z-180 data bus.

2. Switch Matrix and LED Indicators

Ten front panel pushbutton switches are arranged in a matrix, configured as two columns and six rows (the FUNCTION and CONTRAST keys have dedicated rows). These switches are the primary element of the physical user interface to the 2200/2200-D 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, and the remote control interface status; and updates the LED control status indicators accordingly.

Component-Level Description:

S200-S208 and S210 are the front panel pushbutton switches. CR200-CR204 and CR206 are the front panel LED control status indicators. Via decoder IC102, the host microprocessor Z-180 periodically selects data latch IC202 (on the display board) to drive one of the two columns in the switch matrix low, then commands tri-state data buffer IC204 (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 R202, R210-R213 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 IC202 which directly drives the LED Control Status Indicators.

3. LED Meter Circuits

The meter LEDs are arranged in an 8x8 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 alternately sends pairs of mapped 8-bit words to the data bus. One of the words, latched by a “row selector” latch, has a single rotating active bit to select one of the 8 rows. The other word, latched by a “column selector” latch, has active bits corresponding to those of the 8 LEDs in the selected row that are to be lit. The latched words control high-current Darlington transistor arrays which drive the LED matrix.

Component-Level Description:

The meter LED matrix consists of six 10-segment LED bargraph assemblies (CR208-CR213) and three discrete LEDs (CR214-CR216). IC208 contains eight Darlington transistors, each of which is connected to the cathodes of a “row” of the LEDs. Row selector latch IC207, controlled by the Z-180, alternately turns one of the eight transistors on, such that it will sink current through the LEDs selected by column selector latch IC205, also controlled by the Z-180. IC205 turns on the appropriate transistors inside current driver IC206 to drive the selected row of LEDs. IC206 gets its current from a storage capacitor fed directly by the power transformer’s lower voltage secondary winding. Resistors RP200 function as current limiting resistors.

L/R 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 pad and load are enabled or disabled by jumpers that are 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 removes DC from 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, C323 and C324. J301 is a jumper that removes or inserts the optional 600Ω termination load (R300) on the input signal. J302 and J303 are the jumpers that remove or insert the resistive divider (R301-R303) that pads the 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 latches IC304 and IC305. 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 IC302 and associated components. This stage balances the signal and attenuates by 3.5dB to scale the signal to the proper level for the analog-to-digital (A/D) converter. IC301-B and associated components comprise a servo amp to prevent DC from passing to the DSP. R334, R337, C302, and C303 make a simple 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, 18-bit sigma-delta converter, implemented on a dual-chip integrated circuit. The A/D oversamples the audio at 2.048MHz. It applies noise shaping, then it filters and decimates to a 32kHz sample rate. The samples are output in two's complement, 32-bit word, two-word frame serial format, with SPI compatible timing, MSbit first, and transmitted to the DSP. The 32kHz frame clock and 2.048MHz bit clock from the A/D function as master clocks for the 2200/2200-D input to the DSP. For more information on 2200/2200-D input clocking, please refer to "16.384MHz Oscillator and System Clocking."

Component-Level Description:

The balanced left analog input is applied to pins 3(+) and 4(-), and the balanced right analog input is applied to pins 26(+) and 25(-) of the A/D (IC312). The maximum differential signal that the A/D can accept is ±7.36Vpeak. The A/D samples the left and right inputs simultaneously at 64 times the 2200/2200-D sample rate of 32kHz. ICLKD, the master clock input of the A/D (pin 19), is fed an 8.192MHz clock providing the 2.048MHz 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 L/R* (pin 13). The SPI communication standard is used for this audio interface, with A/D as master and DSP IC700 as slave. The SPI format is: 32-bits/word, multiplexed stereo, word clock low represents left data present, MSB first, data transitions occur on rising edge of the bit clock, first 18 bits are valid, trailing bits are set low, MSB delayed one bit period from word clock edge. IC314 provides buffering to reduce the drive requirement of the on-board drivers on the A/D and to ensure that there are no overshoots or undershoots as a result of transmission line reflections that may degrade the performance of the A/D. IC109-D is required to invert the word clock to support the SPI interface.

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 32kHz 2200-D system sample rate. Via a synchronous serial interface, the SRC sends the 32kHz 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. R600 provides a 110Ω input impedance per the AES/EBU specification.

The Z-180 provides the active high reset signal (AES_RST) to IC600 mode control pins 17, 18, and 23, via latch IC609 pin 6. This is used when the 2200-D is asked to respond to analog audio input. When in the reset state, the receiver holds all outputs inactive (except MCK pin 19).

IC600 pins 2 through 6 and pin 27 are an output latch that provides AES/EBU status information, selected by the STATSEL line. The information on this latch is provided to the Z-180 data bus via tri-state data buffer IC601. STATSEL signal from IC611 pin 12 is applied to IC600 pin 16. When STATSEL is high, pins 2 through 6 and pin 27 contain information about the channel status bits. When STATSEL is low, pins 2 through 6 and pin 27 contain input sample rate and error information. The Z-180 reads these to determine if a valid AES/EBU signal and sample rate is present. CHSEL is used to select whether channel A or channel B status bits are present on IC600's output latch. When STATSEL is low, left channel status is made available, and when STATSEL is high, right channel status is made available.

Received AES audio is transmitted from the AES receiver to the input sample rate converter (SRC IC603), in the synchronous serial SPI format. 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. The frame clock is inverted by IC605-F for compatibility with the SRC's input port. 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 sample rate (external sync) is required.

The crystal oscillator (Y602) provides the SRC a master clock of 16.384MHz on pin 2. This MCLK frequency allows the input SRC to operate with input sample rates in the range of 8.192kHz (MCLK/2000) to 57kHz (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 IC609.

The MSDLY_I, BKPOL_I, and TRGLR_I pins of the SRC chip configure the chip for SPI format. 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 L/R input A/D converter (IC312). The SRC generates DIG_IN (data) on pin 23, and receives the bit clock and the word clock on pins 26 and 24 respectively. An inverted version of this word clock is used by the DSP chip to conform with the SPI format it requires.

L/R 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 MDAC stages scale and buffer the D/A output signal to drive the analog output stages to the correct level. The analog output stages drive the analog output XLR connectors with a low impedance, floating 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 single chip, stereo, 18-bit delta-sigma converter.

For information on 2200/2200-D system clocking, please refer to “16.384MHz Oscillator and System Clocking.”

Component-Level Description:

IC400 is the digital-to-analog (D/A) converter for the left and right output signals. The synchronous serial input interface consists of the bit clock, data and latch enable pins that are configured for the SPI format via DIF0 and DIF1 pins (for details on SPI, see page 6-11). The processed digital output (ANLG_OUT) is provided by DSP IC706 on its SAI output port SDO0 (pin 47), and is received by the D/A on pin 18.

An 8.192MHz bit clock is provided from the system clock circuitry to both the DSP and the D/A chips. The DSP output data format is SPI (32 bits per word, two words per frame). DSP chip IC706 receives a 128kHz frame clock at its WST input (pin 50) that sets the word transfer rate to eight words per 32kHz period. The D/A receives a 32kHz clock at its LRCK input (pin 20). 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 fifth sample for the right output. The DSP output samples are formatted to ensure that the D/A uses a left and right output pair 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 RC low-pass filtered and applied to an inverting amplifier having an adjustable (digitally-controlled) gain. The gain is set by an MDAC. The state of the MDAC 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 feeds a programmable de-emphasis filter stage with its response digitally-controlled by JFET switches. The de-emphasis stage feeds 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.

Component-Level Description:

The left channel signal emerging from the digital-to-analog (D/A) converter is RC low-pass filtered by R402 and C407 to remove high frequency images. It is then applied to an adjustable gain amplifier formed by VR400, R404-R406, C409, IC401, and IC402-A. These components form an inverting amplifier circuit. IC401 is an 8-bit MDAC, which is a resistor ladder with a programmable resistance. The control circuit writes an 8-bit word directly to IC401, which has a latch on board to store the word. The word sets the resistance value between pin 15 and pin 1 of IC401. IC402-A forces pin 1 of IC401 to virtual ground. The resistance between pin 1 and pin 16 of IC401, and resistors R404-R406 and VR400 are in the feedback loop of IC402-A. C409 stabilizes this stage. VR400 is a factory gain trim to correct for tolerances in IC401, IC400, and the rest of the analog output circuits.

IC402-A feeds the stage consisting of IC402-B and associated components, which is a programmable de-emphasis filter. JFETs Q400 and Q401 are used to switch C410 and C411, respectively, in or out of the circuit. The state of the JFET switches is set by the outputs of IC305, a digital latch. The latch outputs feed IC407, a quad comparator, which outputs 0V to turn on a FET and –15V to turn off a FET. If neither of the JFETs are on, the circuit is a unity-gain inverting amplifier. The circuit becomes a first-order low-pass filter if one of the JFETs is turned on. If Q400 is on, capacitor C410 is in circuit to create a 75µs time constant. If Q411 is on, capacitor C401 is in circuit to create a 50µs time constant.

IC402-B feeds the stage consisting of IC403-A, IC403-B, IC408-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. IC408-A, R444, R445, R447, and C419 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 L400, L401, 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 32kHz 2200-D 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.

Component-Level Description:

The processed digital output (DIG_OUT) provided at the SAI output port SDO0 (pin 47) of DSP IC706 is received by asynchronous sample rate converter (SRC) IC615 pin 3. An 8.192MHz bit clock is provided from the system clock circuitry to both the DSP and the SRC chips. The DSP output data format is SPI (32 bits per word two words per frame). DSP chip IC706 receives a 128kHz frame clock at its WST input (pin 50) that sets the word transfer rate to eight words per 32kHz period. The SRC receives a 32kHz clock at its L/R*_I input (pin 6). L/R*_I delineates the left and right samples used by the SRC; therefore the SRC uses the first sample received for the left input and the fifth sample for the right input. The DSP output samples are formatted to ensure that the SRC uses a left and right output pair that represent the simultaneously sampled analog input.

The crystal oscillator (Y602) provides the SRC a master clock of 16.384MHz on pin 2. This MCLK frequency allows the output SRC to operate with an output sample rate in the range between 30kHz and 57kHz (operation between 8kHz and 30kHz will result in a one sample delay between the left and right channels). 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 IC609.

The MSDLY_I, BKPOL_I, and TRGLR_I pins of the SRC chip configure the chip for SPI format. 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 IC616, an AES/EBU digital audio transmitter integrated circuit. This interface uses the SPI format with the AES transmitter as master. The transmitter chip encodes the audio data it receives to the AES/EBU interface standard, and transmits it.

The SRC output sample rate and the sample rate that the AES/EBU transmitter transmits with is based on the MCK clock provided to pin 5 of IC616. This clock is received via digital multiplexer chip IC610 which is used to select one of four available clocks. Three 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 128 times the sample rate they represent. They have a frequency stability of ±100PPM. The fourth clock is the EXTMCK clock that is recovered from the 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 sample rate (external sync).

The inter-chip serial data format, the input MCK multiplication factor, and the 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 CS* and RD/WR* control pins (pins 14 and 16) of IC616.

The on-chip RS422 line driver provided by IC616 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. T601 has very low inter-winding capacitance, providing a high level of isolation from high frequency common mode interference.

Composite Output Circuits

This circuitry provides several functions. It interfaces the digital stereo multiplex output from the stereo encoder DSP to a digital-to-analog (D/A) converter, which converts it to an analog signal. The low-pass reconstruction filter removes high frequency images from the D/A converter output and feeds the output buffers. Two output stages with separate level controls buffer the stereo multiplex signal and feed the composite output connectors.

1. Digital-to-Analog (D/A) Converter

The composite D/A is a single chip, 18-bit resistor ladder type. It has a single channel serial input that receives the digital stereo encoded output samples from the DSP.

Component-Level Description:

IC500 is the digital-to-analog converter for the stereo-encoded composite signal. The synchronous serial input interface consists of the bit clock, data and latch enable pins. DSP IC707 provides serial data (COMP_O) to pin 7 of the composite D/A.

An 8.192MHz bit clock is provided from the system clock circuitry to both the DSP and the D/A chips. The DSP output data format is 32-bits per word two words per frame, MSB first (first 24-bits are significant). DSP IC707 receives a 128kHz frame clock at its WST input (pin 50) that sets the word transfer rate to 256kHz. The D/A receives a 256kHz clock at its latch enable (LE) input (pin 6). The D/A uses the last 18-bits received prior to the falling edge of LE (last 18-bits are significant). Flip-flop IC604-A is used to invert and shift the 256kHz system clock to produce an LE signal that has a falling edge aligned with the 18th significant data bit.

Pin 9 is the analog voltage output of IC500. The voltage changes to the current sample value on the falling edge of the 256kHz clock. A full scale output is approximately ±3.0Vpeak, which corresponds to 141% modulation. C517 prevents slew-induced distortion.

2. Analog Reconstruction Filter

The reconstruction filter removes the ultrasonic energy “images” present at the D/A output. It is a passive seventh-order elliptic filter with a cutoff frequency of approximately xx70kHz and xx90dB stopband attenuation above xx203kHz.

Component-Level Description:

The reconstruction filter is a passive seventh-order LC ladder filter, realized by resistors R501, R502 and R504, capacitors C508-C512, C516, and C518, and inductors L500-L502. The frequency response of this filter cannot be measured by applying a swept sine wave at the 2200/2200-D analog inputs. This is because the filter has bandwidth much larger than the analog-to-digital converter. The analog-to-digital converter band-limits the input to 16kHz. Stereo separation is a very sensitive function of the frequency and phase response of this filter in the frequency range of 20-53,000Hz.

The filter is buffered by non-inverting amplifier IC502-A and applied to the output stages. IC501-A is a DC servo to prevent DC from appearing at the composite outputs.

3. Composite Baseband Output Stages

The buffered filter output is applied to two power buffers each capable of driving two 75Ω loads in parallel.

Component-Level Description:

The stereo modulator output is fed into two separate output buffers. The first is made up by IC503-A and IC504. IC504 is a special high slew rate power buffer which is located within the overall amplifier feedback loop. It isolates IC503-A from the destabilizing effects of capacitive loads and also permits 75Ω loads to be driven without degradation. This line driver will drive up to ±1.5Vpeak into 0.047uF in parallel with 37.5Ω before significant nonlinear errors (increases in spurious components as observed on a baseband spectrum analyzer) or linear errors (noticeable deterioration of baseline flatness at 15kHz in the separation test mode) are apparent. Output level is adjusted by varying the feedback resistor VR500. The second output buffer made by IC505-A and IC506 is functionally identical to the one just described.

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, limit, and stereo encode the audio signal. The eight DSP chips, operating at 25 million instructions per second (MIPS) for a total of 200MIPS, provide the necessary signal processing. A 32kHz 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 16.384Mbit/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 2200/2200-D), 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, and extracts the front panel metering data from the DSP chips via this same SHI port.

The left and right analog and digital outputs are sent to the L/R output D/A and the output SRC chip via the SAI port of DSP chip IC706. The last DSP chip (IC707) outputs the composite audio signal on its SAI port where it is directed to the composite D/A.

Component-Level Description:

IC700 thru IC707 are the DSP chips. 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 2.048MHz clock. All DSP chips use their internal PLL to multiply this by 24 to operate the chip's internal oscillator (Fosc) at 49.152MHz. 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 (SS*) (pin 42) of each DSP via latch IC708. The SS* pin is used to enable each of the slaved DSP SPI ports for transfer.

DSP IC700 pins 56 and 57 receive serial stereo 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 communication protocol is SPI with DSP as a slave, and L/R input A/D converter IC312 as master. Left and right data words, each of 32-bit length, constitute a frame. 18 significant bits are received from the analog input A/D and 20 significant bits are received from the digital input SRC. The two serial stereo audio streams are received simultaneously. Both inputs share the same frame clock, L*/R (32kHz) provided to DSP IC700 pin 55 and the same bit clock, SCK (2.048MHz) provided to DSP IC700 pin 51.

Communication between DSP chips IC700 (first) thru 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 SPI communication protocol (two 32-bit words per cycle of the word clock) is used with the DSPs as slaves, and the 2200/2200-D 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 128kHz word clock is provided to the transmit pin 50 and the receive pin 55. An 8.192MHz 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 L/R output D/A and the output SRC from the second to the last DSP chip (IC706). These signals are identical except for any De-Emphasis, J.17 Pre-Emphasis, J.17 Emphasis makeup gain, or output attenuation (DO 100% level) applied to the digital output. The “analog” output is also passed on to the last DSP chip (IC707) for stereo encoding. (“Analog” refers to DSP signal that ultimately gets converted to analog.)

The composite FM stereo signal is output from the last DSP chip (IC707) via its SAI transmitter, formed by DSP IC707 pins 47, 49, and 50. A communication protocol compatible with the composite D/A (IC500) is used with the DSP and D/A as slave and the 2200/2200-D system clocking as master. The serial composite audio bit stream output on pin 47 feeds D/A IC500 pin 7. DSP IC707 pin 50 receives a 128kHz frame clock and pin 49 receives an 8.192MHz bit clock. Two consecutive composite audio data words, each of 32-bit length, constitute a frame.

Power Supply

The power supply converts an AC line voltage input to various power sources used by the 2200/2200-D. Five linear regulators provide ±15VDC and ±5VDC for the analog circuits and +5VDC for the digital circuits. An unregulated voltage powers the LED meters and the LED backlight on the LCD display.

Component-Level Description:

L1 is a power line filter that filters out RFI. F1 is a ½-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 two pairs of secondary windings for stepping down the AC line voltage. The lower voltage pair is configured in parallel, and 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 2200/2200-D. C19 filters the rectified voltage to power the LED backlight on the LCD display, and the LED meters. 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 32kHz pulse at ⅛ 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 2200 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 about 2 Watts 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 2200/2200-D. They also serve as the respective inputs to the voltage regulators IC3 and IC4. These regulators provide the +5VDC and –5VDC analog supplies for the converter chips, which draw only a modest amount of current.

Test points and supply bypass capacitors are placed throughout the PC board. S2 is the ground lift switch used to connect or lift 2200/2200-D 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 ⅛ 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, Beyschlag, Dale, Corning, and Matsushita.
>
> Carbon film resistors have conformally-coated bodies, and are identified by four color bands. They are rated at ¼ watt @ 70°C, ±5%. Orban part numbers 20001-xxx, Manufactured by R-Ohm (R-25), Piher, Beyschlag, 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 ¼ watt, and the 0.140 x 0.375 inch (3.6 x 9.5 mm) size is rated at ½ watt, both ±5% t numbers 2001x-xxx, USA Military Specification MIL-R-11 Style RC-07 (¼ watt) or RC-20 (½ watt). Manufactured by Allen-Bradley, TRW/IRC, and Matsushita.
>
> Cermet trimmer resistors have ⅜-inch (9 mm) square bodies, and are identified by printing on their sides. They are rated at ½ 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 6-33.

REF DES	DESCRIPTION	ORBAN P/N	VEN (1)	VENDOR P/N	ALTERNATE VENDORS (1)	NOTES
MAIN BOARD ASSEMBLY						
Capacitors						
C11-C62	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C65	Met. Polyester, 50V, 5%; 1.0uF	21445-510	PAN	ECQ-V1H105JZ		
C66-C78	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C82-C84	Alum., Radial, 25V, 10%; 10uF	21263-610	NIC	UKLIE101KPAANA		
C85-C90	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C91, C92	Alum., Radial, 25V, 10%; 10uF	21263-610	NIC	UKLIE101KPAANA		
C100, C101	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C102	Alum., Radial, 63V, -20% +100%; 2.2uF	21209-522	SPR	502D 225G063BB1C	PAN	
C103	Ceramic Disc, 100V, 5%; 33pF	21127-033	KEM	C410C330JIG5CA		
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		
C301	Met. Polyester, 50V, 5%; 0.1uF	21445-410	PAN	ECQ-V1H104JZ		
C302-C305	Met. Polyester, 50V, 5%; .0047uF	21445-247	PAN	ECQ-B1H472 F1		
C306, C307	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C308	Tantalum, 10V, 10%; 100uF	21303-710	SPR	196D 107X9010PE4	MANY	
C309, C310	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, C314	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C315	Tantalum, 35V, 10%; 1uF	21307-510	SPR	196D 105X9035HA1	MANY	
C316-C319	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C320	Tantalum, 35V, 10%; 1uF	21307-510	SPR	196D 105X9035HA1	MANY	
C321, C322	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C323-C326	Ceramic, Axial, 100V, 5%; 1000pF	21127-210	KEM	C410C102J1G5CA		
C400	Tantalum, 35V, 10%; 1uF	21307-510	SPR	196D 105X9035HA1	MANY	
C401, C402	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C403	Tantalum, 35V, 10%; 1uF	21307-510	SPR	196D 105X9035HA1	MANY	
C404, C405	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C406	Tantalum, 20V, 10%; 10uF	21305-610	SPR	196D 106X9020JA1	MANY	
C407, C408	Met. Polyester, 50V, 5%; 0.01uF	21445-310	PAN	ECQ-V1H103JZ		
C409	Ceramic Disc, 100V, 5%; 33pF	21127-033	KEM	C410C330JIG5CA		

FOOTNOTES:
(1) See page 6-36 for Vendor abbreviations
(2) No Alternate Vendors known at publication
(3) Actual part is specially selected from part listed, consult Factory
(4) Realignment may be required if replaced, see Circuit Description and/or Alignment Instructions

SPECIFICATIONS AND SOURCES FOR REPLACEMENT PARTS

OPTIMOD-FM 2200

Main Board Assembly - Capacitors.

REF DES	DESCRIPTION	ORBAN P/N	VEN (1)	VENDOR P/N	ALTERNATE VENDORS (1)	NOTES
Capacitors (continued)						
C410, C411	Mica, 500V, 1%; 1500pF	21022-215	CD	CD19-FD152F03	SAN	
C412	Ceramic, Axial, 100V, 5%; 10pF	21127-010	KEM	C410C100J1G5CA		
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, C417	Mica, 500V, 1%; 1500pF	21022-215	CD	CD19-FD152F03	SAN	
C418	Ceramic, Axial, 100V, 5%; 10pF	21127-010	KEM	C410C100J1G5CA		
C419, C420	Met. Polyester, 50V, 5%; 0.1uF	21445-410	PAN	ECQ-V1H104JZ		
C500	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C501	Tantalum, 35V, 10%; 1uF	21307-510	SPR	196D 105X9035HA1	MANY	
C503, C504	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C505	Tantalum, 35V, 10%; 1uF	21307-510	SPR	196D 105X9035HA1	MANY	
C506	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C507	Tantalum, 35V, 10%; 1uF	21307-510	SPR	196D 105X9035HA1	MANY	
C508	Mica, 1500V, 5%; 390pF	21018-139	CD	CD15-FD391F03	SAN	
C509	Mica, 500V +1/2pF -1/2pF; 15pF	21017-015	CD	CD15-CD150D03	SAN	
C510	Mica, 500V, 1%; 820pF	21022-182	CD	CD19-FD821F03	SAN	
C511	Mica, 500V, 1%; 620pF	21022-162	CD	CD19FD621F03		
C512	Mica, 500V, 1%; 220pF	21018-122	CD	CD15-FD221F03	SAN	
C513, C514	Mica, 500V, 1%; 51pF	21018-051	CD	CD15ED510F03		
C515	Ceramic Disc, 100V, 5%; 33pF	21127-033	KEM	C410C330JIG5CA		
C516	Mica, 500V, 1%; 51pF	21018-051	CD	CD15ED510F03		
C517	Ceramic Disc, 100V, 5%; 68pF	21127-068	KEM	C410C680JICG5CA		
C518	Mica, 500V, +1/2pF -1/2pF; 33pF	21017-033	CD	CD15-CD330D03	SAN	
C519	Ceramic Disc, 100V, 5%; 33pF	21127-033	KEM	C410C330JIG5CA		
C520	Met. Polyester, 50V, 5%; 0.1uF	21445-410	PAN	ECQ-V1H104JZ		
C521, C522	Mica, 500V, +1/2pF -1/2pF; 22pF	21017-022	CD	CD15-CD220D03	SAN	
C523	Ceramic, Axial, 100V, 5%; 680pF	21127-168	KEM	C410C681J1G5CA		
C524	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C525	Ceramic, Axial, 100V, 5%; 680pF	21127-168	KEM	C410C681J1G5CA		
C600	Ceramic Disc, 100V, 5%; 33pF	21127-033	KEM	C410C330JIG5CA		
C601	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C602	Met. Polyester, 50V, 5%; .047uF	21445-347	PAN	ECQ-V1H473JZ		
C603	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		

FOOTNOTES:
(1) See page 6-36 for Vendor abbreviations
(2) No Alternate Vendors known at publication
(3) Actual part is specially selected from part listed, consult Factory
(4) Realignment may be required if replaced, see Circuit Description and/or Alignment Instructions

SPECIFICATIONS AND SOURCES FOR REPLACEMENT PARTS

OPTIMOD-FM 2200

Main Board Assembly - Capacitors.

REF DES	DESCRIPTION	ORBAN P/N	VEN (1)	VENDOR P/N	ALTERNATE VENDORS (1)	NOTES
Capacitors (continued)						
C604	Ceramic Disc, 100V, 5%; 150pF	21127-115	KEM	C410C151JIG5CA		
C606	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C607	Alum., Radial, 25V, 10%; 100uF	21263-710	NIC	UKLIE101KPAANA		
C608	Ceramic Disc, 100V, 5%; 33pF	21127-033	KEM	C410C330JIG5CA		
C609, C611	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C612, C613	Ceramic Disc, 100V, 5%; 33pF	21127-033	KEM	C410C330JIG5CA		
C800	Met. Polyester, 50V, 5%; 0.01uF	21445-310	PAN	ECQ-V1H103JZ		
C801	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C802	Met. Polyester, 50V, 5%; 0.01uF	21445-310	PAN	ECQ-V1H103JZ		
C803	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C804	Met. Polyester, 50V, 5%; 0.01uF	21445-310	PAN	ECQ-V1H103JZ		
C805	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C806	Met. Polyester, 50V, 5%; 0.01uF	21445-310	PAN	ECQ-V1H103JZ		
C807	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C808	Met. Polyester, 50V, 5%; 0.01uF	21445-310	PAN	ECQ-V1H103JZ		
C809	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C810	Met. Polyester, 50V, 5%; 0.01uF	21445-310	PAN	ECQ-V1H103JZ		
C811	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C812	Met. Polyester, 50V, 5%; 0.01uF	21445-310	PAN	ECQ-V1H103JZ		
C813	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
C814	Met. Polyester, 50V, 5%; 0.01uF	21445-310	PAN	ECQ-V1H103JZ		
C815	Ceramic, 50V, 20%; 1uF	21131-410	MUR	GRM42-6Z5U104M50BD		
Diodes						
CR102-CR109	Diode, Rectifier, 400V, 1A	22201-400	MOT	1N4004	MANY	
CR300-C307	Diode, Signal, Hot Carrier	22102-001	HP	HP5082-2800	MANY	
Inductors						
L100	Inductor, RF Choke; 7uH	29501-004	OHM	Z-50	(2)	
L300	Filter, EMI, W/BEAD, 50V,1000PF	29508-210	TAI	STB102KB		
L301	Inductor, RF Choke; 1.2mH	29503-000	MIL	73F123AF		
L302	Filter, EMI, W/BEAD, 50V,1000PF	29508-210	TAI	STB102KB		
L303	Inductor, RF Choke; 1.2mH	29503-000	MIL	73F123AF		
L304	Filter, EMI, W/BEAD, 50V,1000PF	29508-210	TAI	STB102KB		

FOOTNOTES:
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SPECIFICATIONS AND SOURCES FOR REPLACEMENT PARTS

OPTIMOD-FM 2200

Main Board Assembly - Capacitors, Diodes, Inductors.

REF DES	DESCRIPTION	ORBAN P/N	VEN (1)	VENDOR P/N	ALTERNATE VENDORS (1)	NOTES
Inductors (continued)						
L305	Inductor, RF Choke; 1.2mH	29503-000	MIL	73F123AF		
L306	Filter, EMI, W/BEAD, 50V,1000PF	29508-210	TAI	STB102KB		
L307	Inductor, RF Choke; 1.2mH	29503-000	MIL	73F123AF		
L400, L401	Inductor, RF Choke; 7uH	29501-004	OHM	Z-50	(2)	
L402, L403	Filter, EMI, W/BEAD, 50V,1000PF	29508-210	TAI	STB102KB		
L404, L405	Inductor, RF Choke; 7uH	29501-004	OHM	Z-50	(2)	
L406, L407	Filter, EMI, W/BEAD, 50V,1000PF	29508-210	TAI	STB102KB		
L500	Inductor, Variable	29705-008	ORB			
L501	Inductor, Variable	29705-012	ORB			
L502	Inductor, Variable	29705-009	ORB			
L600-L603	Filter, EMI, W/BEAD, 50V,1000PF	29508-210	TAI	STB102KB		
L802-L804	Inductor, RF Choke; 7uH	29501-004	OHM	Z-50	(2)	
Integrated Circuits						
IC100	Digital, Microprocessor	24822-000	ZI	Z8018010VSC		
IC101-IC104	Address Decoder	24899-000	MOT	MC74AC138D		
IC105	Digital, EPROM	24829-000	TI	TMS27C010A-10JL		
IC106	Digital, SRAM	24817-000	TOS	TC55257CPL-10		
IC107	Serial EEPROM	24898-000	NAT	NM25C04M8		
IC109	Digital, Inverter	24900-000	TI	SN74HC14AD		
IC110-IC117	Optoisolator, NPN	25003-000	SIE	SFH-601-1		
IC118, IC119	Digital, Inverter	24900-000	TI	SN74HC14AD		
IC120	Digital, Transceiver	24851-000	SIG	74HC245D		
IC122	Power Monitor/Watchdog	24872-000	MAX	1232CPA		
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		
IC304	Digital, Octal Buffer	24902-000	MOT	MC74HC14AD		
IC305	Digital, Latch	24857-000	MOT	MC74HC374ADW		
IC306	Quad Comparator	24710-302	NAT	LM339		
IC307	Quad SPST Switches	24728-302	AD	ADG222		
IC308	Audio Preamp	24727-402	AD	SSM-2017P		

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SPECIFICATIONS AND SOURCES FOR REPLACEMENT PARTS

OPTIMOD-FM 2200

Main Board Assembly - Inductors, Integrated Circuits.

REF DES	DESCRIPTION	ORBAN P/N	VEN (1)	VENDOR P/N	ALTERNATE VENDORS (1)	NOTES
Integrated Circuits (continued)						
IC309	Quad SPST Switches	24728-302	AD	ADG222		
IC310	Linear, Dual Opamp	24209-202	NAT	LF412CN		
IC311	Linear, Dual Opamp	24207-202	SIG	NE5532N	TI,EXR	
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		
IC401	Digital, Multiplying DAC	24714-302	AD	AD7524JN		
IC402	Linear, Dual Opamp	24209-202	NAT	LF412CN		
IC403	Linear, Dual Opamp	24207-202	SIG	NE5532N	TI,EXR	
IC404	Digital, Multiplying DAC	24714-302	AD	AD7524JN		
IC405	Linear, Dual Opamp	24209-202	NAT	LF412CN		
IC406	Linear, Dual Opamp	24207-202	SIG	NE5532N	TI,EXR	
IC407	Quad Comparator	24710-302	NAT	LM339		
IC408	Linear, Dual Opamp	24209-202	NAT	LF412CN		
IC500	Digital, D/A Converter	24742-000	AD	AD1861N		
IC501	Linear, Dual Opamp	24209-202	NAT	LF412CN		
IC502, IC503	Linear, Single Opamp	24008-202	TI	LM318N	NAT	
IC504	Power Buffer	24707-102	LT	LT1010CH		
IC505	Linear, Single Opamp	24008-202	TI	LM318N	NAT	
IC506	Power Buffer	24707-102	LT	LT1010CH		
IC600	Digital, AES Receiver	24847-000	CSC	CS8412CS		
IC601	Digital, Transceiver	24851-000	SIG	74HC245D		
IC602	Digital, NAND Gate	24853-000	MOT	MC74HC00AD		
IC603	Digital, Sample Rate Conveter	24733-000	AD	AD1890JP		
IC604	Digital, Flip-Flop	24858-000	TI	SN74HC74D		
IC605	Digital, Inverter	24900-000	TI	SN74HC14AD		
IC606	Digital, AND Gate	24850-000	MOT	MC74HC08AD		
IC609	Digital, Latch	24857-000	MOT	MC74HC374ADW		
IC610	Digital, Multiplexer	24896-000	MOT	MC74HC153D		
IC611	Digital, Latch	24857-000	MOT	MC74HC374ADW		
IC612	Digital, Flip-Flop	24858-000	TI	SN74HC74D		
IC613	Digital, PAL	44032-100	ORB			
IC614	Digital, AND Gate	24850-000	MOT	MC74HC08AD		

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SPECIFICATIONS AND SOURCES FOR REPLACEMENT PARTS

OPTIMOD-FM 2200
Main Board Assembly - Integrated Circuits.

REF DES	DESCRIPTION	ORBAN P/N	VEN (1)	VENDOR P/N	ALTERNATE VENDORS (1)	NOTES
Integrated Circuits (continued)						
IC615	Digital, Sample Rate Conveter	24733-000	AD	AD1890JP		
IC616	Digital, Audio Interface	24644-000	CRY	CS8401ACP		
IC618	PAL	44031-100	ORB			
IC619	Digital, PLL	24901-000	SIG	74HC4046AD		
IC700-IC707	Digital, DSP	24897-000	MOT	DSP56004FJ50		
IC708, IC709	Digital, Latch	24857-000	MOT	MC74HC374ADW		
IC710	Digital, AND Gate	24850-000	MOT	MC74HC08AD		
Resistors						
R100	Resistor, 1/4W; 0 OHM (Jumper)	20020-025	ROHM	JPW-02A		
R106, R124	Resistor Network, SIP; 100K	20221-101	BEK	L10-1C104		
R127	Resistors, MF, 1/2W, 1%; 300 OHM	20080-300	DAL	RNC55J		
R349	Resistor, 1/4W; 0 OHM (Jumper)	20020-025	ROHM	JPW-02A		
R411	Resistors, MF, 1/8W, 0.1%, 13.3K	20059-133	SEI	RN 1/8 13.3K .1%		
R412	Resistors, MF, 1/8W, 0.1%, 10.2K	20059-102	SEI	RN 1/8 10.2K .1%		
R413	Resistors, MF, 1/8W, 0.1%; 15.4 OHM	20056-154	DAL	RNC55J		
R415	Resistors, MF, 1/8W, 0.1%, 4.64K	20058-464	SEI	RN 1/8 4.64K .1%		
R416	Resistors, MF, 1/8W, 0.1%, 4.53K	20058-453	SEI	RN 1/8 4.53K .1%		
R417	Resistors, MF, 1/8W, 0.1%, 4.53K	20058-453	SEI	RN 1/8 4.53K .1%		
R418	Resistors, MF, 1/8W, 0.1%; 3.01K	20058-301	SEI	RN 1/8 3.01K .1%		
R419	Resistors, MF, 1/8W, 0.1%, 4.64K	20058-464	SEI	RN 1/8 4.64K .1%		
R420	Resistors, MF, 1/8W, 0.1%; 15.4 OHM	20056-154	DAL	RNC55J		
R421	Resistors, MF, 1/8W, 0.1%, 13.3K	20059-133	SEI	RN 1/8 13.3K .1%		
R422	Resistors, MF, 1/8W, 0.1%, 10.2K	20059-102	SEI	RN 1/8 10.2K .1%		
R430	Resistors, MF, 1/8W, 0.1%, 13.3K	20059-133	SEI	RN 1/8 13.3K .1%		
R431	Resistors, MF, 1/8W, 0.1%, 10.2K	20059-102	SEI	RN 1/8 10.2K .1%		
R432	Resistors, MF, 1/8W, 0.1%; 15.4 OHM	20056-154	DAL	RNC55J		
R435	Resistors, MF, 1/8W, 0.1%, 4.64K	20058-464	SEI	RN 1/8 4.64K .1%		
R436, R437	Resistors, MF, 1/8W, 0.1%, 4.53K	20058-453	SEI	RN 1/8 4.53K .1%		
R438	Resistors, MF, 1/8W, 0.1%; 3.01K	20058-301	SEI	RN 1/8 3.01K .1%		
R439	Resistors, MF, 1/8W, 0.1%, 4.64K	20058-464	SEI	RN 1/8 4.64K .1%		
R440	Resistors, MF, 1/8W, 0.1%; 15.4 OHM	20056-154	DAL	RNC55J		
R441	Resistors, MF, 1/8W, 0.1%, 13.3K	20059-133	SEI	RN 1/8 13.3K .1%		

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SPECIFICATIONS AND SOURCES FOR REPLACEMENT PARTS

OPTIMOD-FM 2200

Main Board Assembly - Integrated Circuits, Resistors.

REF DES	DESCRIPTION	ORBAN P/N	VEN (1)	VENDOR P/N	ALTERNATE VENDORS (1)	NOTES
Resistors (continued)						
R442	Resistors, MF, 1/8W, 0.1%, 10.2K	20059-102	SEI	RN 1/8 10.2K .1%		
R501	Resistors, MF, 1/8W, 0.1%; 3.57K	20058-357	DAL	RNC55J		
R502	Resistors, MF, 1/8W, 0.1%; 300OHM	20058-475	DAL	RNC55J		
R602, R603	Resistor, 1/4W; 0 OHM (Jumper)	20020-025	ROHM	JPW-02A		
VR500, VR501	Trimpot, Cermet, 20 Turn; 25K	20512-325	BEK	89PR25K	BRN	
Switches						
S209	Switch, Rotary, Encoder	26081-000	ORB			
Transistors						
Q300-Q303	Transistor, JFET/N	23402-101	NAT	J108		
Q400-Q403	Transistor, JFET/N	23406-101	NAT	J113	SIL	
Miscellaneous						
J101	Connector, D Type, 25-pin	27017-025	AD	JMDF-25S		
J300, J304	Connector, XLR, PC Mount, Female	27054-003	NEU	NC 3 FD-H		
J400, J401	Connector, XLR, PC Mount, Male	27053-003	NEU	NC 3 MD-H		
J600	Connector, XLR, PC Mount, Female	27054-003	NEU	NC 3 FD-H		
J601	Connector, XLR, PC Mount, Male	27053-003	NEU	NC 3 MD-H		
T600, T601	Transformer, Surface Mount	29015-000	ORB			
Y601	Crystal; 11.2896MHz	28071-000	ORB			
Y602	Oscillator; 16.384MHz	28074-001	ORB			

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SPECIFICATIONS AND SOURCES FOR REPLACEMENT PARTS

OPTIMOD-FM 2200

Main Board Assembly - Resistors, Swirtches, transistors Miscellaneous.

REF DES	DESCRIPTION	ORBAN P/N	VEN (1)	VENDOR P/N	ALTERNATE VENDORS (1)	NOTES
	POWER SUPPLY BOARD ASSEMBLY					
	Capacitors					
	Alum, Radial, 16V; 6800uF	~~21255-000~~	PAN	ECOS1CA682AA		
	Alum, Radial, 35V; 1000uF	~~21256-000~~	PAN	ECEA1VGE102		
	Ceramic, Axial, 50V, 20%; 0.1uF	~~21129-410~~	KEM	C410C104M5UCA		
	Alum, Radial, 25V; 47uF	~~21206-747~~	PAN	ECEAIEU471		
	Alum., Radial, 25V, 10%; 100uF	~~21263-710~~	NIC	UKLIE101KPAANA		
	Diode					
	Diode, Rectifier	~~22015-000~~	TAT	SBL-1630CT		
	Diode, Zener, 1W, 5%; 5.6V	~~22004-056~~	MOT	1N4734A		
	Diode, Rectifier, 400V, 1A	~~22201-400~~	MOT	1N4004	MANY	
	Integrated Circuits					
	Regulator, 5V	~~24321-000~~	LT	LT1086CK-5		
	D.C. Regulator, 15V Negative	~~24303-901~~	NAT	LM79M15AUC	TI,MOT	
	D.C. Regulator, 15V Positive	~~24304-901~~	NAT	LM78M15UC	TI,MOT	
	D.C. Regulator, 5V Positive	~~24307-901~~	NAT	LM78M05C	TI,MOT	
	D.C. Regulator, 5V Negative	~~24308~~-901	NAT	LM79M05C	TI,MOT	
	Miscellaneous					
	Transformer, Power	* 55033-000	ORB			
	Varistor	~~22500-271~~	PAN	ERZ-C10DK271U		
	Switch, Slide, Mains voltage selector	~~26148-000~~	SW	EPS2-PC3		
	Switch, Slide, SPDT	~~26146-000~~	ECG	SSP1-S1-M7-Q-E-A		
	Filter, Line	~~28012-000~~	DEL	03ME1		
	Fuseholder, PC Mount	~~28112-001~~	LFE	345-101-01		
	Insulator, Silicone, Thermal (TO-3)	~~15023-001~~	CHM	60-11-4511-1674	Chomerics, Inc.	
	Fuse, 3AG, Slo-Blo, 1/2A	~~28004-150~~	LFE	313.500	BUS	
	Resistor, CF, 1/2W, 5%; 2.0 OHM	~~20021-920~~	ORB			
	Transistor, Power, NPN	~~[illegible]-201~~	TI	TIP120		
	Transistor, Signal, NPN	~~23202-101~~	MOT	2N4400	FSC	

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SPECIFICATIONS AND SOURCES FOR REPLACEMENT PARTS

OPTIMOD-FM 2200

Power Supply Board Assembly - Capacitors, Diodes, Integrated Circuits, Miscellaneous.

REF DES	DESCRIPTION	ORBAN P/N	VEN (1)	VENDOR P/N	ALTERNATE VENDORS (1)	NOTES
FINAL ASSEMBLY						
Miscellaneous						
	Fuse, 3AG, Slo-Blo, 1/2A	~~28004-150~~	LFE	313.500	BUS	

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SPECIFICATIONS AND SOURCES FOR REPLACEMENT PARTS

OPTIMOD-FM 2200

Final Assembly - Miscellaneous.

Vendor Codes

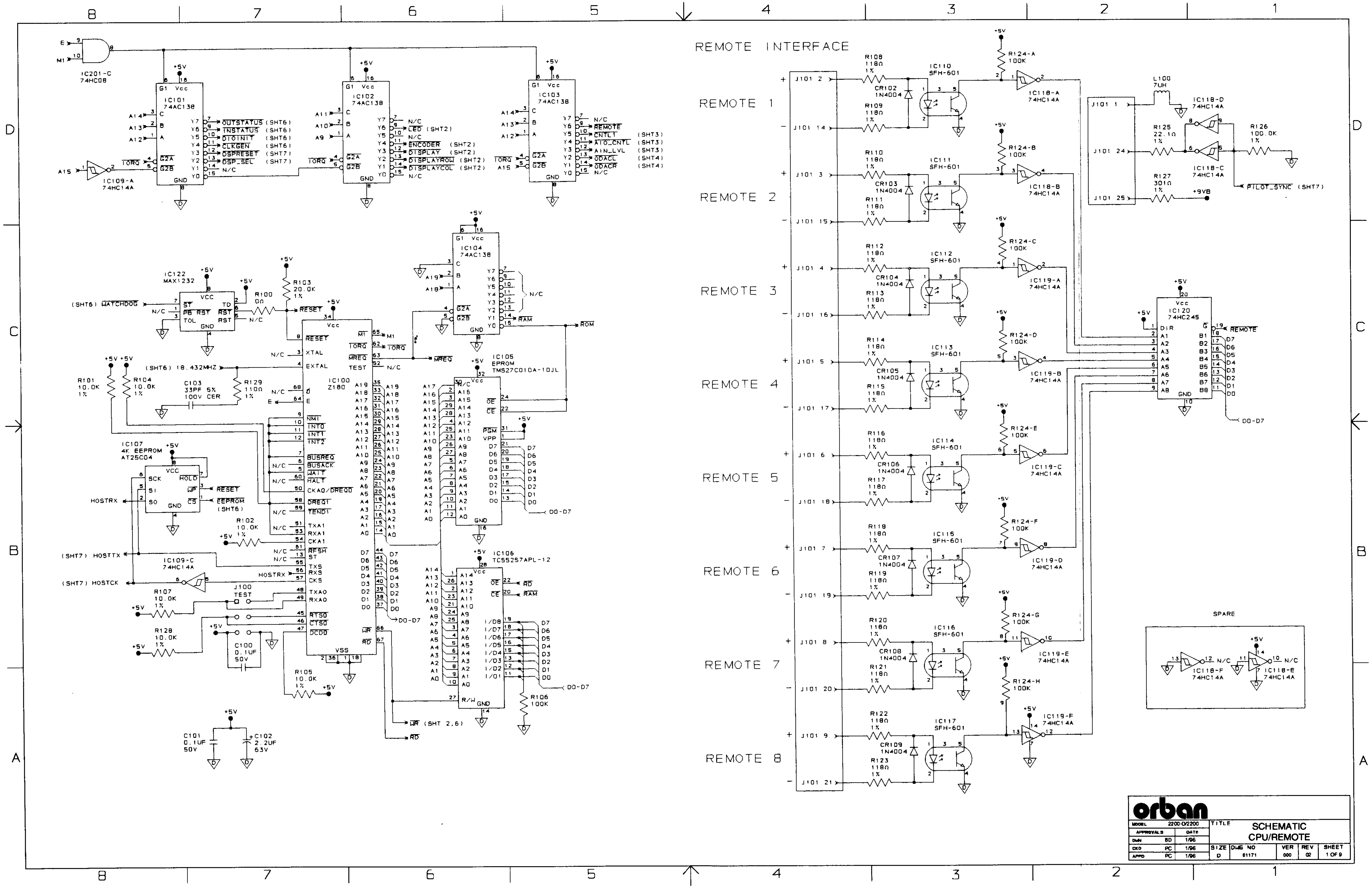
AB	Rockwell Allen-Bradley 625 Liberty Ave Pittsburgh, PA 15222-3123
AD	Analog Devices, Inc. One Technology Way PO Box 9106 Norwood, MA 02062-9106
AKG	AKG Acoustics, Inc. See Orban
AM	Amphenol Corporation 358 Hall Avenue Wallingford, CT 06492
BEK	Beckman Industrial Corporation 4141 Palm Street Fullerton, CA 92635-1025
BEL	Belden Electronic Wire & Cable PO Box 1980 Richmond, IN 47374
BRN	Bourns, Inc Resistive Components Group 1200 Columbia Avenue Riverside, CA 92507
BUS	Bussmann Division Cooper Industries PO Box 14460 St. Louis, MO 63178
CD	Cornell-Dubilier Electronics 1700 Rte. 23 North Wayne, NJ 07470
CRL	Mepcopal/Centralab See Mepcopal
CSC	Crystal Semiconductor Corporation 4210-T. South Industrial Dr. Austin, TX 78744
CTS	CTS Corporation 907 North West Blvd. Elkhart, IN 46514
CW	CW Industries 130 James Way Southampton, PA 18966
DBX	dbx A Harman International Company 8760 South Sandy Parkway Sandy, UT 84107
DEL	Delta Products Corp 3225 Laurel View Ct. Fremont, CA 94538
DUR	Duracell, Inc. Berkshire Industrial Park Bethel, CT 06801
ELSW	Electro Switch 77 King Avenue Weymouth, MA 02188
EMI	Crompton Modutec 920 Candia Rd. Manchester, NH 03109
EXR	Exar Corporation 2222 Qume Dr. PO Box 49007 San Jose, CA 95161-9007
FR	Fair-Rite Products Corp. PO Box J Wallkill, NY 12589
FSC	Fairchild Camera & Instr. Corp. See National Semiconductor
GI	General Instruments Optoelectronics Division See Quality Technologies
HA	Harris Semiconductor 1301 Woody Burke Rd. Melbourne, FL 32901
HO	Hoyt Elect. Inst. Works 19 Linden St. Penacook, NH 03303
HP	Hewlett-Packard Co. Components Group 640 Page Mill Road Palo Alto, CA 94304
INS	Intersil, Inc. See Harris Semiconductor
ITW	ITW Switches An Illinois Tool Works Co. 6615 W. Irving Park Rd. Dept. T Chicago, IL 60634
KEM	KEMET Electronics Corporation Post Office Box 5928 Greenville, South Carolina 29606
KEY	Keystone Electronics Corp. 31-07 20th Rd. Astoria, NY 11105
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
LUMX	Lumex Opto/Components Inc. 292 E. Hellen Road Palatine, IL 60067
MAL	Mallory Capacitor Co. 7545 Rockville Rd. PO Box 1284 Indianapolis, IN 46241
MAR	Marquardt Switches, Inc. 2711-TR Route 20 East Cazenovia, NY 13035
MAT	Matsushita Electric Corp of America One Panasonic Way Secaucus, NJ 07094
ME	Mepcopal/Centralab A North American Phillips Corp. 11468 Sorrento Valley Road San Diego, CA 92121
MID	Hollingsworth/Wearnes 1601 N. Powerline Rd. Pampano, FL 33069
MIL	J.W. Miller Division Bell Industries 306 E. Alondra Gardena, CA 90247
MOT	Motorola Semiconductor PO Box 20912 Phoenix, AZ 85036
MUR	Murata Erie North America 2200 Lake Park Drive Smyrna, GA 30080
NAT	National Semiconductor Corp. 2900 Semiconductor Drive PO Box 58090 Santa Clara, CA 95051
NEL	Crystal Biotech 75 South Street Hopkinton, MA 01748
NOB	Noble U.S.A., Incorporated 5450 Meadowbrook Industrial Ct. Rolling Meadows, IL 60008
OKI	OKI Semiconductor 785 N. Mary Ave. Sunnyvale, CA 94086-2909
OHM	Ohmite Manufactoring Company 3601 Howard Street Skokie, IL 60076
ORB	Orban A Harman International Company 1525 Alvarado Street San Leandro, CA 94577
PAN	Panasonic Industrial Company Two Panasonic Way 7E-2T Secaucus, NJ 07094
QT	Quality Technologies, Inc. 610 North Mary Ave. Sunnyvale, CA 94086
RAL	Raltron Electronics Corp. 2315 NW 107th Ave. Miami, FL 33172
RAY	Raytheon Company Semiconductor Division 350 Ellis Street Mountain View, CA 94039
RCA	RCA Solid State See Harris Semiconductor
ROHM	Rohm Electronics 3034 Owens Dr. Antioch, TENN 37013
SAE	Stanford Applied Engineering, Inc 340 Martin Avenue Santa Clara, CA 95050
SAN	Sangamo Weston Inc. Capacitor Division See Cornell-Dubilier
SCH	ITT Schadow, Inc. 8081 Wallace Road Eden Prairie, MN 55344
SIE	Siemens Components Inc. Heimann Systems Div. 186 Wood Avenue South Iselin, NJ 08830
SIG	Philips Components - Signetics North American Phillips Corp. 811 E. Arques Sunnyvale, CA 94088
SPR	Sprague Electric Co. 41 Hampden Road PO Box 9102 Manifold, MA 02048-9102
S.W.	Seitchcraft A Raytheon Company 5555 N. Elation Avenue Chicago, IL 60630
AT	Taiga America, Inc. 700 Frontier Way Bensenville, IL 60106
TDK	TDK Electronics Corporation 12 Harbor Park Port Washington, NY 11050
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
VARO	Micro Quality Semiconductor, Inc. PO Box 469013 Garland, TX 75046-9013
WES	Westlake See Mallory Capacitor Co.
WIM	Wima Division 2269 Saw Mill Rd. Building 4C PO Box 217 Elmsford, NY 10533
ZI	ZILOG Inc. 210 Hacienda Ave. Campbell, CA 95008

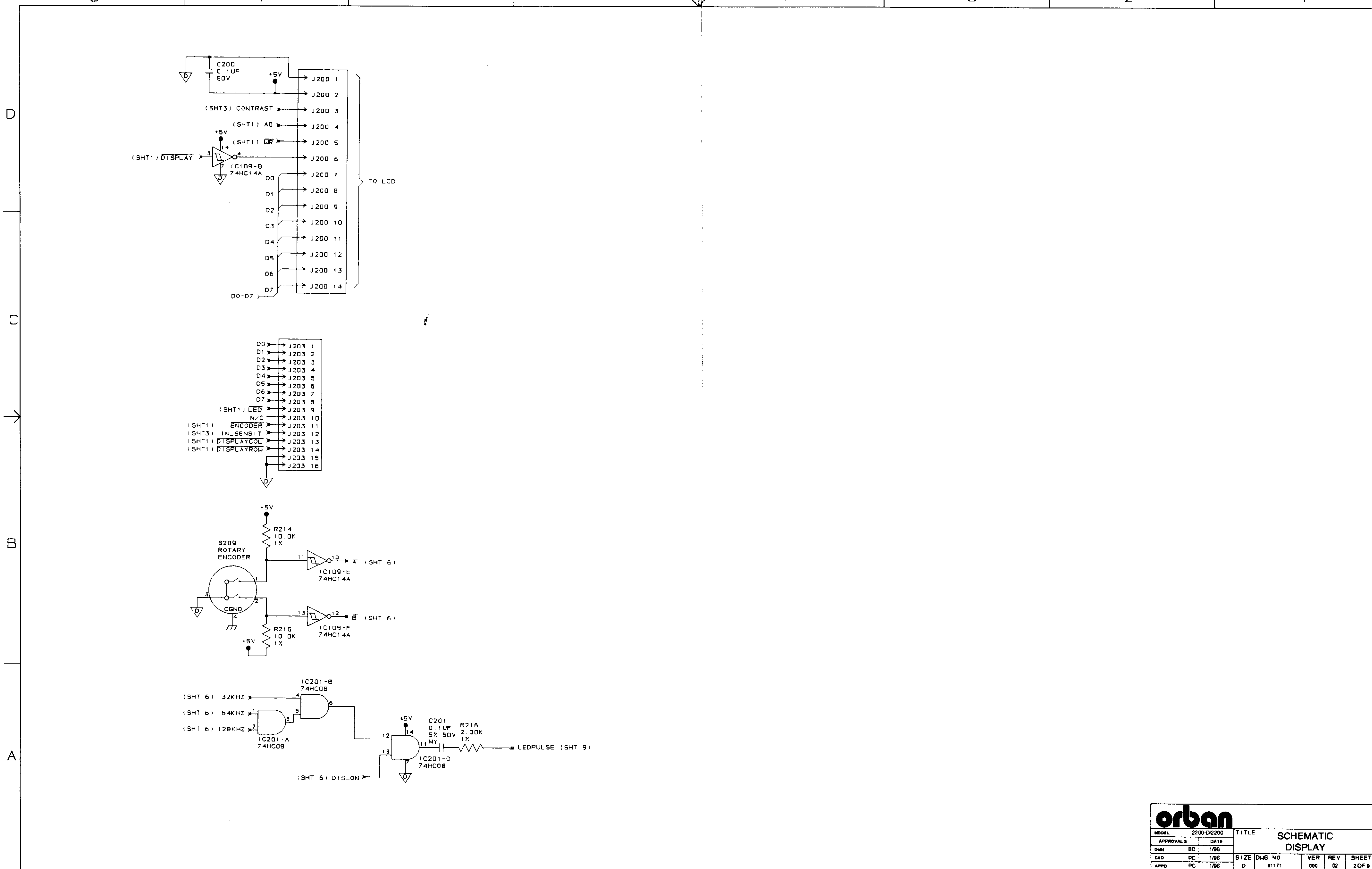
Schematics, Assembly Drawings

The following drawings are included in this manual:

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.

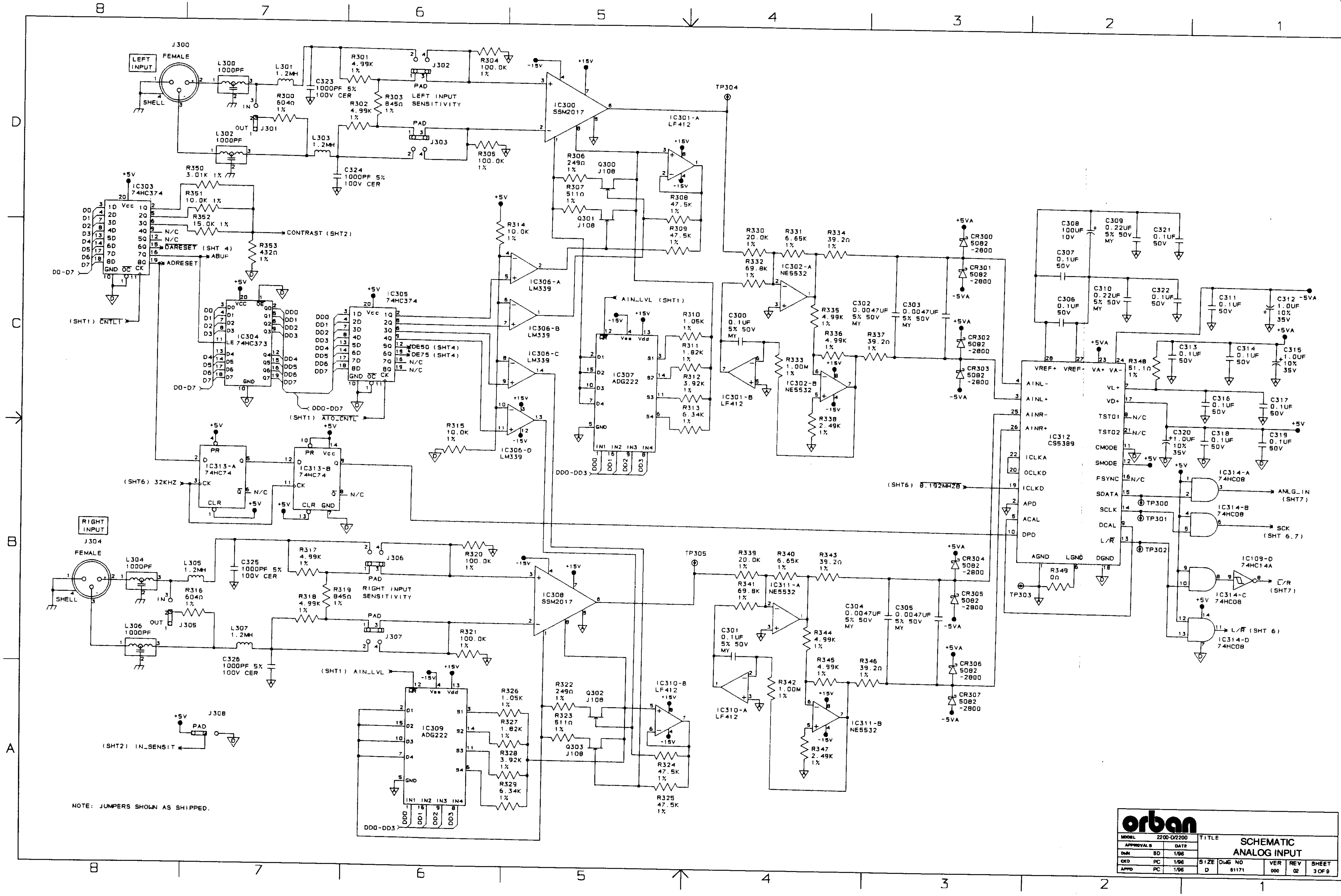
If you intend to replace parts, please read page 6-22

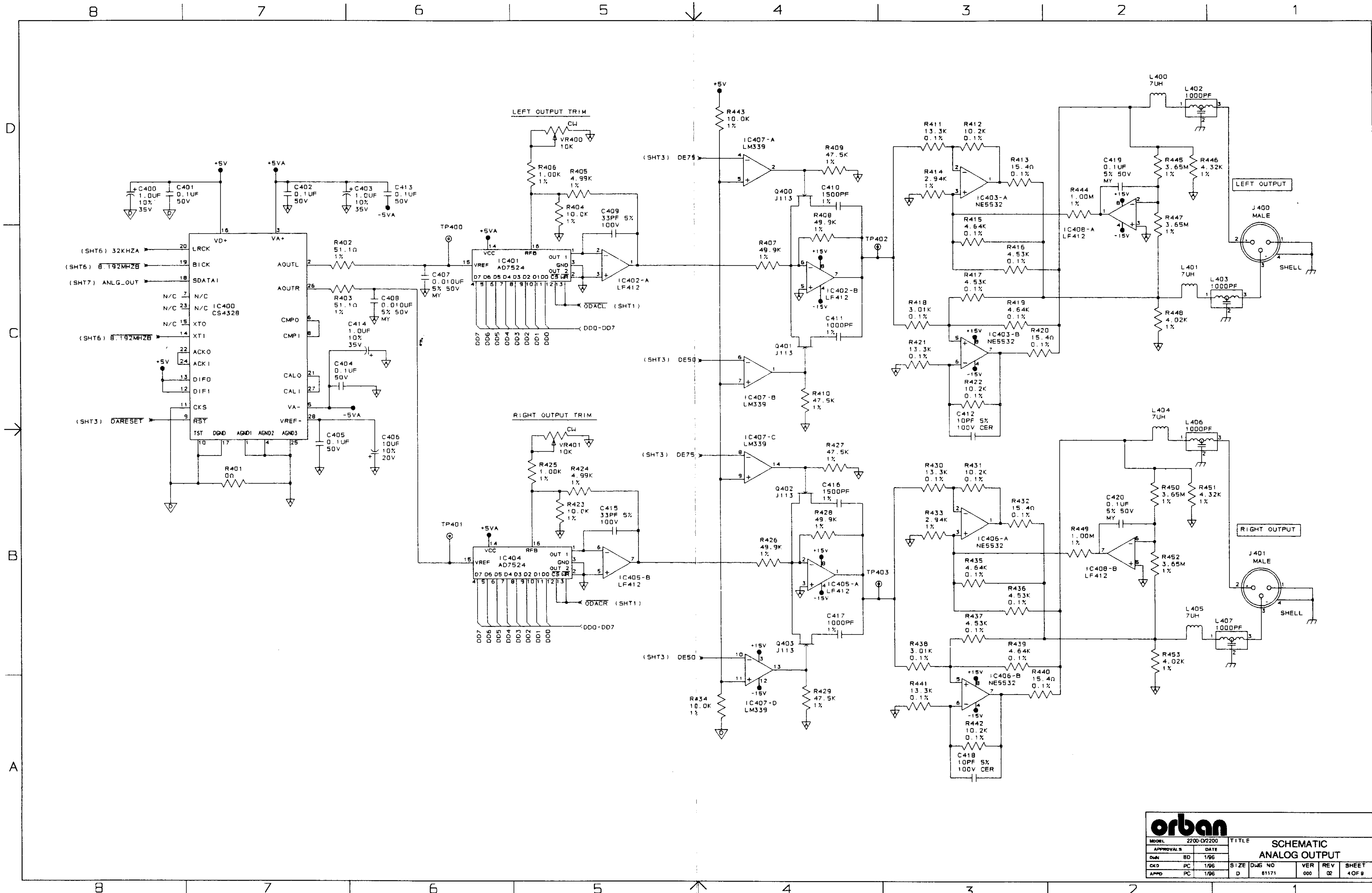

REMOTE INTERFACE
REMOTE 1
REMOTE 2
REMOTE 3
REMOTE 4
REMOTE 5
REMOTE 6
REMOTE 7
REMOTE 8
IC201-C 74HC08
IC101 74AC138
IC102 74AC138
IC103 74AC138
IC104 74AC138
IC109-A 74HC14A
IC122 MAX1232
IC100 Z180
IC105 EPROM TMS27C010A-10JL
IC106 TC55257APL-12
IC107 4K EEPROM AT25C04
IC109-C 74HC14A
IC120 74HC245
IC110 SFH-601
IC111 SFH-601
IC112 SFH-601
IC113 SFH-601
IC114 SFH-601
IC115 SFH-601
IC116 SFH-601
IC117 SFH-601
IC118-A 74HC14A
IC118-B 74HC14A
IC118-C 74HC14A
IC118-D 74HC14A
IC119-A 74HC14A
IC119-B 74HC14A
IC119-C 74HC14A
IC119-D 74HC14A
IC119-E 74HC14A
IC119-F 74HC14A
SPARE
IC118-F 74HC14A
IC118-E 74HC14A
PILOT_SYNC (SHT7)
(SHT6) 18.432MHZ
(SHT6) WATCHDOG
(SHT7) HOSTTX
(SHT7) HOSTCK
HOSTRX
J100 TEST
D0-D7
orban
MODEL 2200-D/2200
TITLE SCHEMATIC CPU/REMOTE
SIZE D
DWG NO 61171
VER 000
REV 02
SHEET 1 OF 9

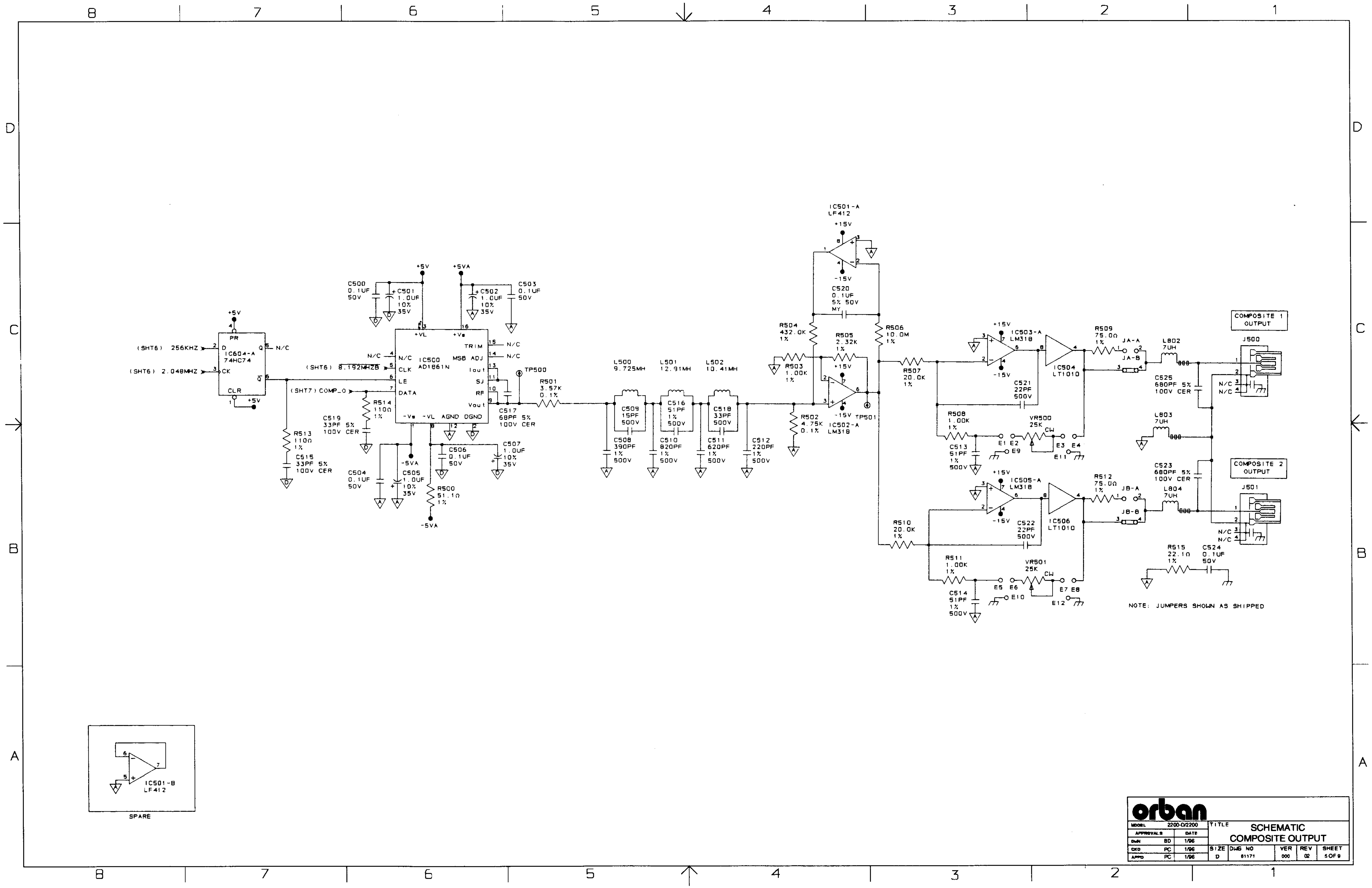


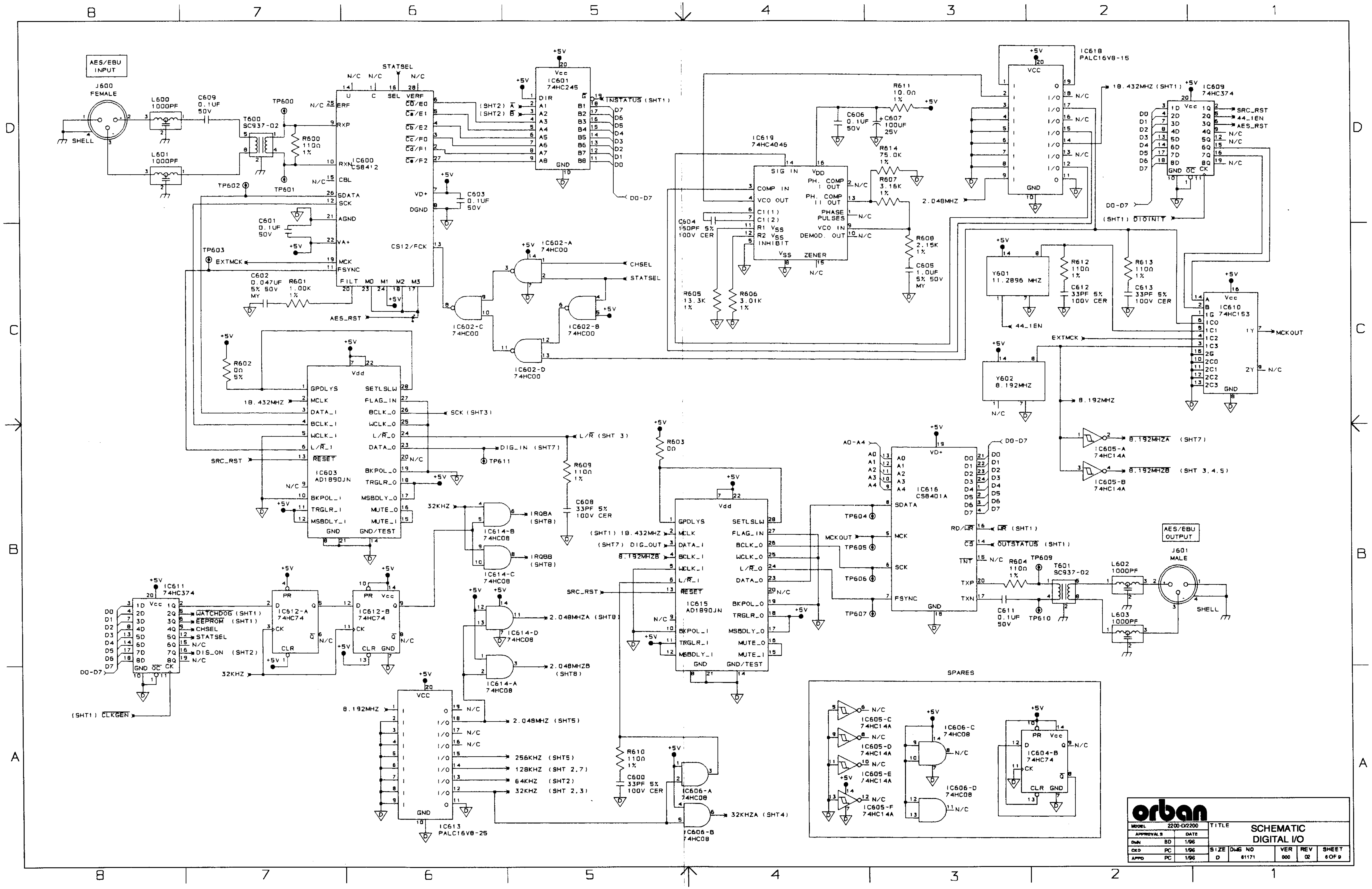
orban

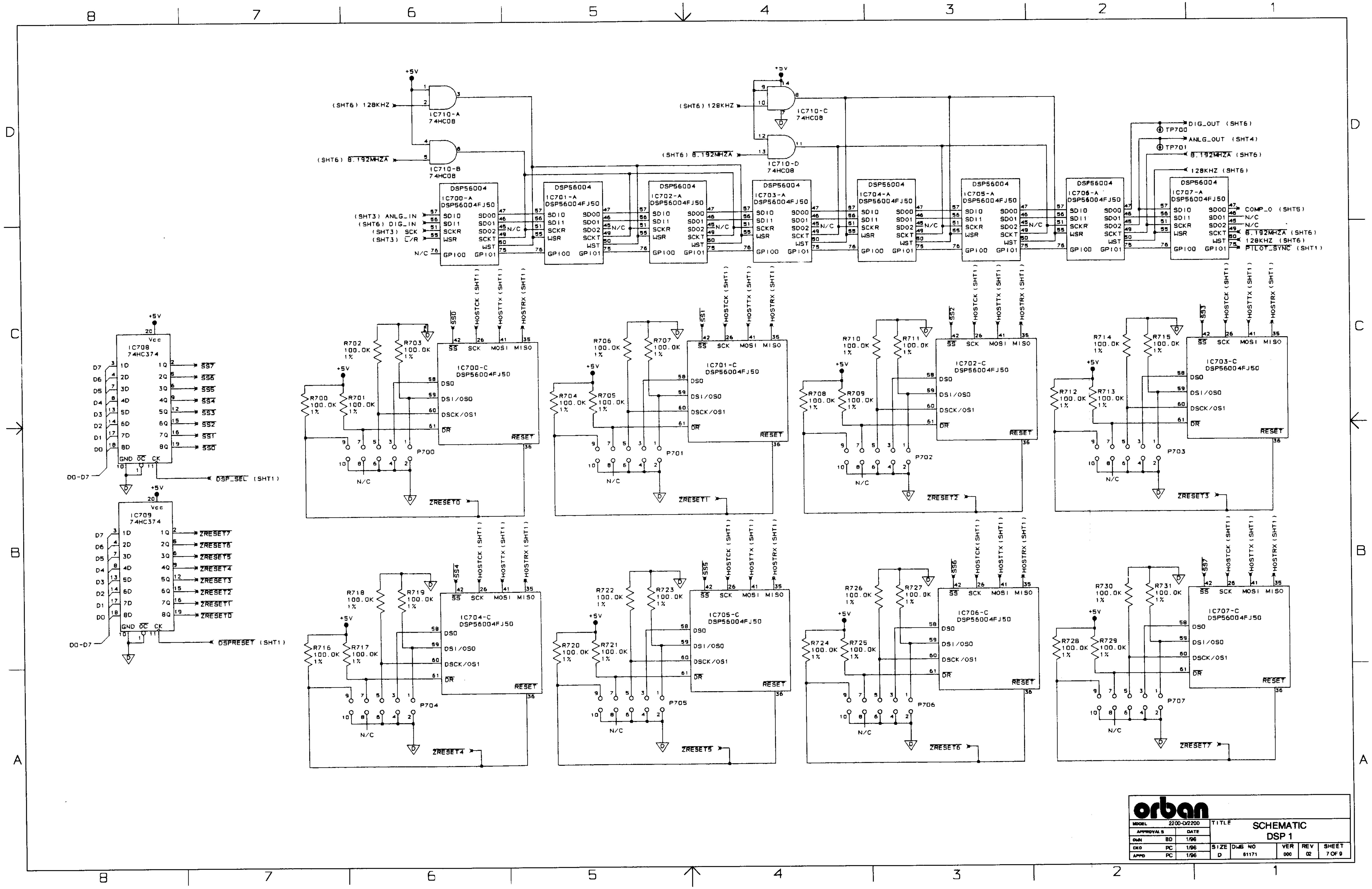
MODEL	2200-D/2200	TITLE SCHEMATIC DISPLAY
APPROVALS	DATE	
DWN BD	1/96	
CKD PC	1/96	SIZE D · DWG NO 61171 · VER 000 · REV 02 · SHEET 2 OF 9
APPD PC	1/96	

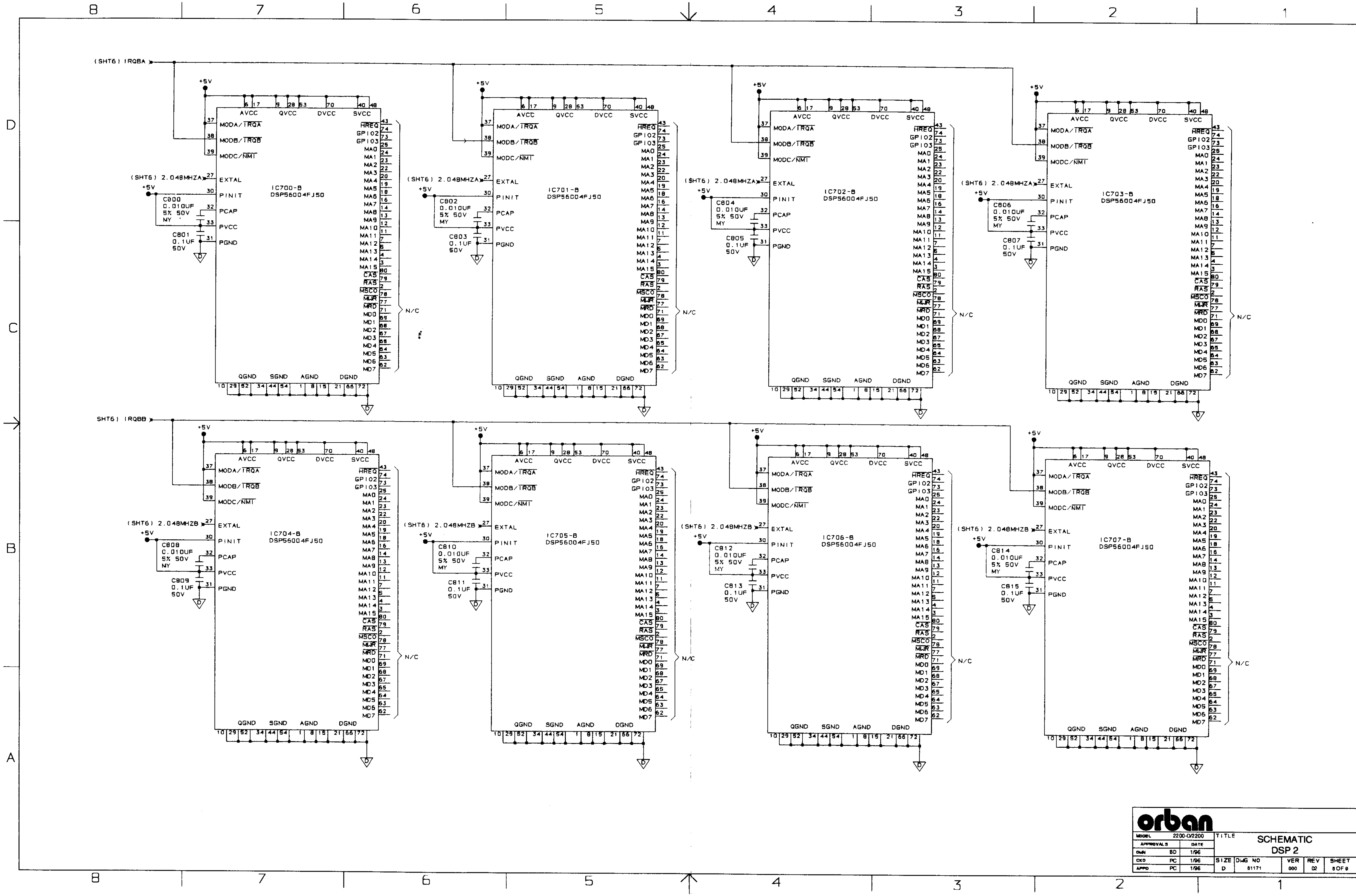

LEFT INPUT
RIGHT INPUT
LEFT INPUT SENSITIVITY
RIGHT INPUT SENSITIVITY
IC300 SSM2017
IC308 SSM2017
IC312 CS5389
IC307 ADG222
IC309 ADG222
NOTE: JUMPERS SHOWN AS SHIPPED.
orban
SCHEMATIC
ANALOG INPUT

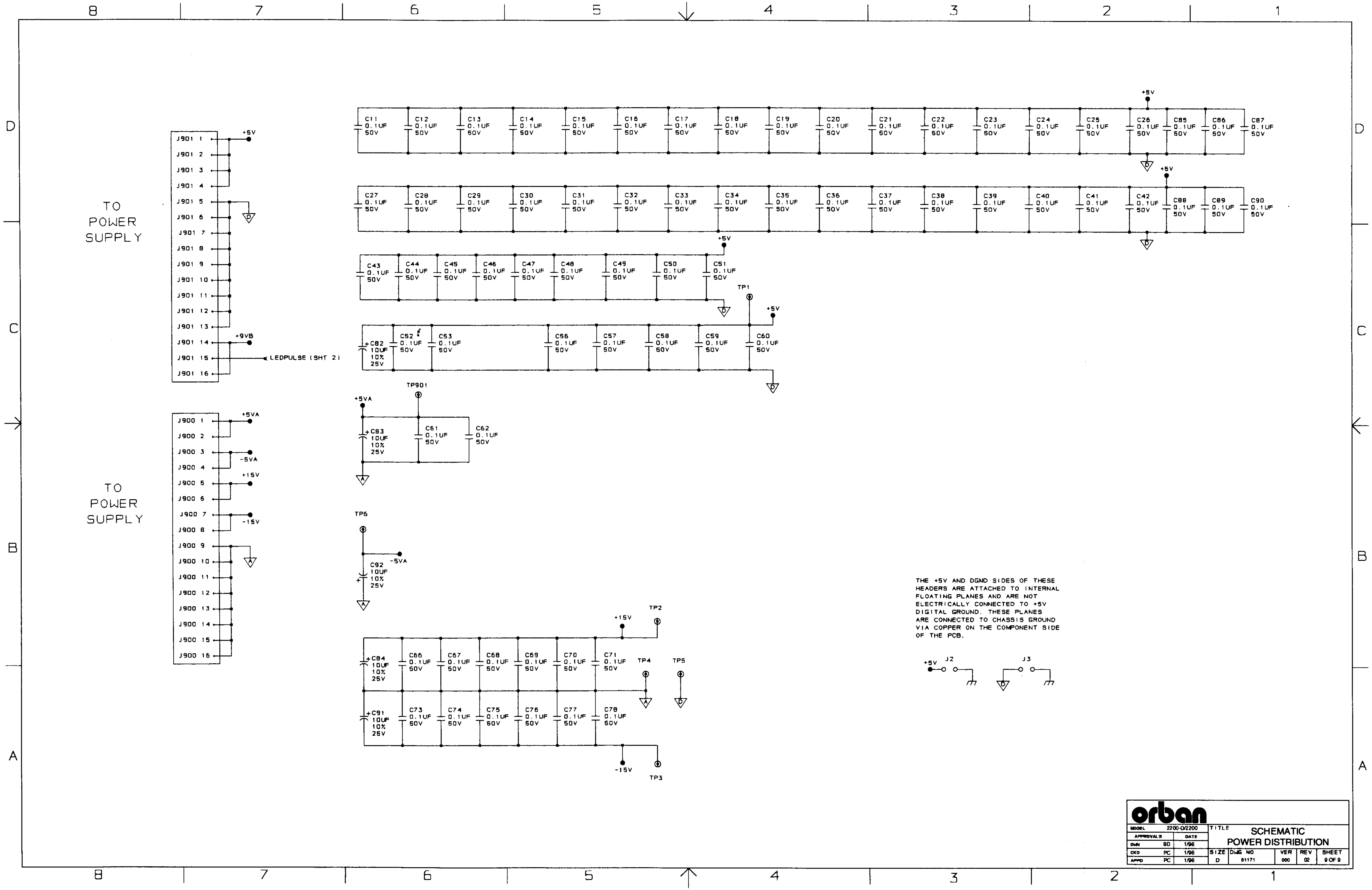

LEFT OUTPUT TRIM
RIGHT OUTPUT TRIM
IC400 CS4328
IC401 AD7524
IC404 AD7524
IC402-A LF412
IC402-B LF412
IC405-A LF412
IC405-B LF412
IC403-A NE5532
IC403-B NE5532
IC406-A NE5532
IC406-B NE5532
IC407-A LM339
IC407-B LM339
IC407-C LM339
IC407-D LM339
IC408-A LF412
IC408-B LF412
Q400 J113
Q401 J113
Q402 J113
Q403 J113
LEFT OUTPUT
RIGHT OUTPUT
J400 MALE
J401 MALE
SHELL
TP400
TP401
TP402
TP403
orban
SCHEMATIC
ANALOG OUTPUT
MODEL 2200-D/2200
SIZE D
DWG NO 61171
VER 000
REV 02
SHEET 4 OF 9


IC501-A
LF412
IC502-A
LM318
IC503-A
LM318
IC504
LT1010
IC505-A
LM318
IC506
LT1010
IC500
AD1861N
IC604-A
74HC74
COMPOSITE 1
OUTPUT
J500
COMPOSITE 2
OUTPUT
J501
L500
9.725MH
L501
12.91MH
L502
10.41MH
NOTE: JUMPERS SHOWN AS SHIPPED
IC501-B
LF412
SPARE
orban
SCHEMATIC
COMPOSITE OUTPUT
2200-D/2200
61171
5 OF 9


AES/EBU INPUT
J600 FEMALE
AES/EBU OUTPUT
J601 MALE
IC600 CS8412
IC601 74HC245
IC603 AD1890JN
IC615 AD1890JN
IC616 CS8401A
IC618 PALC16V8-15
IC613 PALC16V8-25
IC619 74HC4046
IC609 74HC374
IC611 74HC374
IC610 74HC153
Y601 11.2896 MHZ
Y602 8.192MHZ
SPARES
orban
SCHEMATIC
DIGITAL I/O
2200-D/2200
61171
6 OF 9

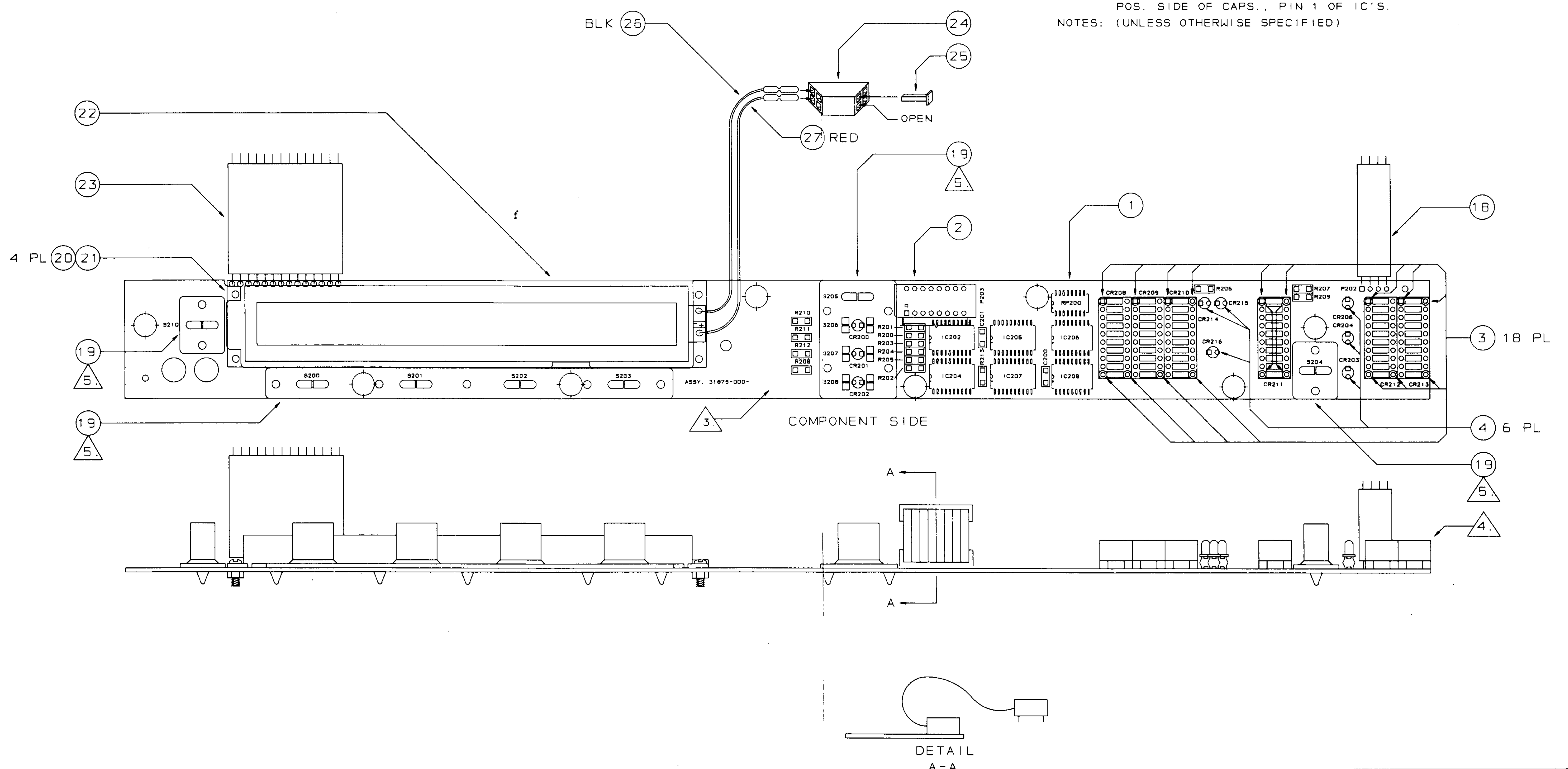

IC710-A
74HC08
IC710-B
74HC08
IC710-C
74HC08
IC710-D
74HC08
(SHT6) 128KHZ
(SHT6) 8.192MHZA
DSP56004
IC700-A DSP56004FJ50
IC701-A DSP56004FJ50
IC702-A DSP56004FJ50
IC703-A DSP56004FJ50
IC704-A DSP56004FJ50
IC705-A DSP56004FJ50
IC706-A DSP56004FJ50
IC707-A DSP56004FJ50
(SHT3) ANLG_IN
(SHT6) DIG_IN
(SHT3) SCK
(SHT3) L/R
DIG_OUT (SHT6)
TP700
ANLG_OUT (SHT4)
TP701
8.192MHZA (SHT6)
128KHZ (SHT6)
COMP_0 (SHT5)
PILOT_SYNC (SHT1)
HOSTCK (SHT1)
HOSTTX (SHT1)
HOSTRX (SHT1)
IC708
74HC374
IC709
74HC374
DSP_SEL (SHT1)
DSPRESET (SHT1)
D0-D7
IC700-C DSP56004FJ50
IC701-C DSP56004FJ50
IC702-C DSP56004FJ50
IC703-C DSP56004FJ50
IC704-C DSP56004FJ50
IC705-C DSP56004FJ50
IC706-C DSP56004FJ50
IC707-C DSP56004FJ50
P700
P701
P702
P703
P704
P705
P706
P707
ZRESET0
ZRESET1
ZRESET2
ZRESET3
ZRESET4
ZRESET5
ZRESET6
ZRESET7
100.0K 1%
N/C
+5V
orban
MODEL 2200-D/2200
TITLE SCHEMATIC DSP 1
SIZE D
DWG NO 61171
VER 000
REV 02
SHEET 7 OF 9
DWN BD 1/96
CKD PC 1/96
APPD PC 1/96


IC700-B DSP56004FJ50
IC701-B DSP56004FJ50
IC702-B DSP56004FJ50
IC703-B DSP56004FJ50
IC704-B DSP56004FJ50
IC705-B DSP56004FJ50
IC706-B DSP56004FJ50
IC707-B DSP56004FJ50
(SHT6) IRQBA
SHT6) IRQBB
(SHT6) 2.048MHZA
(SHT6) 2.048MHZB
N/C
orban
SCHEMATIC
DSP 2

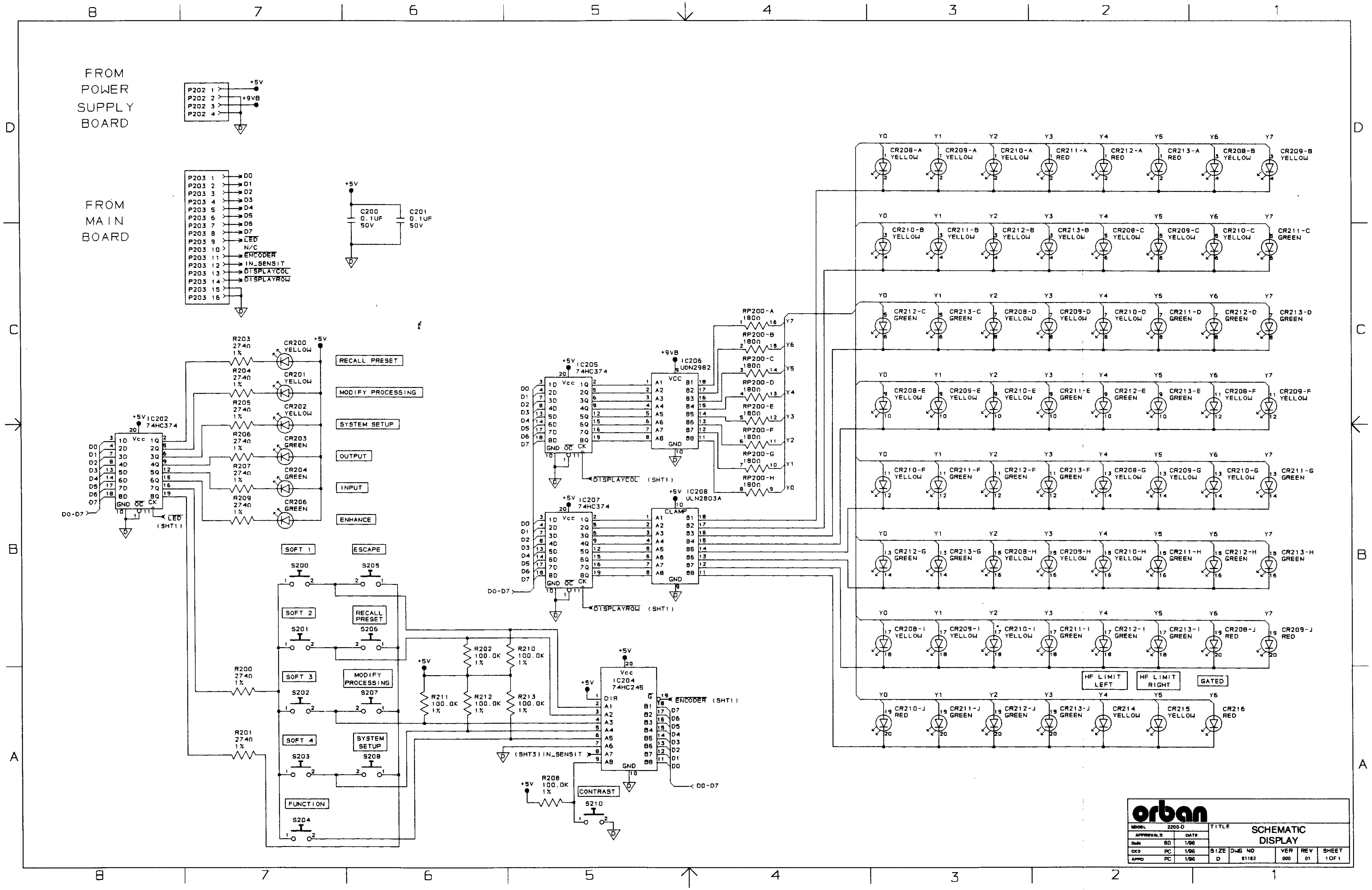

TO POWER SUPPLY
TO POWER SUPPLY
LEDPULSE (SHT 2)
THE +5V AND DGND SIDES OF THESE HEADERS ARE ATTACHED TO INTERNAL FLOATING PLANES AND ARE NOT ELECTRICALLY CONNECTED TO +5V DIGITAL GROUND. THESE PLANES ARE CONNECTED TO CHASSIS GROUND VIA COPPER ON THE COMPONENT SIDE OF THE PCB.
orban
SCHEMATIC
POWER DISTRIBUTION
MODEL 2200-D/2200
SIZE D
DWG NO 61171
VER 000
REV 02
SHEET 9 OF 9

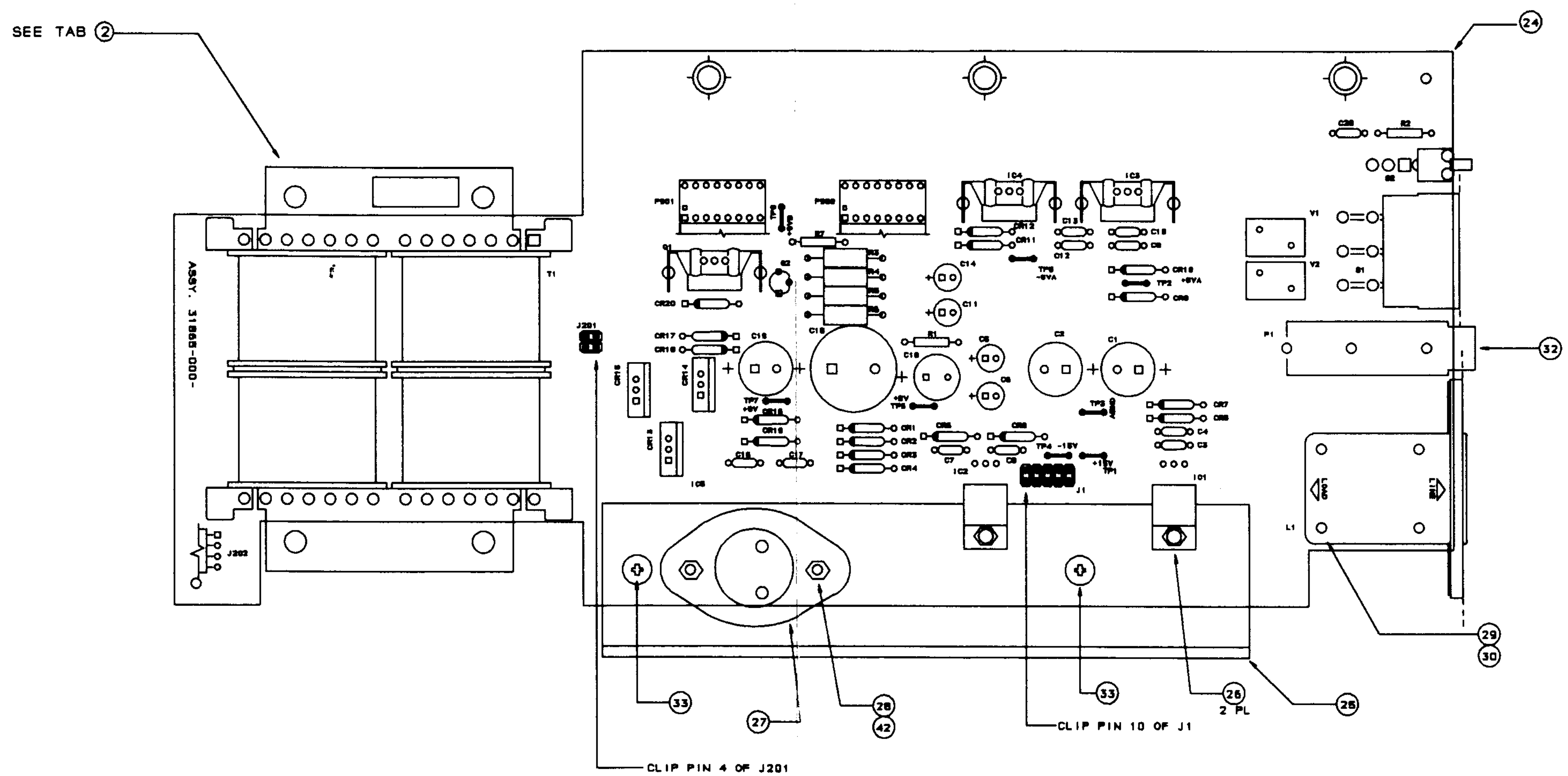
5. ITEM 19 IS AN ELASTOMER KEY PAD WHICH IS CUT INTO 4 PIECES.
4. TOP SURFACES OF CR208 THRU CR213 MUST BE FLUSH WITH ONE ANOTHER AND, AS A GROUP, MUST BE PARALLEL TO THE PRINTED CIRCUIT BOARD.
3. ADD ASSEMBLY REVISION LEVEL IN SPACE PROVIDED.
2. REFERENCE SCHEMATIC DRAWING NO. 61182.000.01
1. SQUARE PADS INDICATE PIN 1 OF CONNECTORS, CATHODE OF DIODES POS. SIDE OF CAPS., PIN 1 OF IC'S.

NOTES: (UNLESS OTHERWISE SPECIFIED)



orban			TITLE PCA DISPLAY 2200			
MODEL	2200					
APPROVALS		DATE				
DWN	BD	3/96				
CKD	PC	3/96	SIZE C	DWG NO 31875	VER 000	REV 02	SHEET 1 OF 1
APPD	PC	3/96				


FROM POWER SUPPLY BOARD
FROM MAIN BOARD
RECALL PRESET
MODIFY PROCESSING
SYSTEM SETUP
OUTPUT
INPUT
ENHANCE
SOFT 1
ESCAPE
SOFT 2
SOFT 3
SOFT 4
FUNCTION
CONTRAST
HF LIMIT LEFT
HF LIMIT RIGHT
GATED
orban
SCHEMATIC DISPLAY

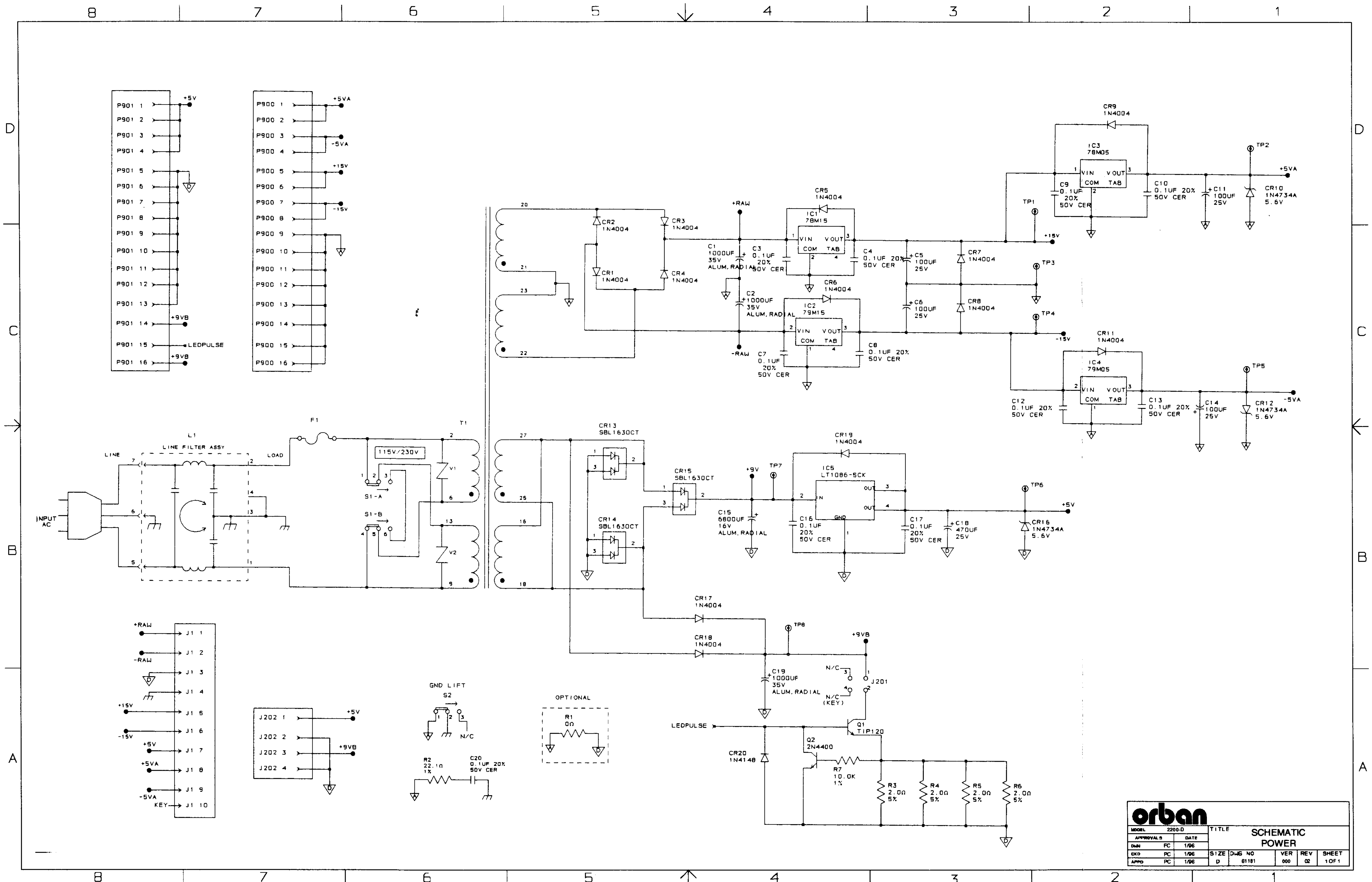


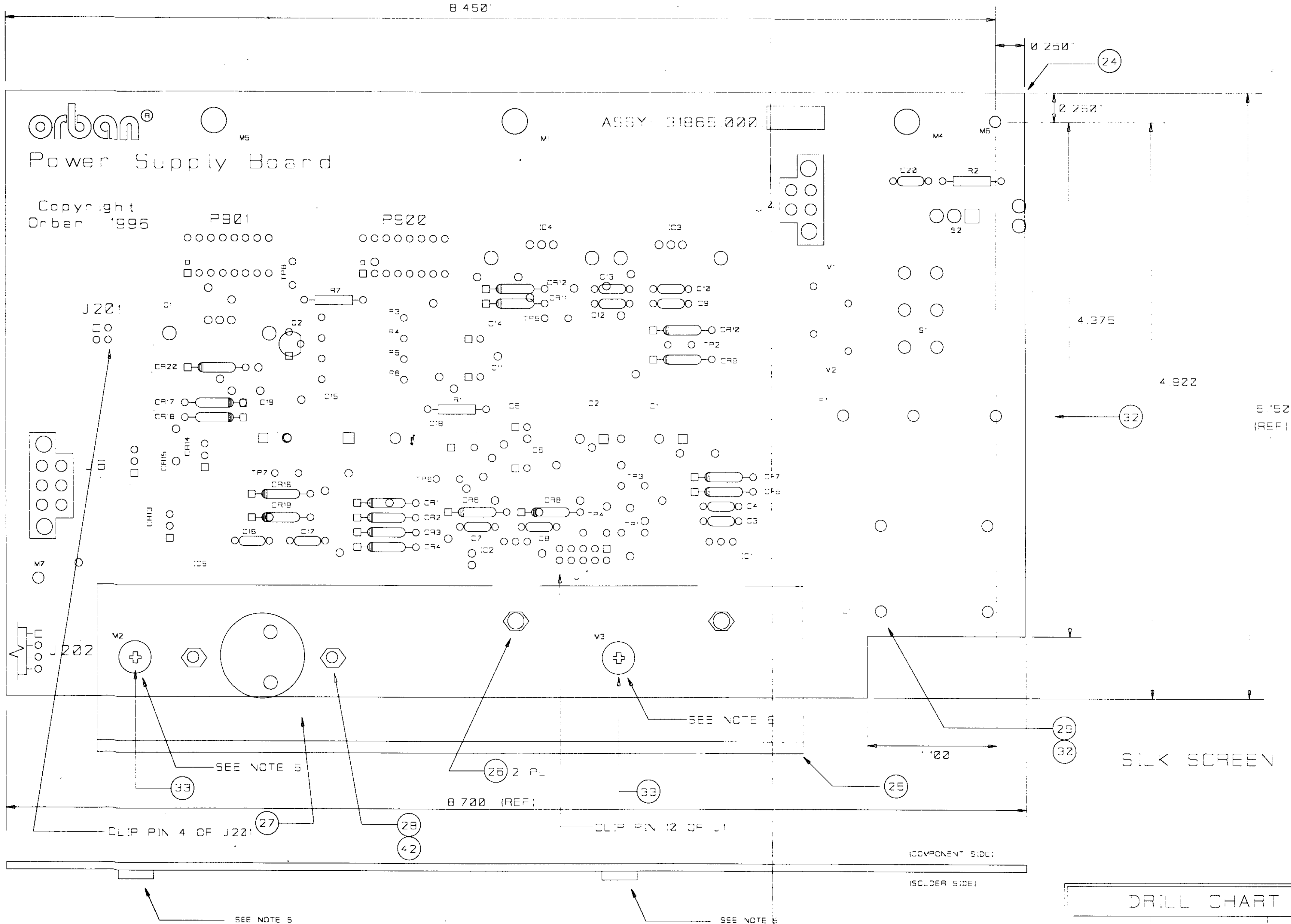
COMPONENT SIDE

TABULATION	
VER	XFMER
000	55033-000
001	55034-000

orban

MODEL	2200-D/2200	TITLE	PCA POWER SUPPLY, 2200
APPROVALS	DATE		
DWN BD	3/96		
CKD PC	3/96	SIZE C	DWG NO 31865
APPD PC	3/96	VER 000 / REV 02	SHEET 1 OF 1


SCHEMATIC
POWER
orban
2200-D
61181



7. BOARD MUST BE UL RECOGNIZED, IDENTIFIED WITH MFGR'S LOGO & TYPE DESIG. ETCHED IN ANY CLEAR AREA ON SOLDER SIDE.
6. SILKSCREEN LEGEND COMPONENT SIDE, COLOR: WHITE.
5. INSTALL PEM FASTENER (KF2 - 440) 2 PLACES SOLDER SIDE AS SHOWN.
4. TRACE TOLERANCE: FINISHED BOARD TRACE WIDTHS MUST NOT VARY MORE THAN +/- 15% FROM FILM SUPPLIED.
3. USE LPI GREEN FOR SOLDERMASK, UL 94V-0, SMOBC, BOTH SIDES.
2. ALL HOLES PLATED THRU (EXCEPT NOTE **)
1. MATERIAL: 0.062 IN THICK EPOXY GLASS FR-4, 2 OZ EACH SIDE OUTER LAYERS.

NOTES: (UNLESS OTHERWISE SPECIFIED)

DRILL CHART

LETTER KEY	FINISHED HOLE SIZE	COUNT
A	0.040"	253
B	0.055"	10
C	0.067"	17
D	0.077"	[illegible]
E	0.118"	10
F	0.125" *	2
G	0.141"	4
H	0.157"	3
J	0.166"	2

FINISHED SIZE AFTER PLATING +/- 0.003"
* 0.125" TOOLING HOLES NOT PLATED

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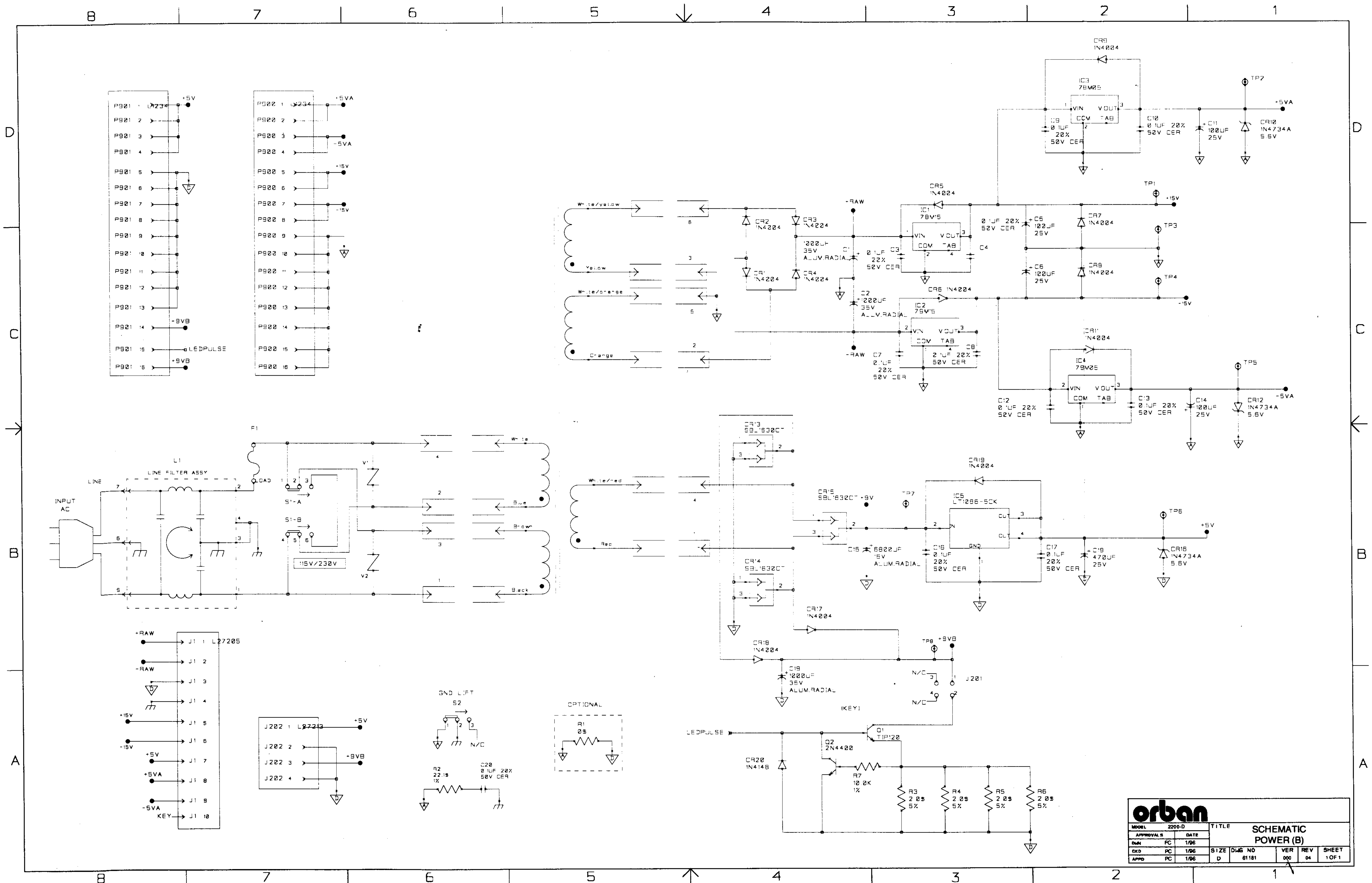
MODEL	2200-D/2200	TITLE PCA POWER SUPPLY (B), 2200				
APPROVALS	DATE					
DWN BD	3/96	SIZE	DWG NO	VER	REV	SHEET
CKD PC	3/96	C	31865	000	04	1 OF 1
APPD PC	3/96					


orban
MODEL 2200-D
SCHEMATIC
POWER (B)
SIZE D
DWG NO 61181
VER 000
REV 04
SHEET 1 OF 1

001

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

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