Operating Manual

OPTIMOD-FM®

MODEL 8100A/1

orban

Operating Manual

OPTIMOD-FM®

MODEL 8100A/1

orban

Orban Associates Inc., 645 Bryant St., San Francisco, CA 94107 USA
Toll Free (800) 227-4498; In California, (415) 957-1067 Telex: 17-1480

95039-000-04 5/86

U.S Patents #4,460,871; #4,249,042; #4,208,548 U.K. Patent #2,001,495
Other Patents Pending. "OPTIMOD-FM" is a Registered Trademark of Orban Associates Inc.

S.J.

FIRST EDITION Issue 2 (revised July 1986)

This manual is for use with all OPTIMOD-FM^R Models 8100A/1 to date. It is not directly applicable to the 8100A/1's immediate ancestor, the Model 8100A.

The Model 8100A/1 processor differs from the Model 8100A in that it has been modified for use with other Orban products such as:

8100A/XT Six-Band Limiter Accessory Chassis
which is used to obtain improved source-to-source consistency and/or presence on smaller radios and in cars.

Cards #5, #6, and #8/9 have been reconfigured in the Model 8100A/1 to allow use of the optional Model 8100A/XT Six-Band Limiter Accessory Chassis. The motherboard has been changed, and a prewired Accessory Port has been included to interface the Accessory Chassis.

In the current revision of this manual, references to the Dolby^R 334 Noise Reduction Processor (including the interface information formerly contained in **Appendix G**) have been deleted. Please contact Customer Service for Dolby 334 interface information. References to FM quadraphonic broadcasting have also been deleted.

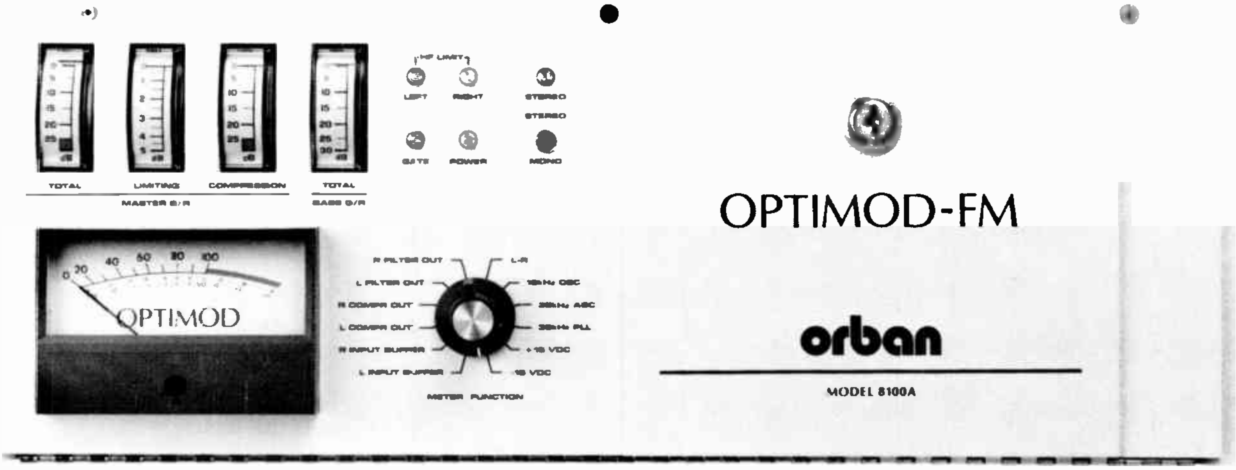
A new **Appendix G** gives information on changing the unit's preemphasis for those countries with a preemphasis standard different from the one used in the USA.

New information on the optional FM Filter Card (#0) has been added. **Appendix K (Audio Quality Considerations in FM Plants)** has been updated, as have references to FCC regulations.

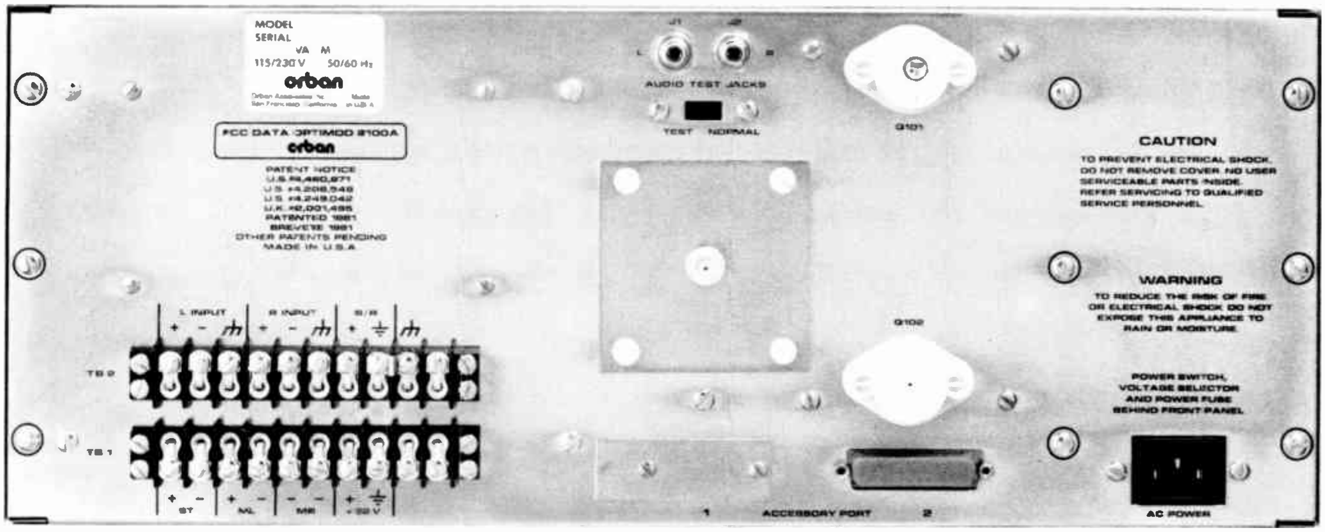
CAUTION

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.

(per UL 813)



Front Panel



Rear Panel

Table of Contents

Table of Contents

APPENDIX K: AUDIO QUALITY CONSIDERATIONS IN FM PLANTS

APPENDIX L: SPECIFICATIONS

APPENDIX M: FUNCTIONS OF JUMPERS ON PC CARDS

Preface

This Manual is organized into two major sections. The first contains information on how to plan your installation, how OPTIMOD-FM^R interfaces with other station equipment, how to set up and adjust OPTIMOD-FM, how to do an in-system performance verification, and brief comments on routine maintenance. You should read Parts 1, 2, and 3 before attempting to install OPTIMOD-FM.

The second section contains Appendices which provide useful information that you may need at some time during the life of OPTIMOD-FM. This is primarily reference material, and you do not need to digest it to install, set up, or operate your unit.

There is no Index, so the TABLE OF CONTENTS should be used to help you find the information you want. The TABLE OF CONTENTS provides an overview of the organization of the manual, and lists in some detail the topics discussed.

Registration Card: The original purchaser should have received a postpaid Registration Card packed with this manual.

Registration is of benefit to you because it enables us to tell you of new applications, possible performance improvements, service aids, etc., which may be developed over the life of the product. It also provides us with the date of sale so that we may more promptly respond to possible claims under Warranty in the future (without having to request a copy of your Bill of Sale or other proof of purchase).

Please fill in the Registration Card and return it to us.

If the Registration Card has become lost or you have purchased the unit used, please photocopy the image of the card reproduced on the following page and send it to us in an envelope. Use the address shown on the title page.

Do not allow your Dealer to submit the card for you. If he forgets, you can miss important future mailings and may be delayed in obtaining Warranty service.

Packing Material: The carton in which your OPTIMOD-FM was shipped was carefully designed to prevent damage from the stresses ordinarily encountered in commercial shipments. SAVE THE CARTON AND ALL PACKING MATERIAL in case you ever have to ship the OPTIMOD-FM chassis back to the factory for service.

User Feedback Form: We are very interested in your comments about this product. Your suggestions for improvements to either the product or this manual will be carefully reviewed. A User Feedback Form is provided for your convenience. If it is missing, please write us at the address on the title page. Thank you.

FCC Filing (U.S.A): A verification has been filed with the Commission that the stereo generator section of OPTIMOD-FM Model 8100A/1 meets all requirements of FCC 73.322 when used with a direct-FM exciter originally designed with sufficient bandwidth to accept a stereo generator. As of this writing, there is no requirement that the FCC be notified that you have changed stereo generators.

Security (Keys and Locks): To control access to the setup controls, the access door is fitted with a lock. Two keys are supplied. These can be duplicated as desired.

The dealer from whom your 8100A/1 was purchased can supply additional keys, as can the factory. In either case, your Registration Card must be on file at the factory, and you must supply your serial number to obtain replacement keys.

If all keys are lost, you can obtain access by removing the three hex-socket screws from the top of the main front panel with a 5/64" hex wrench (one was supplied with the unit).

If you wish to make the unit's adjustments more secure, obtain similar splined-socket or aircraft tri-point screws (and tools), and use these in place of the hex-socket screws supplied. (Tools for these are not commonly found in hardware stores or other places D.J.'s might frequent.) The screws are 6-32 x 3/8" 82° flathead, nickel-plated steel.

Model # 8100A/1 ______ Serial # ______

Name and/or Title ______

Organization ______

Street ______

City/State/Country ______

Zip or Mail Code ______ Phone # ______

Purchased from ______ City ______ Date of Purchase ______

Nature of your application ______

How did you hear about it? ______

Comments: ______

______

Duplicate Registration Card

PART 1:

Introduction

Function of OPTIMOD-FM: OPTIMOD-FM is an integrated signal-processing system which replaces conventional compressors, limiters, clippers, and stereo generators. Each part of the OPTIMOD-FM system has been precisely engineered to be compatible with all other parts to achieve optimum performance.

OPTIMOD-FM should be fed unprocessed audio. NO OTHER AUDIO PROCESSING IS NECESSARY, OR DESIRABLE.

Briefly, the OPTIMOD-FM system performs the following functions:

1. It rides gain over a range of as much as 25dB, compressing dynamic range and compensating for gain riding errors on the part of operators. The amount of dynamic range reduction ordinarily produced is adjustable. When OPTIMOD-FM is operated with an optimum release time setting, gain riding and compression are virtually undetectable because of advanced program-controlled time constants and multiband compression.

2. It prevents aliasing distortion in the stereo generator by means of bandwidth-limiting 15kHz lowpass filters. Full overshoot compensation is provided for these filters. OPTIMOD-FM thus provides extremely tight control over peak modulation, preventing overmodulation and controlling the baseband spectrum simultaneously.

3. When OPTIMOD-FM's dual-band compressor is operated in "independent" mode, OPTIMOD-FM can make audio quality more consistent by correcting frequency balances between bass and midrange material. When operated in "wideband" mode, OPTIMOD-FM will preserve frequency balances and will produce an output which sounds almost precisely like its input.

4. OPTIMOD-FM prevents peak overload and overmodulation due to the effects of the FM preemphasis curve.

5. OPTIMOD-FM generates a stereo baseband signal with outstanding separation, low crosstalk, and vanishingly low distortion and spurious components.

Optional accessories for the OPTIMOD-FM provide for separate studio and main chassis (8100A/ST), six-band limiting (8100A/XT), and special filtering (Accessory Kit 22). These accessories are discussed in greater detail at the end of this part of the Operating Manual.

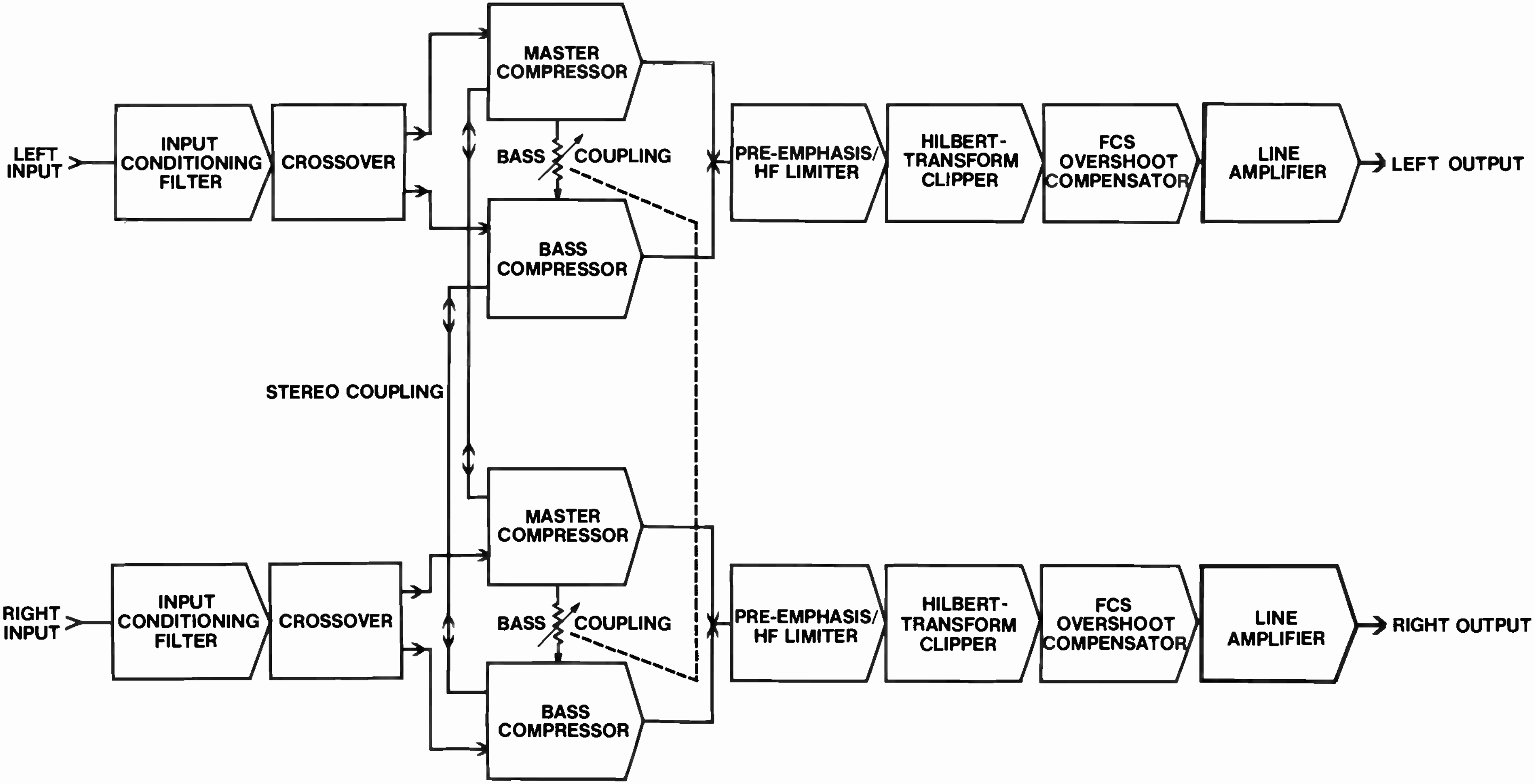


Fig. 1-1: OPTIMOD-FM Model 8100A/1 Signal Flow Diagram

SIMPLIFIED SYSTEM DESCRIPTION

OPTIMOD-FM Model 8100A/1 consists of six basic blocks. See cover and figure 4-5 for illustration of controls.

1. **Input Conditioning Filter:** This consists of an allpass phase scrambler to make peaks more symmetrical (thus reducing clipping distortion and permitting higher loudness), and a 30Hz 18dB/octave highpass filter to prevent subsonic information from disturbing the operation of the audio processing or exciters' AFC's. Even if an AFC doesn't unlock, it can attempt to "track" subsonic information, thus producing IM distortion. The 30Hz highpass filter can be defeated (although we have purposely made it slightly inconvenient to do so); the phase scrambler is an essential part of the system and is non-defeatable.

2. **Dual-Band Compressor:** This consists of two compressors in parallel: "Bass" which processes audio below below 200Hz (12dB/octave crossover), and "Master" which processes above 200Hz. A BASS COUPLING control adjustable by the user determines if the two bands will operate discriminately ("independent" mode), or if the "Bass" band will be forced to track the "Master" band ("wideband" mode), thus preserving frequency balances. Intermediate partial-crosscoupling settings are also available.

Even in "wideband" mode, the bass control loop is still active. Therefore, heavy bass will cause a momentary reduction in the gain of the "Bass" band rather than forcing gain reduction of the entire signal (as in a true wideband system), thus avoiding pumping.

Time constants and other parameters of the Dual-Band Compressor have been adjusted so that the summed and preemphasized output of the two bands can be connected directly to the FM Smart Clipper^{TM}. No further gain reduction is required for distortion control, and maximum naturalness is preserved.

The threshold of limiting is adjustable over a 6dB range by the CLIPPING control. This determines the output level of the compressor, and thus the amount of HF limiting and clipping which occur later in the system.

The release time of the "Master" band only is adjusted with the RELEASE TIME control, thus permitting loudness/fatigue tradeoffs according to format requirements.

Both "Master" and "Bass" compressors are gated such that the release time is slowed by a factor of approximately 50:1 when the input level drops below a threshold adjustable by the GATE THRESHOLD control. This prevents noise rush-up during program pauses, and makes the 25dB gain reduction range usable. Simultaneously, the gain does not get "stuck" forever, so low-level musical passages are eventually increased in level. Since gain recovery takes over one minute to occur in GATED mode, the gradual increase in level cannot be perceived.

Gain reduction in both "Master" and "Bass" compressors is metered by edgewise-reading meters calibrated with a dB-linear scale. To provide best value, no attempt has been made to make these meters extremely accurate, and their readings may disagree with the actual gain reduction by a much as ±20%. This accuracy is fully adequate for the purpose, since the amount of gain reduction varies widely with variations in program material and operator gain riding.

3. Preemphasis And High Frequency Limiter: The summed outputs of the two compressors are applied to a phase corrector, 24dB/octave 15kHz lowpass filter, preemphasis network, and high frequency limiter. The purpose of the lowpass filter is to prevent out-of-band components from affecting the operation of the high frequency limiter and to avoid intermodulation between out-of-band frequency components and in-band frequency components in the clipper. Phase correction reduces the peak level increase caused by filter ringing and preemphasis to the theoretical minimum, thus reducing the amount of clipping.

The high frequency limiter is controlled by high frequencies only (rather than by the peak level of the preemphasized signal, as in the old Model 8000A), thus eliminating any possibility of modulation of high frequency content by low frequency material.

The threshold of limiting of the high frequency limiter is user-adjustable over a 3dB range, permitting brightness and high frequency distortion to be traded off according to format requirements. Because the FM Smart Clipper incorporates IM distortion cancellation, substantially more clipping can be accomplished without objectionable distortion than in the old Model 8000A, and significantly improved high frequency power handling capability is achieved through the system.

4. FM Smart Clipper: The Smart Clipper provides the peak limiting function, and contains filters to assure that the clipping does not introduce out-of-band frequency components above 19kHz which could cause aliasing distortion.

The output of the HF limiter is applied to a clipper with automatically-varying threshold. This clipper performs the basic peak limiting function. The output of the clipper is subtracted from its input, thus deriving the distortion added by the clipper. This distortion component is lowpass filtered at 2.2kHz (the knee of the 75us preemphasis curve), and then added to the clipped signal. This "smoothing signal" cancels all clipper-induced distortion below 2.2kHz by 30dB or more, and is particularly effective in eliminating the effects of high-frequency IM, such as sibilance splatter.

The 2.2kHz distortion-cancelling lowpass filter has a time delay. To assure proper distortion cancellation, the main clipped signal must be delayed by an equal amount before the main signal and distortion-cancelling signal are added. The main signal is delayed by a phase-corrected 15kHz lowpass filter, which also removes all out-of-band harmonics caused by the clipping process.

5. Frequency-Contoured Sidechain (FCS) Overshoot Corrector: The output of the Smart Clipper contains overshoots due to the addition of the distortion-cancelling signal, and due to unavoidable overshoots in the 15kHz filter. These overshoots must be eliminated without adding out-of-band frequency components. This is done in the FCS Overshoot Corrector.

The FCS circuit first derives that part of the signal exceeding the 100% modulation point by means of a "center clipper". If these overshoots were then subtracted from the input signal, the overshoots would be cancelled -- in fact, doing so would be equivalent to simple clipping. Unfortunately, this can't be done because the overshoots contain out-of-band frequency components which would cause aliasing distortion if applied directly to the stereo generator.

We therefore lowpass-filter the overshoots to eliminate out-of-band components. If the overshoot filter had a flat response to its cutoff frequency, this filtering action would reduce the amplitude of high-frequency overshoots (by removing out-of-band harmonics which make the overshoots "spikey"). This would result in incomplete cancellation of the overshoots after subtraction. The overshoot filter is therefore designed to have a rising response at 15kHz, effectively increasing the gain of the fundamentals of the higher-frequency overshoots and compensating for the fact that their harmonics have been removed. The overshoot extractor and this filter are the "Frequency-Contoured Sidechain".

The overshoot filter has phase shift. Phase shift networks are therefore included in the main path to make sure that the overshoot subtraction process works correctly, and that the overall FCS system has constant time delay.

The rising response of the overshoot filter means that essentially no extra subtraction gain (compared to the system operated without the filter as a simple differential clipper) is required. Any low frequency IM introduced by the FCS circuit is therefore no worse than the low-frequency IM caused by a simple clipper.

Because the FCS circuit is an instantaneous system and uses no gain reduction or dynamic filtering, it causes neither pumping nor dulling of program material.

6. Stereo Generator: The stereo generator accepts the processed outputs from the left and right FCS circuits, and produces the stereo baseband signal. It is characterized by high stability, very low distortion, and minimal spurious outputs. 19kHz oscillator level, pilot phase, and separation are all controlled and stabilized by means of servo loops.

The stereo baseband is generated by the "matrix" technique, as opposed to the more common "switching" design. The "matrix" design modulates only the L-R component, and passes the L+R component through to the output without degradation due to switching. Since the L+R component almost always dominates, this results in maximum audio quality. In addition, no baseband lowpass filter is required. Such a filter would add to system cost, and could compromise separation.

To facilitate adjustment of pilot phase and measurement of main-channel-to-subchannel and subchannel-to-main-channel crosstalk, two special TEST modes are provided. These apply the right channel audio directly to the main or sub inputs of the stereo generator. For testing, these stereo generator audio inputs can be accessed by means of a pair of TEST jacks on the rear chassis apron. This provides an alternate means of measuring crosstalk and other stereo generator performance parameters.

The stereo generator can be operated in STEREO, MONO LEFT, or MONO RIGHT modes. All three modes can be selected by remote control by means of optically-isolated remote control terminals. STEREO and one of the two MONO modes can be selected locally by means of a front-panel switch. An internal strap determines which of the two MONO modes is selected. Another internal strap determines which of the three modes will be entered on powerup.

VU Meter: The front-panel VU meter can monitor the level of the left and right audio at three different points in the circuitry. (See **Block Diagram** [p. J-21] for metering points.) The meter also monitors the difference between left and right channels (L-R) to aid channel balance adjustments.

Three stereo generator parameters are monitored: 19kHz pilot oscillator level, 38kHz AGC control voltage, and 38kHz Phase-Locked Loop control voltage. The latter two readings will vary, and are used to check whether these servo loops are within their ordinary operating range, or whether they are saturated.

Finally, the +15 and -15V power supply voltages are monitored.

Accessories Currently Available:

8100A/ST STUDIO ACCESSORY CHASSIS

(A separate manual is supplied with this unit.)

In some installations, it may be desirable to perform the compression function ahead of the STL to optimize system signal-to-noise ratio by relaxing STL headroom requirements. (STL's are discussed further in **Part 2**). The Model 8100A/ST Accessory Chassis (which performs this function) consists of a chassis shell assembly with a power supply/metering card and three empty card locations. Cards #3, #4, and #5 (the Dual-Band compressor cards) are removed from the Main Chassis and installed in these locations. Dummy cards provided with the Accessory are then installed in the Main Chassis to preserve the signal path.

The Accessory Chassis provides convenient access at tne studio to all operating controls except for H-F Limiting and stereo/mono mode switching. It provides the same gain reduction metering as the Main Chassis, and several diagnostic metering functions as well.

More complete information about the 8100A/ST is provided in a separate manual shipped with the unit.

8100A/XT SIX-BAND LIMITER ACCESSORY CHASSIS

(A separate manual is supplied with this unit.)

The optional 8100A/XT Accessory Chassis has been created to provide aggressive multiband processing for stations that desire bright, loud, "highly-processed" audio. Derived from OPTIMOD-AM Model 9100A, 8100A/XT consists of a stereo bass equalizer which drives a stereo six-band limiter cascaded with the exclusive Orban distortion-cancelling multiband clipping system. When added to the basic 8100A/1 system, the 8100A/XT creates a dense, consistent sound without pumping or other obvious processing artifacts.

Functionally, the unit replaces the high frequency limiter within the 8100A/1 and permits the following objectives to be met:

-- For most types of program material, increased loudness for a given level of audible processing side-effects (by comparison to an unmodified 8100A/1 or 8100A);

-- Improved consistency from source to source due to the "automatic equalization" effect of the six-band limiter; and,

-- Increased presence and intelligibility on smaller radios and in autos.

The 8100A/XT is most appropriate for contemporary popular music formats, although its ability to improve consistency and intelligibility can make it useful for talk and news formats as well. It is generally unsuited for "beautiful" or classical formats.

The Model 8100A/1 is prewired to readily accept the 8100A/XT. (Older 8100A's must be converted by means of retrofit kit "RET-27" before they can accept the 8100A/XT.)

The 8100A/XT uses a great deal of the existing 8100A/1 circuitry in the interest of achieving efficiency and economy. The 8100A/XT (which requires two rack units) is mounted directly below the 8100A/1 and is then plugged into a multipin connector on its rear panel. Jumpers on Cards #5, #6, #8, and #9 within the 8100A/1 must then be moved according to instructions in a manual supplied with the 8100A/XT.

The normal positions of 8100A/1 card jumpers when the 8100A/XT is not in use are tabulated in **Appendix M** of this Manual.

The 8100A/XT is fully compatible with the Studio Accessory Chassis Model 8100A/ST, allowing use of the 8100A/XT in either single- or split-chassis installations. In all cases, the 8100A/XT is mounted immediately below the main (8100A/1) chassis.

Full instructions for installing and operating the 8100A/XT are included in the **Operating Manual** for that unit. That Manual also includes a complete discussion of the 8100A/XT's principles of operation, how it relates to the 8100A/1, and other information important to those who plan to use the unit.

ACCESSORY KIT 22: FM FILTER CARD (#0)

(Separate installation and alignment instructions are included in this kit.)

The OPTIMOD-FM Model 8100A/1 has been designed to meet all FCC Rules regarding crosstalk between the main channel and subchannel and vice-versa. However, higher performance than this is sometimes desirable to fully protect SCA subcarriers from interference caused by the operation of the 8100A/1's safety clippers (on cards #8 and #9) on extremely densely-processed program material.

Card #0 replaces each safety clipper with two cascaded overshoot-compensated lowpass filters. This reduces "splatter" in the SCA region of the baseband by 25-30dB compared to the "stock" 8100A, and simultaneously reduces any residual overshoot by about 5%. This processing can achieve loudness within 0.3dB of that produced by composite clipping (an inaudible difference), while protecting the SCA region about 40-50dB better than a composite clipper.

Card #0 is installed in the signal path between the output of the existing cards #8/9 and the input of card #7 (stereo generator). The frequency response of the card is typically +0, -0.1dB, 30-15,000Hz. An 8100A/1 containing this card will therefore perform well within its ±0.75dB specification.

Card #0 has been specifically designed to be used in conjunction with the 8100A/1's existing FCS Overshoot Corrector. Card #0 is therefore unusable with our older Optimod-FM Model 8000A, or with other manufacturers' equipment.

This concludes the **Introduction and Simplified System Description.** The next part of this manual (**Application**) should now be read carefully to assure that your installation produces optimum results.

PART 2:

Application

This Part of the Manual provides essential information on how OPTIMOD-FM fits in with the rest of the equipment at your station. **Appendix K** contains further information on achieving high audio quality.

Studio/Transmitter Links: There are five types of studio/transmitter links in common use internationally in FM stereo service. These are:

1) Analog land-lines (telephone lines)
2) Dual-microwave STL's
3) Composite baseband microwave STL's
4) PCM (Pulse-Code Modulation) links
5) Video STL's with PCM adapters

All except (3) carry both audio channels either directly, or in some encoded form other than the standard 19kHz "pilot tone" stereo baseband. These links are ordinarily fed both left and right audio channels in non-encoded form, and their receiver output is the regenerated left and right channels.

The composite STL (3) carries the standard "pilot tone" stereo baseband, and is therefore fed from the output of a stereo generator like the one in the 8100A/1. The receiver output of the composite STL is the stereo baseband signal, which is applied directly to wideband input of the FM broadcast transmitter's exciter.

In general, highest quality is obtained by use of a composite microwave STL provided that a line-of-sight transmission path of less than 10 miles or so exists between studio and transmitter. If not, RF signal-to-noise ratio, multipath distortion, and diffraction effects can cause serious quality problems.

The dual-microwave system provides more noise immunity. However, problems include gain- and phase-matching of the left and right channels, preemphasis-induced overloads, and requirements that the audio applied to the microwave transmitters be processed to prevent overmodulation.

Land-line quality is extremely variable, ranging from excellent to atrocious. The decision on whether to employ land-lines depends a great deal on the quality locally available. However, even the best land-lines tend to slightly veil audio quality due to line equalizer characteristics, phase shifts, and repeaters of indifferent quality.

PCM links are generally unavailable in the USA as of this writing, although they are widely used in Europe. They achieve good noise performance and consistency at the expense of a very sharp high-frequency cutoff, rapid changes in group delay around cutoff (unless elaborate phase equalization is used), and quantization distortion. At the moment, there is considerable disagreement over how elaborate the coding must be to render quantization distortion inaudible to critical listeners, and no PCM system should be accepted without critical listening tests.

Some stations are sending PCM-encoded audio through a video STL at frequencies above 23GHz. Typically, consumer PCM adapters (from Sony or dbx, for example) are being used. The quality of signal received at the transmitter through this type of STL is high.

OPTIMOD-FM Model 8100A/1 is available in either single- or dual-chassis configurations. The dual-chassis splits the system at a point between the output of the Dual-Band Compressor and the input of the HF Limiter.

The dual-chassis configuration is ordinarily used with STL's of types (1), (2), and (4) of modest performance characteristics. By performing initial compression before the STL input, the dual-chassis version can prevent STL overload and can aid in achieving superior STL signal-to-noise ratio.

The single-chassis arrangement is suited for composite STL's, or for installations where studio and transmitter are at the same site or are connected by short, high-quality lines. Because it is less expensive than the dual-chassis version, the single-chassis version is also suited for use with any STL having extremely wide dynamic range (80dB or better) such that unprocessed audio can be passed to the compressor without danger of noise build-up when the compressor's gain increases towards its maximum.

It is important to note that the compressor section alone does not control peak levels accurately, and does not compensate for overloads caused by preemphasis. (Peak limiting and high frequency limiting are performed later in the system.) It is therefore necessary to allow headroom in the STL to accommodate compressor overshoots. If the STL is preemphasized at 50 or 75us (as is the case with many dual-microwave systems), further headroom must be allowed to accommodate the peak level increases caused by the preemphasis. Precise STL setup recommendations are provided in **Part 4** of this Manual.

If STL preemphasis can be readily modified, use of 25us preemphasis will match headroom to the typical spectral distribution of contemporary recorded material, thus achieving optimum STL signal-to-noise ratio.

Exciters: OPTIMOD-FM will interface with most direct-FM exciters although some older exciters designed before FM stereo standards were adopted may have to be modified to extend their bandwidths. If an input transformer is an essential part of such an older exciter (driving a push-pull input or providing a bias path for the input stage, for example), it may be possible to replace such a transformer with a Western Electric 111C coil. When loaded with 470 ohms, these coils can pass a stereo composite signal such that all requirements in part 73.322 of the FCC Rules are met.

WARNING!

(U.S.A. customers)

> Exciter modifications must conform to the requirements of 73.257 of the FCC Rules.

OPTIMOD-FM cannot be used with exciters using phase modulators. Such exciters include Phasotron™ and Serrasoid™ designs.

In general, modern solid-state exciters provide both vastly improved reliability and audible improvements in sound quality when compared to pre-1961 designs, and such older exciters should be retired if at all possible. In fact, the latest generation of exciters provides audibly improved performance over the earlier generation of solid-state units as well.

To our knowledge, the baseband output of OPTIMOD-FM can be directly connected without additional interfaces to the wideband input of all direct-FM exciters designed after 1963, with the exception of the following:

RCA BTE-15: Obtain a jumper plug directly from Orban Associates, Inc.

Gates (Harris) TE-1, TE-3: Obtain the Orban Associates ATE-3F wideband interface panel, which fits in the chassis in place of the TE-3 stereo generator.

Collins 310Z-1: Obtain a factory update from Continental to convert to a 510R-1, and obtain the 785E-1 wideband interface card directly from Continental.

Continental 510R-1 (Collins 310Z-2): Obtain a wideband interface card directly from Continental.

Most wideband inputs are BNC connectors, and require a BNC-to-BNC shielded cable to connect to OPTIMOD-FM. Ordinarily, RG-58A/U cable should be used. (The Collins A830-2 exciter has an RCA phono jack input, and requires a BNC-to-RCA cable.)

Detailed information for interfacing OPTIMOD-FM with the BTE-15, TE-1, TE-3, 310Z-1, and 310Z-2 exciters is found in **Appendix H.**

RF Amplifiers And Antennas: It is important that the RF amplifier and antenna be sufficiently wideband to pass all sidebands of the FM signal without attenuation or phase shift. Ideally, the bandwidth should exceed 500kHz. Narrowband amplifiers and/or antennas may make the highs sound "gritty". This may become particularly noticeable due to the 8100A/1's outstanding high frequency power handling capability.

Any audible degradation can be assessed by using a high-quality FM tuner in "wideband" mode to compare the sound normally produced on the air with the sound produced by picking up the output of the exciter alone. The exciter must be correctly terminated by a resistive 50-ohm load. (If you leave it connected to the transmitter, VSWR resulting from impedance mismatches between exciter and RF amplifier can also cause distortion.) A tuner is specified because many modulation monitors use older design techniques and are unsuited for accurately assessing audio of exceptional quality.

SCA: OPTIMOD-FM operates well with SCA's because OPTIMOD-FM provides excellent baseband spectrum control, thus protecting that part of the baseband occupied by the SCA. No special SCA precautions need be taken; SCA should be implemented according to the instructions of the exciter manufacturer.

Although the 8100A/1 has been designed to meet all FCC Rules regarding crosstalk between the main channel and subchannel, extremely densely-processed program material may make it desirable to more fully protect SCA subcarriers from interference caused by the operation of the 8100A/1's safety clippers. The optional FM Filter Card (available as Accessory Kit 22) replaces each safety clipper with two overshoot-compensated lowpass filters. This processing can achieve loudness within 0.3dB of that produced by composite clipping (an inaudible difference), while protecting the SCA region about 40-50dB better than would a composite clipper.

Older exciters may not have separate SCA inputs. In most cases, one can be added simply by passively summing into the modulated oscillator through a resistor and small capacitor. Note, however, that such older exciters often suffer from narrow RF bandwidths which may cause SCA "birdies" due to intermodulation. Such problems may also occur if SCA injection is not limited to 10% modulation and deviation to ±4kHz.

If you are using a composite STL, be sure to limit SCA modulation to the amount recommended by the STL manufacturer or IM between the SCA and the main program may occur in the STL.

Remote Control Functions: OPTIMOD-FM's three operating modes (STEREO, MONO LEFT, and MONO RIGHT) can be selected by remote control. The MONO LEFT and MONO RIGHT functions are useful in the case of failure of one channel of the STL or one channel of 8100A/1 audio processing. The good channel can be fed in mono, and programming can be continued by selecting the good channel and MONO mode.

The remote control ports are easily interfaced with virtually all commercial remote controls.

A remote TOTAL MASTER GAIN REDUCTION meter can also be fitted to enable the operator to observe the amount of gain reduction his levels are producing.

Details on how to interface the remote functions is provided in **Part 3** of this manual.

DIFFICULT ENVIRONMENTS

1. Where humidity is typically high, the environment should be controlled to prevent moisture from condensing on circuit cards of all plant equipment, including OPTIMOD-FM, as this can degrade performance. Using some of the exhaust from the transmitter to heat the building slightly above ambient temperature is often sufficient to prevent problems.

2. If electrical storms are frequent, it may be advisable to add suitable varistors between each incoming wire (AC, remote control, and audio) and a solid earth ground as indicated by local experience.

3. After a power failure, bear in mind that the stereo generator mode-control logic will come up according to the power-up mode for which it was strapped (see **Initialization Options** in **Part 3**), unless the power interruption was less than one second or so. In this case, the powerup circuitry may not have sufficient time to reset, and an undesired mode may be entered.

PART 3:

Installation

Registration Card: If you have not already done so, please fill out the Registration Card fully and mail it to the factory. (See **Preface.**)

Unpacking And Initial Inspection: You are now ready to proceed with unpacking and installation of your OPTIMOD-FM.

Sometime during the life of your OPTIMOD-FM, you may wish to re-ship or return it. Since it is expensive and heavy, it is advisable to ship it only in the original packing materials which have been carefully designed to protect it. For this reason, it is wise to mentally note the method of packing and to save all packing materials.

If you might be returning it:

--Don't cut the grounding pin from the power cord (use the adapter provided if you must defeat the safety grounding provision);

--Set the unit only on soft, clean surfaces to prevent damage to painted or plated surfaces (a folded newspaper will do);

--Use the nylon-washered rack screws supplied to protect the panel from paint chipping.

Sage advice for repacking and reshipping your unit is contained at the end of **Appendix F.**

Various items are packed with OPTIMOD-FM:

(1) Line Cord
(4) 10-32x3/4" Rack Screws
(1) 3-wire AC Adapter (USA)
(1) Operating Manual
(1) 5/64" Allen Wrench (for front panel screws)
(2) Keys For Adjustment Door
(1) 24" BNC-to-BNC Cable (for composite output)
(2) 620-ohm ±5% 1/4-watt carbon film resistors
(for input termination, where required)
(1) Final Factory Qualification Test Results

Physical Examination: Perform a general inspection of the perimeter of the unit to check for obvious damage.

DAMAGE CLAIMS MUST BE MADE BY YOU AGAINST THE CARRIER IMMEDIATELY UPON DISCOVERY. Save packing and other evidence of damage for the carrier's inspector.

Set the unit on a flat, soft surface. Remove the three hex-socket screws at the top of the front panel using the wrench provided. The front panel, which is hinged at the bottom, will then tilt downward and reveal the interior. Look for IC's or other loose parts which may have fallen out during shipment.

Remove the subpanel through which the controls protrude by twisting the four DZUS fasteners 1/4 turn counterclockwise. Tilt the panel to remove it. This reveals the "card cage".

Various components are mounted in sockets for servicing convenience. It is possible (but improbable) that a component could be dislodged by heavy shocks in shipment.

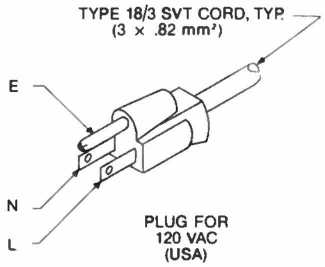
Starting at the left, using the card ejector tabs, carefully remove each card in turn, examine it, and replace it. Make sure that all components are properly seated in their sockets. Check with particular care to make sure that none of the can-type IC's are held in their sockets by one row of leads only.

Power Considerations: OPTIMOD-FM will operate on 115/230V ±15% 50-60Hz AC power. Without applying power to the line cord, turn the power switch ON and check the position of the LINE VOLTAGE SELECTOR switch. Units are shipped with this switch in the "115 Volt" position, unless labeled otherwise on the line cord. Adjust the selector switch so that the appropriate voltage is indicated. (If OPTIMOD-FM is installed within a transmitter, 230V may be the only power available.) Check the fuse, and replace with the following values if necessary:

115 VOLT: 1/2 Amp, 1/4 x 1 1/4 SLO-BLO (or 5 x 20 mm, type T)
230 VOLT: 1/4 Amp, 1/4 x 1 1/4 SLO-BLO (or 5 x 20 mm, type T)

AC connection to the chassis is made through an RF filter with IEC-standard mains connector. This filter is designed to meet the standards of all international electrical safety authorities, and leaks less than 0.5mA to the chassis when operated from 230V mains.

A U.S.A.-standard "U-ground" power cord is supplied to connect to the IEC socket, unless a different cord and connector is ordered. Users in other countries will ordinarily be able to easily obtain a power cord compatible with their country's standard. If you choose to cut the "U-Ground" plug from the cord and replace it with a plug appropriate to your standards, refer to Fig. 3-1 below.



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

Fig. 3-1: AC Mains Cord Detail

In areas where power lines are frequently struck by lightning, it may be advisable to connect voltage-dependent resistors (varistors) between each side of the line and earth, sized according to local experience.

Initialization Options: This section describes how to change certain operating characteristics of OPTIMOD-FM to suit your needs. If your needs correspond to the "factory-standard" characteristics, then no modifications need to be made, and you may skip to **Reassembly** below. **Appendix M** provides a quick summary of all 8100A/1 jumper options.

All modifications are made on the plug-in circuit cards. If the steps regarding physical inspection above have been followed, the cards are now readily accessible.

1) **Powerup Mode:** OPTIMOD-FM is shipped to power up in STEREO mode. To restrap so that it powers up in either MONO LEFT or MONO RIGHT mode, you must move a jumper plug on Card #7 according to Fig. 3-2.

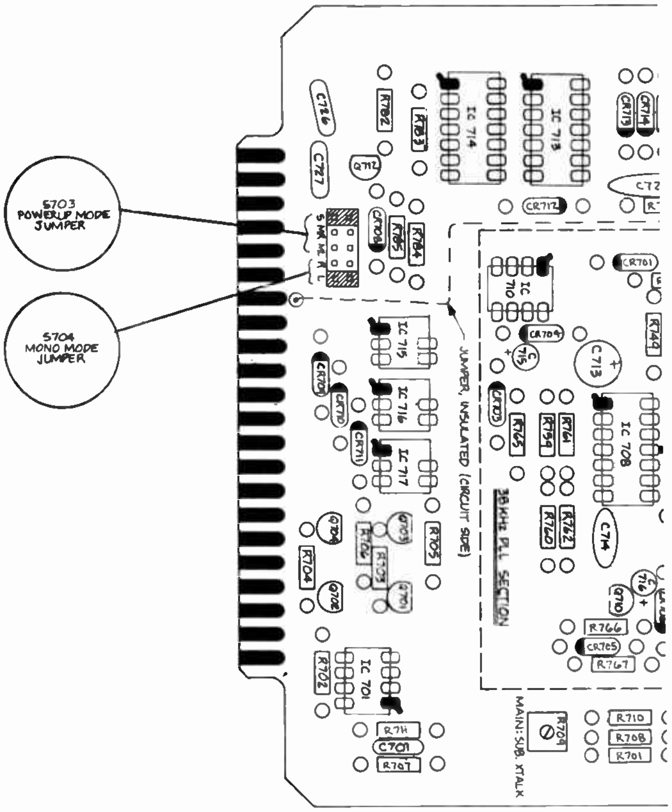


Fig. 3-2: Card #7 Powerup and Mono Mode Jumpers

2) **Mono Mode:** The front-panel STEREO/MONO switch ordinarily forces MONO LEFT mode when MONO is selected by this switch. If MONO RIGHT is desired, move a jumper plug on Card #7 according to Fig. 3-2.

3) **Input Attenuator Pads:** OPTIMOD-FM is shipped with 20dB pads ahead of the input buffer amplifiers. These are located on Card #3 (left channel) and Card #4 (right channel), and are suited for nominal input levels from -10 to +10dBm. If lower input levels from -30 to -10dBm are present, the pads must be defeated. To do this, remove Card #3. Reposition the jumper straps according to Fig. 3-3. Repeat for Card #4, and replace both cards.

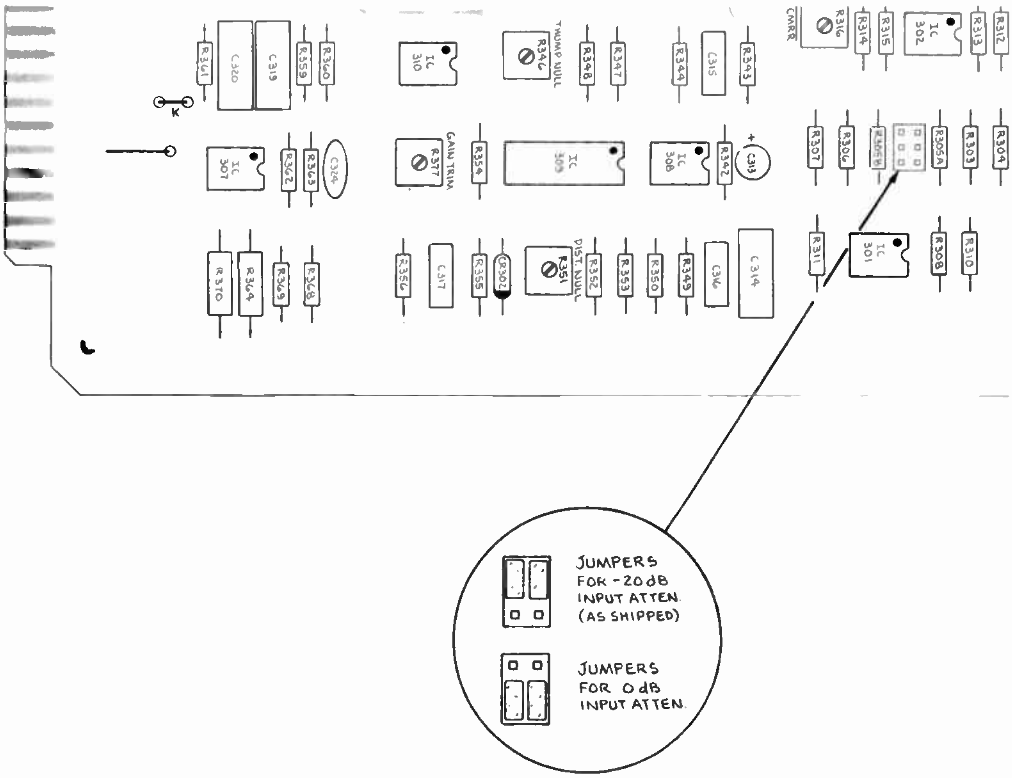


Fig. 3-3: Cards #3/#4 Input Attenuation Jumpers

4) **Defeating The 30Hz Highpass Filters:** There is no formal jumper for defeating these filters because we feel that the overall broadcast system will work better with the filters in for reasons discussed in detail in paragraphs **1.b** of **Appendix A.** We recognize that there are those who will disagree. Those who wish to defeat the filters can do so by connecting a jumper between pins 3 and 7 of IC302 (Card #3) and IC402 (Card #4).

Reassembly: When the physical examination, line voltage adjustment, and optional initialization procedures are completed, replace the subpanel. The subpanel, besides carrying knob identification and calibrations and holding the cards in place, also provides RF shielding for the cards. So, all four DZUS fasteners should be engaged by turning 1/4-turn clockwise.

The front panel may now be closed and fastened using the three hex-socket screws. Normally, all access required from now on can be achieved through the smaller access door (equipped with a key lock).

Initial Electrical Checkout: Plug the power cord into an outlet whose voltage corresponds to the setting of the internal LINE VOLTAGE SELECTOR switch. The unit should spring to life. Check to make sure that the following conditions occur:

A) The green POWER LED is illuminated;
B) The red GATE LED is illuminated;
C) The green STEREO LED is illuminated (unless you have restrapped Card #7 for MONO powerup);
D) Both yellow HF LIMIT LED's are off;
E) The VU meter readings are approximately identical to those provided in the "Final Factory Qualification Test Results" supplied separately from this Manual.

If anything is abnormal, repeat the **Physical Inspection** described above to make sure that you didn't miss anything. A preliminary diagnosis should be made, and, if necessary, the factory should be consulted.

If you wish to perform a more rigorous and complete checkout before installation, **Appendix** D **(Field Audit-Of-Performance Procedure)** provides complete instructions.

Equipment Location: OPTIMOD-FM is supplied in either single-chassis or dual-chassis versions. The dual-chassis version splits the system at the output of the Compressor. The studio chassis may be located in any convenient rack space in the studio. It is important to bear in mind that its RFI suppression is modest because it was assumed that the unit would be operated at a considerable distance from high-powered transmitters.

The main chassis, which is highly RFI-suppressed, is ideally located within 6 feet (1.8m) of the transmitter's exciter or composite STL transmitter. (The BNC/BNC cable supplied is 24" [61cm] long; longer cables must be supplied by the customer.) It requires 4 units (7", 17.8cm) in a standard 19" rack. If it is impossible to locate the OPTIMOD-FM chassis within 15 feet of the transmitter-mounted exciter, we recommend that you remove the exciter from the transmitter and mount it close to the OPTIMOD-FM chassis, rather than attempting to run a long baseband cable in a high RF field. Another advantage of moving the exciter away from the vibration of the transmitter blower is that microphonic effects will be reduced, thus improving signal-to-noise ratio.

If this is totally impractical, the OPTIMOD-FM stereo generator may be modified to drive higher capacitance. To do this, replace IC704 with an Analog Systems (P.O. Box 35879, Tucson, AZ 85740) type MA-332 operational amplifier and replace R739 with a 10-ohm ±5% 1/4-watt carbon film resistor. This will permit OPTIMOD-FM to drive up to 30 feet of RG-58A/U without problems due to capacitive loading. However, RF pickup in a cable of this length may prove problematical. (The MA-332 is pin-for-pin compatible with the stock AD-518 opamp and simply plugs into its socket.)

If the exciter is moved, be sure to use coaxial cable between exciter and IPA which properly matches the exciter output impedance. The IPA grid circuit should be carefully tuned for minimum VSWR between exciter and IPA.

Equipment life will be lengthened if the equipment is operated at moderate room temperatures and humidity, and if the air is reasonably dust-free and non-corrosive.

Although good audio monitor systems seem rare at transmitter sites, such a system which can be clearly heard at the OPTIMOD-FM will facilitate subjective adjustments. An alternative is the relocation of the controls to the studio location by use of the 8100A/ST Studio Accessory Chassis (see **Part 1.**)

Mounting And Grounding: It is important that the OPTIMOD-FM chassis (like all transmission equipment) be properly connected to a good earth ground. Wire is totally ineffective at VHF; the best way to ground the OPTIMOD-FM chassis is to mount it solidly in a well-grounded rack (or the transmitter cabinet). The rack or cabinet must be connected to earth through a wide, thin copper ground strap.

To assure good electrical contact between the OPTIMOD-FM chassis and the rack, it may be necessary to scrape the paint from the rack and/or the OPTIMOD-FM mounting flanges. Measure the resistance between the OPTIMOD-FM chassis and rack, and verify that it is less than 0.5 ohm.

Input Signal Connections: These instructions apply to the audio inputs of single-chassis OPTIMOD-FM's, and to the audio inputs of both studio and transmitter chassis in dual-chassis OPTIMOD-FM's.

In a high RF field, the audio input to OPTIMOD-FM must be fully-balanced, and should be run in 100% foil-shielded cable like Belden 8451. The shield should be connected to earth (chassis) ground at both ends. In addition, you should make sure that the telephone line termination box or STL receiver is properly grounded to earth.

In low-RF environments, the shield should be grounded at one end only, and audio may be run unbalanced over distances of less than 20 feet (6m).

OPTIMOD-FM should be operated with its integral 20dB input pad for levels between -10 and +10dBm, and without the pad for levels between -30 and -10dBm. Instructions for restrapping the pads are found above in **Initialization Options.**

The OPTIMOD-FM input is bridging, and its impedance is 200K with the 20dB pad defeated and 11.2K with the 20dB pad operative. If the source requires a 600-ohm termination (such as a telephone line), connect a 620-ohm ±5% 1/4-watt carbon film resistor across each audio input. Two such resistors are provided for your convenience.

It is important that both left and right audio inputs be in phase. This is ordinarily assured simply by connecting the red and black wires within all shielded cables symmetrically and consistently when wiring the two stereo channels. If a phasing error occurs, it will be indicated in on-air testing by failure of the OPTIMOD-FM L-R meter to null when OPTIMOD-FM is fed mono material, and by the stereo monitor's indicating more L-R than L+R level.

Studio Chassis Output Connections: In the dual-chassis version of OPTIMOD-FM, the output of the studio chassis presents a 600-ohm pure-resistive source impedance, balanced to ground, with a nominal output level of +10dBm when loaded by 600 ohms. It is thus suited for driving a land-line directly, or for driving the balanced input of a microwave STL transmitter.

If you wish to drive an unbalanced input, connect such an input between the studio chassis "+" output and circuit ground. **Do not ground the "-" output;** while no damage will occur, it will short the output of the "-" line amplifier to ground through a 300-ohm resistor.

Composite Output Connection: The composite output is capable of driving greater than 4V p-p into 10K. Its output impedance is 470 ohms, independent of the setting of the OPTIMOD-FM OUTPUT ATTEN control. We recommend that less than 6 feet (1.8m) of RG-58A/U cable be used to connect the exciter to OPTIMOD-FM, lest capacitive loading compromise the stereo performance due to frequency response rolloff and phase shift, or excessive RF pickup occur in the cable.

If a longer cable must be used, a low-capacitance coaxial cable such as RG-62A/U may be used for runs up to 15 feet (4.6m). Alternately, the stereo generator output can be modified as described in **Equipment Location** above. However, we suggest that the exciter be moved instead.

The composite output appears on a BNC connector. It is mounted on a metal plate which is insulated from the chassis by means of a thin polyester film. This assembly forms an RF-bypass capacitor of approximately 1000pF. The shell of the BNC connector is connected to OPTIMOD-FM circuit ground through an RFI filter.

The purpose of this arrangement is to enable unbalanced wideband exciter inputs to be driven without introducing hum-inducing ground loops. Different techniques must be used for interconnection, depending on whether the exciter input is balanced or unbalanced.

1) **Balanced Exciter Input:** (The most common of these include the Continental 510R-1 (Collins 310Z-2) with Continental 785E-1 Wideband Interface Card, the Gates TE-1 and TE-3 with Gates-supplied transformer-type Wideband Interface, and the Broadcast Electronics FX-30.) Connect the OPTIMOD-FM circuit ground and chassis ground terminals at the rear-panel barrier strip. This will create the required connection between exciter and OPTIMOD-FM circuit grounds.

2) **Unbalanced Exciter Input:** (This includes the Gates TE-1 and TE-3 with Orban ATE3-F Wideband Interface, and most other exciters. Check your exciter manual to be sure.) OPTIMOD-FM circuit ground will be automatically connected to exciter circuit ground through the shield of the baseband connector coax. Ordinarily OPTIMOD-FM circuit and chassis grounds will not be jumpered, as this will tend to create a ground loop. However, high RF fields often force reconsideration of conventional grounding rules, and if hum is a problem, you should try jumpering OPTIMOD-FM circuit and chassis grounds to see if hum is reduced.

Remote Control: Three sets of remote control terminals for selecting STEREO, MONO LEFT, and MONO RIGHT modes are located on the barrier strip on the OPTIMOD-FM rear panel. These are optically-isolated, RF suppressed, and may be floated ±50V above ground. Mode switching can be effected by applying a pulse as short as a few milliseconds to the appropriate terminals. Either AC or DC from 6 to 24 volts may be used. To use 48 volts, connect a 1K ±10% 2W carbon composition resistor to each terminal for current limiting.

IMPORTANT

> The life of the opto-isolators will be somewhat shortened if switching is effected by supplying continuous voltage (instead of a single pulse) to the terminals. If this is the only practical way to operate your particular remote control, we advise adding a series resistor to the remote control terminals to limit current to 10mA. If this is done, a life of many years can be expected.

If the remote control can provide voltage pulses from its internal power supply, this is the simplest means of activating the functions. The current requirement is approximately 1.9mA/volt. If the pulses are DC, be sure to observe the polarity indicated on the OPTIMOD-FM barrier strip.

If the remote control can provide only contact closures, then you can supply the +22V unregulated DC from OPTIMOD-FM through the contacts in the remote control to activate the functions. A suitably current-limited source of +22V is available on the OPTIMOD-FM barrier strip. If you choose this mode of operation, then connect all three "-" OPTIMOD-FM remote control terminals to chassis ground.

WARNING!

> Do not apply voltage to more than one set of remote control terminals at a time. Extreme overmodulation can result.

Remote Gain Reduction Meter: A negative voltage approximately proportional to the Total Master Gain Reduction is available between the OPTIMOD-FM rear-panel G/R terminal and ground. The voltage scale is approximately -0.33V per dB of gain reduction, and the source impedance is 8.87K. A standard 0-25dB Orban gain reduction meter can be connected directly between this terminal (-) and ground (+).

The Orban meter has a sensitivity of 1mA f.s. and a DC resistance of 880 ohms. Full-scale corresponds to 30dB G/R. Because only 25dB G/R can be achieved, the last 5dB of the scale is printed in red. The purpose of this is to match the scale to that of the BASS G/R meter, which is capable of, and fully calibrated to, 30dB G/R.

If an external meter with different characteristics is used, it is easy to calculate the required additional multiplier resistor for a 0-30dB scale by the formula: $M=(9.75/F)-(8870+R)$, where:

M is the required multiplier resistor in ohms,
F is the full scale meter sensitivity in amps, and
R is the internal DC resistance of the meter in ohms.

If M is negative, the meter you wish to use is not sensitive enough, or has too high an internal resistance.

If you wish to interface the G/R output to a remote control for telemetry, bear in mind that the input impedance of the remote control will load down the G/R output and reduce the voltage according to the gain factor: $G=X/(X+8870)$, where X is the input resistance of the remote control in ohms. The scaling of the remote control should therefore be -0.33xG volts per dB gain reduction.

PART 4:

Initial Setup Procedure

If you have a single-chassis OPTIMOD-FM, you may skip to **Stereo Generator,** below.

If you have a dual chassis OPTIMOD-FM, you must first align the gain of the STL and transmitter chassis to a standard to assure that both STL and transmitter chassis are driven at correct levels.

Dual-Chassis Alignment: This procedure is repeated twice: once for the left channel and once for the right. It is assumed that the STL is a pair of land-lines, a pair of microwave STL's, or a PCM link.

1) Adjust the operating controls on the studio chassis as follows:

Proof/Operate Switch:	OPERATE
L and R Input Attenuators:	0
Clipping:	+2
Release Time:	10
Bass Coupling:	10
Gate Threshold:	0
HF Limiting:	10
L and R Output Level:	Fully CW (up to 18 turns)

2) Connect an audio oscillator to the LEFT INPUT of the studio chassis. Set its frequency to 1kHz, and its output level to produce 10dB G/R as indicated on the studio chassis MASTER TOTAL G/R meter.

With the L and R OUTPUT LEVELs fully CW, the studio chassis will produce an output level on a 1kHz tone of 1.17Vrms (+3.6dBm) when loaded by 600 ohms. This is equivalent to 0VU in a +8dBm system when fed with program material.

3) Feeding phone lines: With the L and R OUTPUT LEVEL controls fully CW, the output level and impedance of the studio chassis are appropriate for directly driving a USA-standard telephone line requiring a nominal input level of +8dBm and a resistive balanced driving impedance of 600 ohms.

If your phone lines require a lower drive level to prevent clipping of audio by in-line amplifiers, reduce the OUTPUT LEVEL controls accordingly.

4) Feeding microwave systems: The frequency spectrum of audio at the output of the studio chassis approximates a 25us deemphasis. Therefore, an STL system with 25us deemphasis/preemphasis is most appropriate for maximizing signal-to-noise ratio while preventing clipping. Consult the STL manufacturer for information on converting STL preemphasis/deemphasis to 25us.

If the STL is un-preemphasized, or preemphasized at 25us, adjust the OUTPUT LEVEL of the studio chassis and/or the STL's input level to produce a level 9dB below 100% modulation.

If the STL is preemphasized at 50 or 75us, adjust the OUTPUT LEVEL of the studio chassis and/or the STL's input level to produce a level 15dB below 100% modulation.

5) Connect the output of the STL receiver to the LEFT INPUT of the OPTIMOD-FM transmitter (main) chassis. Place the VU meter FUNCTION switch in L COMPR OUT. Adjust the LEFT INPUT ATTEN on the OPTIMOD-FM transmitter chassis to make the VU meter read 100%.

Jumper Cards #3TX and #4TX are shipped with 20dB pads before the input amplifiers. If the reading is too low with the INPUT ATTEN fully CW, and the input pads are strapped for 20dB attenuation, restrap them for 0dB attenuation. This is done by removing Cards #3TX and #4TX from the chassis according to the instructions on p. C-1 of **Appendix C**, and by moving the jumpers according to Fig. 3-5. The cards and subpanel are then replaced.

If the reading is too high with the INPUT ATTEN fully CCW, and the input pads on Cards #3 and #4 in the transmitter chassis are strapped for 0dB attenuation, follow the same instructions to restrap the pads for 20dB attenuation.

6) Repeat steps (2) through (4) for the RIGHT CHANNEL.

Stereo Generator: From this point on, the procedure is identical for single- and dual-chassis units. "OPTIMOD-FM INPUT" means the input of the studio chassis in dual-chassis systems, and the main input in single-chassis systems.

1) Adjust the OPTIMOD-FM operating controls to the positions specified in step 1 of the **Dual-Chassis Alignment** section above. Apply a 1kHz tone to the left OPTIMOD-FM INPUT, and adjust the oscillator output level to produce 10dB TOTAL MASTER G/R.

2) Turn the 15-turn OPTIMOD-FM OUTPUT ATTEN control fully CCW (zero). Turn the OPTIMOD-FM HF LIMITING control to 10. Turn the OPTIMOD-FM PILOT switch OFF.

3) Turn on the carrier. Watch the TOTAL MODULATION meter on your stereo monitor, and turn the OPTIMOD-FM OUTPUT ATTEN control CW until the TOTAL MODULATION meter reads 68%.

4) Turn the OPTIMOD-FM PILOT switch ON, and adjust the OPTIMOD-FM PILOT LEVEL control until the monitor reads 9% on its PILOT LEVEL meter. TOTAL MODULATION should now read 75%. This procedure adjusts the OPTIMOD-FM output level to produce 100% modulation on program material, accurate to within a few percent.

5) Remove modulation, and listen to the demodulated carrier for abnormal hum, buzz or noise. If any of these are present, the problem should be fixed before proceeding further. In a dual-chassis installation, verify that the STL is not causing noise problems. Hints regarding OPTIMOD-FM/exciter interface are found in **Composite Output Connection** in **Part 3 (Installation).**

6) **Separation:** Connect a DC-coupled oscilloscope with at least 5MHz vertical bandwidth and triggered sweep to the WIDEBAND OUTPUT of your FM monitor. DO NOT USE AN ATTENUATOR PROBE; it may compromise the accuracy of the adjustment. Trigger the scope externally from the oscillator.

Turn the OPTIMOD-FM PILOT switch OFF. Continue to modulate the left channel with 1kHz. Adjust the scope's vertical sensitivity and sweep rate to produce a trace similar to Fig. 4-2. Note the flat baseline in Fig. 4-2, indicating ideal separation. Adjust the OPTIMOD-FM SEPARATION (L-R GAIN) control to secure a maximally-flat baseline. The vertical display should be expanded 10x to make the final adjustment.

CAUTION!

> Do not adjust separation by observing your stereo monitor. Most monitors are insufficiently stable to accurately indicate separation. The oscilloscope method specified above is the <u>only</u> satisfactory way to make this adjustment!

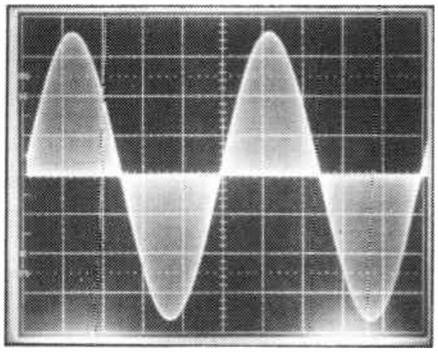


Fig. 4-2: Separation Trace

You should now measure left-into-right <u>and</u> right-into-left separation at 50, 1000, and 15,000 Hz to make sure that adequate separation is achieved through your system. The undriven OPTIMOD-FM input should be shorted or properly terminated to avoid crosstalk.

Separation can be approximately calculated from the scope trace by the formula: $S=20\log(D/P)$, where:

S is the separation (dB)
D is the peak-to-peak deviation of the baseline from flatness (volts)
P is the peak-to-peak level of the total baseband signal (volts)

Most separation problems are due to system problems or measurement error. If you cannot meet separation specifications, you should verify the performance of OPTIMOD-FM alone using the procedure in **a.6** of the **Stereo Generator** section in **Appendix D.**

Dried-out coupling capacitors in your FM monitor can cause failure to correctly measure 50Hz separation because excellent low frequency response and phase linearity are necessary to avoid distorting the signal upon demodulation. Similarly, if you have accidentally left your scope AC-coupled, it will cause measurements to be completely inaccurate at low frequencies.

Real separation problems can be caused by:

a) Incorrect phase adjustments in your exciter Wideband Interface.

b) Insufficiently wide frequency response or inadequate phase linearity in composite STL or exciter.

c) Mistuned or severely narrowband RF amplifiers and/or antenna.

7) **Pilot Phase:** Connect the oscillator to the right OPTIMOD-FM input. Switch the OPTIMOD-FM CROSSTALK TEST switch to SUB-TO-MAIN. Switch the OPTIMOD-FM PILOT switch ON.

You should see a trace on the scope like Fig. 4-3. If pilot phase is correct, the "tips" on this waveform will be perfectly horizontal, as in Fig. 4-3.

Expand the vertical scale of the scope by 10x, and expand the sweep to look more closely at the "tips", as in Fig. 4-4. Adjust the OPTIMOD-FM PILOT PHASE control until the tips are horizontal, as in Fig. 4-4.

Return the OPTIMOD-FM CROSSTALK TEST switch to OPERATE.

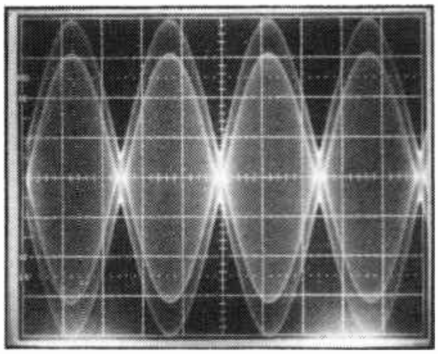


Fig. 4-3: Pilot Phase Trace

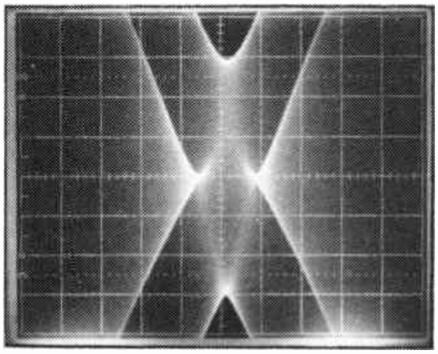


Fig. 4-4: Pilot Phase Trace, 10x

Program Tests: These listening tests are made with OPTIMOD-FM set up according to our recommended initial control settings. They are intended to detect obvious problems with audio quality which must be resolved before final adjustments are made. Once initial listening tests are passed, you can proceed to adjust OPTIMOD-FM setup controls according to format and competitive requirements.

a) Adjust OPTIMOD-FM controls according to Fig. 4-5. DO NOT adjust the OUTPUT ATTEN and INPUT ATTEN controls at this time. If you have a dual-chassis system, DO NOT READJUST THE TRANSMITTER CHASSIS INPUT ATTEN CONTROLS UNDER ANY CIRCUMSTANCES!

b) Play program material typical of your format. Set your console in MONO mode, such that both channels are putting out identical levels. Peak the console VU meters at 0VU.

c) Adjust the OPTIMOD-FM INPUT ATTEN controls (in a dual-chassis unit, on the STUDIO chassis) to "0". Advance the LEFT INPUT ATTEN until the MASTER TOTAL G/R meter reads approximately 10dB G/R.

d) Observe the L-R meter position (or the L-R stereo monitor meter in the case of a dual-chassis unit), and advance the OPTIMOD-FM RIGHT INPUT ATTEN until the meter nulls.

AC | CHR

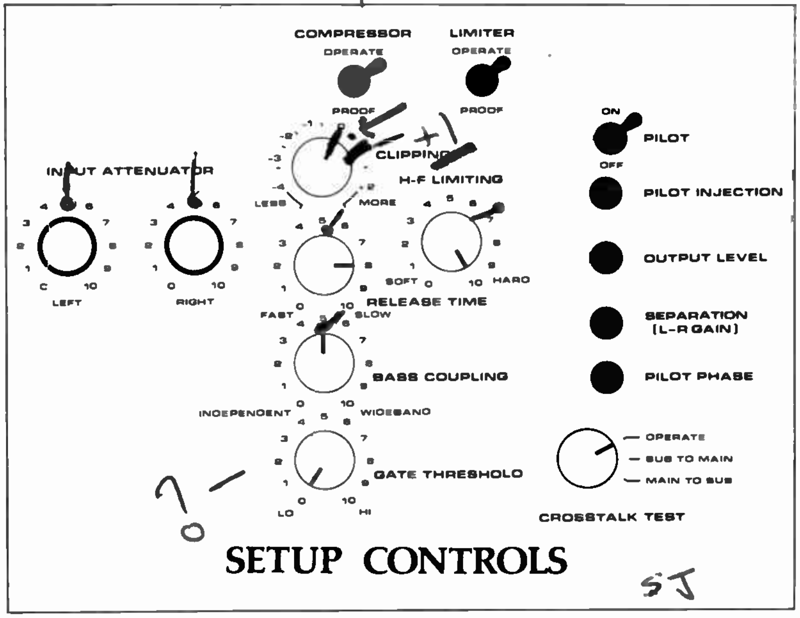


Fig. 4-5: Setup Controls

e) Place the console in STEREO mode. Observe the TOTAL MODULATION meter on the FM monitor. Make slight adjustments to the OPTIMOD-FM OUTPUT ATTEN as necessary to achieve desired modulation levels.

> Many FM modulation monitors more than a few years old exhibit problems with low-frequency tilt and high-frequency ringing. The LF tilt is caused by insufficient low-frequency response (LF response should be -3dB at 0.15Hz or below). High-frequency ringing is usually not as much of a problem.
>
> Tilt becomes a problem at the comparators that control the peak flashers. This can cause flashers to turn on when no overmodulation actually exists. LF tilt problems show up when the monitor is measuring program material, resulting in an indication of modulation that is higher than the actual percentage of modulation. This is true even though the monitor reads flat on sine waves from 50-15,000Hz. A 50Hz square wave can be used to test for tilt: you must connect the output of the square-wave generator to the composite input of the exciter to test the monitor. (If the exciter has LF tilt problems, you will see these in addition to any problems in the monitor.)
>
> Additionally, if an RF amplifier is used in the monitoring environment, any multipath picked up in the system will be indicated as additional modulation on the monitor (it probably will show on the peak flasher before it will be seen on the meter).

f) Observe the PILOT LEVEL meter on your stereo monitor, and adjust the OPTIMOD-FM PILOT LEVEL control as necessary to produce 9% pilot injection.

g) Listen to the audio quality of the air sound on a good monitor system, and verify that it sounds natural and free from noise and distortion. Comparing "AIR" and "PROGRAM" may reveal a bass increase in "AIR" due to the "hybrid" operation of OPTIMOD-FM as initially set up.

h) You may now proceed to **Part 5 (Operating Instructions)** of this Manual, and adjust OPTIMOD-FM's setup controls to your specific requirements.

PART 5:

Operating Instructions

This part describes how to adjust the OPTIMOD-FM setup controls to achieve a sound appropriate to your taste, format, and competitive situation. It is written so that both the Program Director and engineering staff can benefit.

Best results will be achieved if both Engineering and Programming 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 Programming fully understand the tradeoffs involved in optimizing one parameter, such as loudness, at the expense of others, such as brightness or distortion.

You've Got To Know Where You're Coming From Before You Can Decide Where You're Going.

So we'll start with some basic philosophy.

There is a spectrum of sonic preference, ranging from purist "it's got to sound just like the record", to "it's got to be the loudest thing in town." OPTIMOD-FM can be adjusted for either of these preferences, or for anything in between. About the only thing it can't be adjusted to do is to produce unnecessary pumping, gasping, wheezing, or other common processor artifacts which are the result of bad processor design, and which are not an essential part of any sound. There are three basic OPTIMOD-FM setups:

1) You can adjust it so that the output sounds as close as possible to the input at all times; or,

2) You can adjust it so that it still sounds "open", but more uniform in frequency balance (and often more dramatic) than the input; or,

3) You can adjust it so that it sounds dense, quite squashed, and very loud, so that it will jump out of car and table radios, but may be fatiguing and invite tune-outs on home hi-fi sets.

> IN ANY OF THESE SETUPS, THERE IS A DIRECT TRADEOFF BETWEEN LOUDNESS, BRIGHTNESS, AND DISTORTION. You can improve one only at the expense of the other two. (This is true of any processor.)

Perhaps the hardest part of adjusting any processor is deciding what this tradeoff is going to be. We feel that it is usually wiser to give up ultimate loudness to achieve brightness and low distortion. A listener can compensate for loudness in a fraction of a second by adjusting his volume control. But there is nothing -- not one thing -- he can do to make a dirty signal sound clean again, or to undo the effects of excessive high frequency compression. 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 openness, depth, and "punch" -- even on small radios -- which cannot be obtained when the signal is excessively squashed.

Because of these requirements, it is vital that the station be equipped with a monitor system which is sufficiently good to fully reveal the quality (or lack of same!) of the airsound. In too many stations, the best monitor is significantly inferior to the receivers found in many listeners' living rooms! **Production Practices** in **Appendix K** discusses in some detail how to efficiently create an accurate monitoring environment.

Appendix K also discusses how the audio plant can be brought up to state-of-the-art quality. Barely meeting the old FCC Proof of Performance requirements is not nearly enough!

If women form a significant portion of the station's target 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 ratings (such as "beautiful"), greatest care should be used not to subconsciously "turn-off" women by excessive stridency, harshness, or distortion.

BE PARTICULARLY SENSITIVE TO THE PROBLEMS OF LISTENING FATIGUE AND ITS EFFECT ON QUARTER-HOUR MAINTENANCE. DON'T INVITE TUNEOUTS BY OVERPROCESSING! REMEMBER THE DECLINE AND FALL OF AM; DON'T DRIVE LISTENERS TO THEIR RECORDS AND TAPES BY DESTROYING THE AUDIO QUALITY OF FM, TOO!

Functions Of The Setup Controls: This information is basic to all further discussions, and must be well-understood if you are to succeed in getting the sound you want.

1) **Input Attenuators** (Left and Right): Determine the amount of gain reduction (for a given audio level going into the processor), and therefore how much the loudness of soft program material is increased.

2) **Release Time:** Determines how fast the gain of the "Master" compressor increases when the program material gets soft. Mostly, this control affects the density of the sound, which is a fundamental fatigue-determining factor. There is a clearly "optimum" setting (8) which gives the most natural sound and least detectable compression -- although not the most loudness.

3) **Clipping:** Determines how much of the signal is peak-limited by clipping -- a distortion generating process. The loudness/distortion tradeoff is primarily determined by this control.

4) **High Frequency Limiting:** Determines a tradeoff: the degree to which the highs are controlled by filtering (which can make them dull), or by clipping (which can make them distorted).

5) **Bass Coupling:** Determines if the compressor will operate "wideband", completely "multiband", or somewhere in between. In "wideband" mode, the air sound is most faithful to the sound of the original record; in "multiband" mode, bass balances are more uniform from cut to cut and bass is often increased.

6) **Gate Threshold:** Determines what input level the system considers "noise". Below this level, the release time of the compressor is slowed by a factor of 50 or so to prevent "breathing".

IMPORTANT: OTHER CONTROLS MUST BE ADJUSTED ONLY BY INSTRUMENT. IF THEY ARE MISADJUSTED, THE SYSTEM MAY NOT COMPLY WITH FCC RULES.

A reference chart relating control adjustments for desired sonic characteristics is found at the end of this Part.

Start At The BASS COUPLING Control: Everything else depends on how you decide to adjust it.

-- "WIDEBAND" mode (BASS COUPLING close to 10) makes the output sound most like the input. It is most useful for Fine Arts, Classical, Beautiful, and "purist" MOR and AOR formats. It is not suited for fast release times and ultimate loudness. As long as you use the "optimum" release time (8), it sounds good even with large amounts of compression.

Because setting the BASS COUPLING control at "10" will sometimes cause a bass loss (because the "Bass" compressor can never take more gain than the "Master" compressor, and will sometimes take less), the most accurate frequency balance will often be obtained with this control between "7" and "10". The setting depends of the amount of gain reduction and the setting of the RELEASE TIME control. The control is most readily adjusted by watching the BASS G/R and COMPRESSOR G/R meters, and adjusting the control until the meters track as closely as possible.

-- "MULTIBAND" mode (BASS COUPLING at 0) is appropriate for most other formats and processing styles:

With the "optimum" release time (8), it sounds very open, natural, and non-fatiguing. It will provide a bass boost on some program material which is bass-shy. This mode sounds fine even with large amounts of compression, and is suited for CHR, Jazz, Black, and the more aggressive MOR and AOR formats.

With fast release times (close to "0"), the MULTIBAND mode provides maximum loudness and density on small radios. However, it may fatigue listeners with high-quality stereos, and it requires more careful operator gain riding than slower release times do. Potential formats (in markets where competitive loudness is important) include CHR and Black.

-- HYBRID WIDEBAND/MULTIBAND mode (BASS COUPLING between 2 and 7) compromises between the two sounds. This mode is useful if you feel that the MULTIBAND mode provides too much bass for your taste -- you can use the BASS COUPLING control to adjust the balance between bass and the rest of the spectrum. This HYBRID mode is ordinarily used with the "optimum" release time setting (8).

Adjusting The Other Controls: Once you have chosen your adjustment of the BASS COUPLING control, you must adjust the rest of the controls to finish "customizing" your sound. MAKE ADJUSTMENTS JUDICIOUSLY, AND LISTEN CAREFULLY TO ALL TYPES OF PROGRAM MATERIAL WITHIN YOUR FORMAT.

Clipping: This is the prime control over the loudness/distortion tradeoff. In our opinion, "0" is the best setting, giving results that sound "undistorted" even on high quality receivers -- provided that input program material is clean and that the "optimum" release time setting (8) is used. If faster release time settings are used, or if input program material is somewhat distorted, the CLIPPING control should be turned down.

Particularly in WIDEBAND mode, the CLIPPING control can sometimes be turned up if you are operating with small amounts of gain reduction, slow release times, and ultra-clean program material.

Ultimately, your ears must judge how much distortion is acceptable. But use worst-case program material like live voice and piano to make your final decision.

Release Time And Amount Of Gain Reduction: The action of the RELEASE TIME control has been optimized for resolution and adjustability. Compared to the similar control in the old 8000A, the 8100A/1's control provides more resolution in the "fast" part of the scale, stretching the range equivalent to the 8000A's "Fast-to-Limit Only" all the way from "0" to "5". Consequently, the "Slow" range (equivalent to the 8000A's "12:00-to-Full Slow") occurs between "8" and "10".

When the "optimum" release time setting (8) is used, the amount of gain reduction is surprisingly non-critical. (The amount of gain reduction is controlled both by the setting of the INPUT ATTENUATOR controls and where the console VU meter is being peaked.)

The gating feature prevents noise from being brought up during short pauses, and pumping does not occur at high levels of G/R. So the primary danger of using large amounts of G/R is that the level of soft passages in wide-dynamic-range input material may eventually be increased unnaturally. (In case you were wondering: OPTIMOD-FM uses gain-freezing gating instead of expansion so that the unit cannot "go to sleep" on long, quiet musical passages, turning them down into inaudibility. Expansion is never necessary to achieve noise reduction because the amount of gain reduction can be reduced instead.)

Release becomes faster as gain reduction is increased between 0 and 10dB, making the program progressively denser and creating a sense of increasing loudness. This preserves some feeling of dynamic range, even though peak levels are not increasing. Once 10dB of gain reduction is exceeded, no further increase in short-term density occurs as more gain reduction is taken. 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.

It is important to realize that the processing in the 8100A/1 was designed to perform both a compression and limiting function. Full loudness is not achieved until approximately 10dB of gain reduction is used because of the gradual transition that occurs between no gain reduction and 10dB gain reduction. Usually, the smoothness of the control makes 10dB of gain reduction quite acceptable (and desirable) even for "beautiful" or classical formats.

Regardless of release time setting, we feel that the optimum amount of gain reduction in popular music formats is 10-15dB. If less gain reduction is used, loudness can be lost (as explained above), although this can be partially compensated by moderately speeding up the RELEASE TIME control and advancing the CLIPPING control.

When OPTIMOD-FM is operated with fast release times, the sound will change substantially as more gain reduction is used. This means that operator gain riding is more critical, and also that you must decide on the basis of listening tests how much gain reduction gives you the dense sound you want without a feeling of overcompression and fatigue.

Unlike the metering on some familiar processors, the red in the OPTIMOD-FM gain reduction meter means business! When the meter is in the red, it means that the compressor has run out of gain reduction range, that the circuitry is being overloaded, and that various nastinesses are likely to commence. Because the compressor has 25dB of gain reduction range, this problem should never occur if OPTIMOD-FM has been set up for a sane amount of gain reduction under ordinary program conditions. But beware the different peak factors on voice and music -- if voice and music are peaked identically on a VU meter, voice may cause up to 10dB more peak gain reduction than does music!

High Frequency Limiting: This control trades off distortion against high frequency loss. When the control is moved toward "soft" (more hf limiting), the sound will become duller but less "gritty". When the control is moved toward "hard", the sound will become brighter, but more gritty and "smeared".

Because the clipper in OPTIMOD-FM cancels distortion at low frequencies, the HF LIMITING control will have a different effect on clipping distortion than you might expect. Gross breakup (principally sibilance splatter) will not occur, and you must listen to the upper midrange and the highs to hear the effect of the clipper. Highly equalized hi-hat cymbals is good program material to illustrate the effect of adjusting the control.

When the CLIPPING control is operated at "0" or below and the "optimum" release time setting is used, it is possible to operate the HF LIMITING control at "10" (full hard) without objectionable distortion, provided that the program material is super-clean. If the CLIPPING control is operated beyond "0" and/or faster release times are used (such that greater level and density is produced), it is usually necessary to readjust the HF LIMITING control closer to "soft" to avoid objectionable distortion. Fortunately, the hf limiter "knows" that greater density and level have been produced when these other controls are operated more aggressively, and most of the necessary increases in hf 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 tradeoff discussed above. (Engineering aside: the main reason for readjusting the HF LIMITING control is to compensate for the HF limiter's relatively low compression ratio.)

Gate Threshold: The GATE THRESHOLD control is adjusted so that noise is not brought up during short pauses.

The gain will eventually recover to maximum even when the compressor is in a gated condition, but recovery is so slow that it is essentially imperceptible. This is to avoid the compressor's getting "stuck" on a long, low-level musical passage immediately following a loud passage which has caused a great deal of gain reduction.

It is not unusual to operate the GATE THRESHOLD at "0". Higher settings are primarily useful for stations doing radio drama, sports remotes, and other non-musical programming. Slightly higher settings may also increase the musicality of the compression by slowing down recovery on moderate- to low-level musical passages. If 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 mechanism effectively slows down the release time as the input gets softer and softer, thus preserving musical values in wide-dynamic range material, like classical.

Why Certain Controls Aren't There: If you are used to adjusting conventional triband systems, you may wonder why OPTIMOD-FM does not have certain controls which are available on the triband.

A triband, for example, usually has a THRESHOLD and GAIN control on its bass compressor. The GAIN control can create a fixed bass equalization, and the THRESHOLD determines the average amount of bass produced by the system.

We decided early on that OPTIMOD-FM was <u>not</u> going to be an equalizer. In FM, this can get you into far too much trouble, since it assumes that <u>every</u> record is incorrectly equalized. This is clearly absurd. <u>Some</u> records (mostly older ones) are wrong, but these should be fixed in the production studio. Today, <u>most</u> records are right because of improved monitoring practices in recording studios.

Adjusting a conventional triband THRESHOLD control to produce bass that is balanced to your taste involves serious compromises, because it usually results in excessive reduction of heavy bass that is <u>supposed</u> to be there to make a musical point. A better solution is the use of OPTIMOD-FM's BASS COUPLING control. This can control bass balances <u>without</u> unnecessarily reducing bass punch.

Also missing from OPTIMOD-FM are any attack and release time adjustments for the high and low frequency bands. The reason is simple: there is a clearly optimum choice for these time constants, and making them adjustable would simply be an invitation to trouble.

Finally, there is no HIGH BAND GAIN control which would permit OPTIMOD-FM to be used as an high frequency equalizer. The argument for avoiding such a control is even stronger here than it is for avoiding it in the bass band. The ear is far more sensitive to the frequency balance between midrange and highs than to the balance between midrange and bass. So not only have we made it impossible to use OPTIMOD-FM as a fixed high frequency equalizer, but the OPTIMOD-FM high frequency limiter operates such that it can never <u>increase</u> the highs. The conventional triband structure which permits the high band to take more gain than the midband always, to our ears, results in the high end's having an artificial, hyped, "phony" quality. There is no substitute for a pair of real, live human ears correcting high-end frequency balances in the production studio -- there is no "automatic equalizer" that passes critical listening tests!

To achieve a particular sound, some stations boost highs and lows with a parametric equalizer before the audio signal is fed to the OPTIMOD-FM. The 8100A/1 handles this well, but we recommend high-end preprocessing be done in moderation (3-4dB equalization) to avoid further increase in overload distortion and clipping which could result from highly-preprocesssed material being reprocessed to match the 8100/1's preemphasis curve.

SUMMARY

The basic FM medium is capable of very high quality, and many listeners are equipped to receive and appreciate such quality when it is broadcast.

OPTIMOD-FM is capable of an outstandingly advantageous tradeoff between loudness, brightness, and distortion. This tradeoff is fundamental, and must be made in a wise and artistic way.

The OPTIMOD-FM setup controls are functionally independent. However, changing one usually requires that others be readjusted to rebalance the loudness/brightness/distortion tradeoff. An explanation of how this is done was provided above.

Bear in mind that poor-quality program material automatically forces a poorer tradeoff than could otherwise be obtained. So any effort expended in cleaning up the audio chain prior to OPTIMOD-FM will pay extra dividends. **Appendix** K discusses (in engineering language) many of the things that must be done to obtain audio quality that assures that OPTIMOD-FM will be operated to its best advantage.

Never lose sight of the fact that loudness is the only one of the three fundamental factors which is easily compensated by the listener. If the others are permitted to audibly degrade the sound of the original program material, then the signal is irrevocably contaminated and the original quality can never be recovered.

GETTING THE SOUND YOU WANT

ALWAYS START WITH OUR SUGGESTED INITIAL SETTINGS (SEE FIG. 4-5) AND WORK FROM THERE.

— To obtain more loudness

1. Operate "multiband" (BASS COUPLING at "0") with fast release times. Turn down CLIPPING and H-F LIMITING as necessary to avoid objectionable distortion.
2. Clean up audio. Super-clean audio can be processed harder without objectionable side-effects.
3. Use SCA Protection Filter (card #0).
4. Use 8100A/XT Six-Band Limiter Accessory Chassis.

— To obtain more brightness

1. Turn the H-F LIMIT CONTROL fully clockwise. To avoid objectionable distortion with fast RELEASE TIME, you may have to turn down the CLIPPING control. This will further increase brightness at the expense of loudness.
2. Be sure that program material is properly equalized, and that STL is flat to 15kHz (see **Appendix K).**
3. Use Orban 622B Equalizer ahead of 8100A/1.

— To obtain more bass

1. Operate the BASS COUPLING control towards "0".
2. Use Orban 622B Equalizer ahead of 8100A/1.

— To obtain less bass (retaining original program material balance)

1. Operate the BASS COUPLING control towards "10".

— To make "Air" sound most like "Program"

1. Operate with the BASS COUPLING close to "10". (Adjust the control to make the BASS and COMPRESSOR G/R meters track as closely as possible.)
2. Operate with the RELEASE TIME at "8" (optimum).
3. Use lesser amounts of gain reduction by backing off the INPUT ATTENUATORS.
4. Minimize the amount of clipping and h-f limiting by operating H-F LIMITING at "10" (full hard), and backing off the CLIPPING as far towards "0" as required to avoid audible distortion on difficult material like male voice or piano.

— To obtain "open" sound with no audible compression

1. Operate the RELEASE TIME control at 8.
2. Do not pre-compress program material in the production studio.
3. Use relatively small amounts of gain reduction. (This may allow you to advance the CLIPPING control to compensate for loudness loss.)

— To obtain a "heavily-processed" sound

1. Operate the RELEASE TIME control at "0" and the BASS COUPLING control at "0". (You may have to back off the CLIPPING and H-F LIMITING controls to avoid objectionable distortion. D.J. gain riding will also become more critical.)
2. Use 8100A/XT Six-Band Limiter Accessory Chassis.

— To avoid "noise pump-up"

1. Operate with smaller amounts of gain reduction.
2. Adjust the GATE THRESHOLD more clockwise.
3. Use slower RELEASE TIME.

— To achieve more subtle gain riding in wide-dynamic range material

1. Critically adjust the GATE THRESHOLD control so that medium- to low-level passages cause the GATE lamp to flash on and off, thus slowing down the release time as the music gets softer.

— To avoid excessive sibilance (particularly on women's voices)

1. Use an Orban 536A Dynamic Sibilance Controller on the microphone chain only. (While the 8100A/1 will not distort sibilance, its excellent h-f power handling will result in its passing high-energy sibilance present at its input, instead of limiting it.)

PART 6:

System Performance Verification

The FCC (USA) has eliminated requirements for periodic Proof-of-Performance measurements. However, performance standards specified in the FCC Rules must still be met. Many stations will still wish to make periodic equipment performance measurements. The text below provides the general information which is needed to perform measurements verifying the performance of a transmission system including the 8100A/1. Instructions for bench-top verification of 8100A/1 performance outside of the transmission system are found in **Appendix D: Field Audit-of-Performance.**

Mono Performance Verification: This is totally straightforward. Merely enter the MONO LEFT or MONO RIGHT modes, switch both PROOF/OPERATE switches to PROOF, and drive the appropriate OPTIMOD-FM input with test signal. Sufficient headroom is available to modulate well beyond 100% at all frequencies from 50-15,000Hz.

NOTE

> OPTIMOD-FM frequency response drops off extremely rapidly above 15.0kHz. If the test oscillator is miscalibrated, OPTIMOD-FM may appear not to meet proof at 15.0kHz. Before blaming OPTIMOD-FM, measure the output frequency of the test oscillator with an accurate counter to make sure that it is actually producing 15.0kHZ, and not some slightly nigher frequency.

Stereo Performance Verification: As of this writing, the law does not require that these measurements be made and be on file. However, the station is required to meet these performance specifications, and many stations therefore make these measurements as part of a routine performance verification.

Part 73.322 of the FCC Rules refers to the performance of the transmitter only (starting with stereo generator input terminals), and measurements may be made by connecting the test oscillator directly to the OPTIMOD-FM main audio inputs. All stereo measurements are made with both OPTIMOD-FM PROOF/OPERATE switches in PROOF. Following is an outline of the required measurements and how to perform them.

1) **Main Channel:** The main channel (L+R) must meet all mono requirements for frequency response, total harmonic distortion, and noise. Compliance may be verified by driving both OPTIMOD-FM main inputs in-phase, slightly adjusting the right INPUT ATTEN (studio chassis in dual-chassis versions) to null the L-R meter on your stereo monitor, and then using the L+R meter of your stereo monitor for measurement. If L-R fails to null below -20dB, suspect a differential phase error between the left and right channels. 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 can be caused by certain failures in the phase correctors located on Cards #6, #8, and #9. (See **Appendix F** for troubleshooting information).

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.

2) **Subchannel:** Mono requirements for frequency response, harmonic distortion, and noise must also be met for the stereo subchannel (L-R). L-R can be generated by reversing the polarity of the oscillator connection to the OPTIMOD-FM right audio input only, and by slightly trimming the OPTIMOD-FM right INPUT ATTEN (on the studio chassis in dual-chassis units) to null the L+R meter on your stereo monitor.

Measuring L-R noise is particularly problematical because most stereo monitors have no provision for applying deemphasis 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:

$$S = 10 \mathrm{x} \log(10^{(L/10)} - 10^{(M/10)}), \text{ where:}$$

S is the L-R noise in dB
L is the left or right channel noise in dB
(assuming L and R noise measurements are almost equal)
M is the L+R noise in dB

3) Careful reading of 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, 50-15,000Hz, left-into-right and right-into-left.

IMPORTANT

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. It is particularly important not to (mis)realign the OPTIMOD-FM stereo generator to compensate for a misaligned stereo monitor. In general, the only stable and reliable way of aligning the OPTIMOD-FM stereo generator for correct separation is the oscilloscope baseline method described in section **a.**6 of **Stereo Generator** in **Appendix** D of this Manual.

Pilot phase also affects separation. Pilot phase should be verified according to section **a.**7 of **Stereo Generator** in **Appendix** D. This method is more accurate than use of your stereo monitor.

4) Crosstalk: Measurement of main-channel-to-subchannel and subchannel-to-main-channel crosstalk is facilitated by the OPTIMOD-FM's internal CROSSTALK TEST switch. To make these tests, simply drive the OPTIMOD-FM right audio input, switch the OPTIMOD-FM CROSSTALK TEST switch to the appropriate mode, and read crosstalk on your stereo monitor. (The CROSSTALK TEST switch applies the output of the right channel audio processing directly to either the main channel or subchannel stereo generator input, and scales internal gains appropriately in the stereo generator to keep total modulation constant.)

NOTE

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.

5) 38kHz Subcarrier Suppression: Using the same setup as in **Crosstalk,** above, enter the SUB-TO-MAIN mode using the OPTIMOD-FM CROSSTALK TEST switch. Modulate the carrier to 100% using 7.5kHz, and read the 38kHz suppression on your stereo monitor.

NOTE

The two CROSSTALK TEST modes in OPTIMOD-FM will cause slight internal offset changes which will translate into somewhat poorer 38kHz suppression than that provided by the normal OPERATE mode. However, the suppression should never deteriorate even close to the -40dB legal limit.

6) Pilot Frequency: This is most conveniently measured by opening the access door and connecting the frequency counter input across two terminals (Fig. 6-1) located on the P.C. card mounted on the rear of the rotary switch to the left of the access door opening.

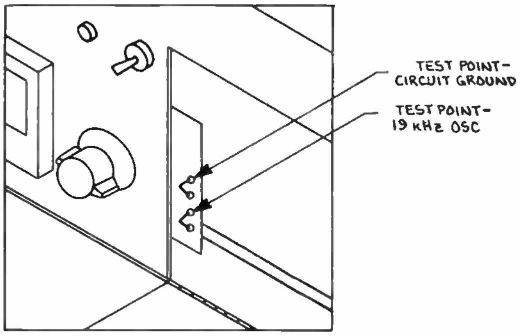


Fig. 6-1: Pilot Test Point

7) **Pilot Injection:** This is straightforwardly measured on your stereo monitor.

Rear-Panel TEST Jacks: The inputs of the stereo generator are available on the RCA phono jacks on the rear panel of OPTIMOD-FM when the rear-panel NORMAL/TEST switch is in TEST. (When the switch is in NORMAL, the output of the audio processing appears at these jacks.)

These inputs are unbalanced, apply no preemphasis, and require approximately 3.3V rms to produce 100% modulation, including 9% pilot injection.

To produce proper operation of the stereo generator, these jacks must be driven by a voltage source such as that produced by the output of an opamp. The 600-ohm output of a typical oscillator is too high-impedance to produce correct operation. IF THE SIGNAL SOURCE IS CONNECTED TO ONLY ONE JACK, THE OTHER MUST BE GROUNDED TO PRESERVE CORRECT STEREO GENERATOR OPERATION.

Orban Associates Inc., has been providing schematics upon request for construction of a separation-and-crosstalk test fixture for its older model OPTIMOD-FM, the 8000A. If you have already built such a fixture, be assured that it is also appropriate for driving the Model 8100A/1 test jacks. However, the 8100A/1's internal CROSSTALK TEST modes are usually much more convenient to use.

PART 7:

Routine Maintenance

OPTIMOD-FM is a highly stable device which uses solid-state circuitry throughout. Recommended routine maintenance is minimal.

1) Keep the outside of the unit clean. If the panel becomes dirty, it can be washed with a mild household detergent and water. Stronger solvents may damage plastic parts, paint, or the silkscreened lettering, and should not be used.

2) Particularly in humid or salt-spray environments, check periodically for corrosion around metal-to-metal contacts such as the audio and control wiring, and those places where the OPTIMOD-FM chassis contacts the rack.

3) Check for loss of grounding due to corrosion or loosening of rack mounting screws.

4) Familiarize yourself with the normal VU meter readings, and with the normal performance of the G/R meters. If any meter reading becomes abnormal, refer to **Appendix F (Trouble Diagnosis).**

5) A good ear will pick up many failures. Familiarize yourself with the "sound" of OPTIMOD-FM as you have set it up, and be sensitive to changes or deteriorations. But if problems arise, please don't blame OPTIMOD-FM by reflex. Refer to **Appendix F** for systematic troubleshooting instructions which will also help you determine if the problem is in OPTIMOD-FM or is somewhere else in the station's equipment.

ROUTINE PERFORMANCE VERIFICATION

This procedure can be performed very quickly, and provides tests of some of the more important OPTIMOD-FM performance parameters. A much more thorough and rigorous procedure is provided in **Appendix D (Field Audit-of-Performance Procedure).**

Stereo Generator Tests: These tests are made with normal program material, and can therefore be performed in seconds, without seriously interrupting normal programming.

1) **Dynamic Separation:** With bright music playing, suppress one of the two stereo input channels to OPTIMOD-FM, and observe the suppressed channel's meter on your stereo monitor. Ordinarily, the indication will be better than 45dB below 100% modulation.

Restore the suppressed channel, and repeat the test for the other channel.

If the undesired crosstalk into the "dead" channel sounds clean and distortion-free, this probably means that the SEPARATION adjustment on the stereo monitor has drifted, and that the problem is not actually in the transmission system. This should be verified by repeating the test with another monitor or high-quality tuner in WIDEBAND mode. If the problem is observed on more than one receiver, the OPTIMOD-FM stereo generator has probably drifted, and the cause of the drift should be investigated.

If the crosstalk sounds highly distorted (particularly if distortion is worst when considerable high frequency energy is present on the other channel), the distortion may be due to aliasing. If the problem occurs only in one direction (say, left-into-right), then the OPTIMOD-FM FCS Overshoot Corrector circuitry should be investigated. If the problem occurs symmetrically in both directions, check for clipping or severe non-linearity in exciter, composite STL, or the OPTIMOD-FM stereo generator.

2) **38kHz Suppression:** Briefly interrupt programming (or wait for a short pause), and observe the 38KHZ position on your stereo monitor. Verify that suppression is well below -40dB.

3) **Pilot Injection:** Measure this routinely on your stereo monitor and verify that it is between 8% and 10% modulation.

Audio Processing: There are no effective, quick instrument tests that can be made using ordinary program material. Your ear is the best test instrument here.

If a minute or so can be spared from normal programming, the "standard level" test can be made using a sinewave input. This is done as follows:

1) Record the settings of the CLIPPING, BASS COUPLING, RELEASE TIME, and H-F LIMITING controls so that they can be restored when you have completed the test.

2) Set the OPTIMOD-FM controls to the following "standard" settings:

Proof/Operate Switch:	OPERATE
CLIPPING:	+2
RELEASE TIME:	10
BASS COUPLING:	10
H-F LIMITING:	10

3) Drive the OPTIMOD-FM left channel (probably through a console input) with a 1kHz sinewave. Adjust the oscillator level until the OPTIMOD-FM MASTER TOTAL G/R meter reads 10dB G/R.

4) Verify that the OPTIMOD-FM L COMPR OUT VU meter switch position causes the meter to read 0VU, ±0.5VU., and that the OPTIMOD-FM L FILTER OUT meter position causes the meter to read 0VU, ±1.0VU.

5) Repeat steps (3) and (4) for the RIGHT channel.

6) Restore the OPTIMOD-FM setup controls to their normal settings.

Failure to produce these standard levels indicates a failure somewhere within the audio processing circuitry. Refer to **Appendix F (Trouble Diagnosis).**

APPENDICES

APPENDIX A:

System Description

The purpose of this Appendix is to provide the installing engineer with an overview of system design, to answer questions and deal with uncertainties about various unconventional aspects of the design, and to provide the service technician with a moderately detailed overview of the system.

Each card is numbered. Reference will be made in each section to the number of the card on which the described circuitry is located.

The paragraphs in **Appendix** B **(CIRCUIT DESCRIPTION)** that correspond with topics in this Appendix have identical numbers and titles in order to expedite access to further information on a topic of interest.

REFER TO THE BLOCK DIAGRAM (page J-21)

1.a) Input Amplifier: (on Cards #3 and #4)

The audio is applied to an RFI suppression network and pad, the latter strappable for 0 or 20dB attenuation. The RFI-suppressed audio is then applied to a low-noise true instrumentation amplifier, whose "+" and "-" inputs are symmetrical and high impedance. The gain of this amplifier is adjustable from 0.88 to approximately 10.5 (a 21.5dB range). If this range does not yield the desired amount of gain reduction, then the input pad should be restrapped.

In order to avoid distortion due to imperfections in the large-value coupling capacitors that would be necessary, the input is DC-coupled. Therefore, only small amounts of differential DC should be applied to the input. Ordinarily, the input is fed by the output of a transformer or capacitively-coupled amplifier, and no difficulty should arise. Slight amounts of DC offset are eliminated in the 30Hz highpass filter following the input amplifier.

1.b) 30Hz Highpass Filter: (on Cards #3 and #4)

The output of the input buffer is applied to a third-order Chebychev highpass filter with 30Hz cutoff frequency (0.5dB down) and 0.5dB ripple. Unlike the identical filter in the old Model 8000A, this filter is not conveniently bypassable. It was felt that the advantages of this filter (i.e., elimination of modulation-wasting subsonic energy from turntable rumble and other sources, elimination of subsonic energy's introducing distortion by modulating the compressor control voltages, and prevention of destabilization and/or distortion introduction in exciters' AFC's) merited the filter's inclusion as a standard part of the system.

The cutoff frequency of the filter is sufficiently low that the only commonly-found musical instrument producing lower fundamental frequencies is the pipe organ. Most records cut off at 30Hz, and no rock-and-roll instruments have fundamentals below 40Hz.

The ringing introduced by the filter is insignificant. The ear is very insensitive to ringing in this frequency range. Further, the ringing is comparable to that introduced by a well-designed vented box loudspeaker with 30Hz cutoff.

If there seems to be an on-air problem with bass response, please don't blame this filter! First investigate such problems as obviously measurable bass rolloff in the chain up to OPTIMOD-FM 8100A/1, excess numbers of transformers in the audio chain, non-linear group delay in phone lines, and rising bass harmonic and IM distortion at program levels (which are, in general, at least 14 dB higher than tone level at "0" VU). The 30Hz highpass filter does not cause significant loss of bass "tightness", and certainly does not introduce "thinness".

1.c) **Allpass Phase Scrambler And Preemphasis/Deemphasis** (on Cards #3 and #4)

The FM medium has symmetrical positive and negative overload points (±100% modulation). Some music, and voice in particular, have highly asymmetrical waveforms. Therefore, maximum loudness consistent with the overload constraints of the FM medium is enhanced by processing waveforms to make their peaks more symmetrical.

In OPTIMOD-FM 8100A/1 this is achieved by a combination of the crossover network in the master/bass multiband compressor and a third-order non-minimum-phase filter. This crossover is 12dB/octave, and when its outputs are summed, it provides a single-order phase shift to complete the phase scrambler function.

The frequency response of the second stage of the phase scrambler is slightly peaked, and provides preemphasis into the multiband compressor to improve its accuracy. A deemphasis stage after the bands are summed restores flat response.

The phase scrambler is a low "Q" circuit which does not introduce ringing. Its audible effect is extremely subtle. It can be heard as a very slight change in the "sound" of some voices. Music, in general, is audibly unaffected. Despite the fact that square waves emerging from the scrambler no longer look like square waves, the purist should not fear that it is degrading audio quality. It is in fact significantly improving subjective distortion performance of the system.

2.a) **Dual-Band "Master/Bass" Compressor:** (audio on Cards #3 and #4; control on Card #5)

We feel that operating the third band of a conventional triband compressor independently of the rest of the bands yields very unnatural high frequency response when auditioned on high quality receivers. In addition, operating the low frequency band independently may result in unnatural frequency balances with certain music -- particularly "beautiful" or classical. For this reason, the multiband compressor in the 8100A/1 is quite dissimilar to a familiar triband unit, and offers unprecedented versatility in combination with very natural, unfatiguing sound.

The major part of the 8100A/1 compressor is the "Master" channel. This carries all program material above 200Hz. It is a feedback compressor, and its control voltage can be summed in a dB-linear manner (U.S. patent #4,249,042) with the control voltage developed by the "Bass" compressor to control the gain of the "Bass" VCA, which passes frequencies between 30 and 200Hz.

The summation is variable from none at all (in which case the "Master" and "Bass" bands operate independently, as in a conventional triband compressor) to unity gain (in which case the "Bass" channel always takes as much gain reduction as the

"Master" channel). In the latter "quasi-wideband" case, the feedback compressor control loop in the "Bass" channel is still active, and causes further gain reduction in the "Bass" VCA when program material with excessive bass energy is present. This avoids the pumping which would occur in a fully-wideband system if excess bass were to force gain reduction of the entire program.

2.b) Crossover and Bass Clipper: (on Card #3 and #4)

OPTIMOD-FM Model 8100A/1 employs a 12dB/octave crossover. The 12dB/octave configuration is simply two identical 6dB/octave filters in series, with the polarity of the "Bass" band inverted. It can be shown that the sum of the two outputs has a perfectly flat magnitude response, but exhibits an overall phase shift. This phase shift is purposely used as part of the "phase scrambler" to make peaks more symmetrical.

In OPTIMOD-FM Model 8100A/1, this crossover is realized as a "distributed crossover" (U.S. patent #4,249,042). This means that the first 6dB/octave section is before the VCA, and the second 6dB/octave section is after the VCA and the control voltage rectifier. The control voltage circuitry is therefore fed from a 6dB/octave crossover only.

The advantage of this configuration is that it permits insertion of a soft clipper immediately after the "Bass" VCA to eliminate overshoots which would otherwise intermodulate with the output from the "Master" VCA when the sum of "Bass" and "Master" is preemphasized and clipped. The second part of the "Bass" crossover is after the "Bass" clipper, thus lowpass-filtering the clipper output and rolling off harmonics and out-of-band IM introduced by the clipping process. In-band IM is negligible because of the relatively narrow bandwidth processed by the "Bass" channel.

The sum of the "Bass" and "Master" channels is applied to a deemphasis network to "undo" the preemphasis introduced in the phase scrambler circuitry (see **1.c**).

2.c) Voltage-Controlled Amplifier (VCA) Operation:

The voltage-controlled gain block used throughout OPTIMOD-FM Model 8100A/1 is a proprietary Class-A VCA which operates as a two-quadrant analog divider with gain inversely proportional to a current injected into the gain-control port. A specially-graded Orban IC contains two matched non-linear gain-control blocks with differential inputs and current outputs. The first of these is employed in the feedback loop of an opamp to perform the gain control function. The inputs of the first and second gain-control blocks are connected in parallel, and the output of the second block is a distortion-corrected current which is transformed into the desired gain-controlled voltage by means of an opamp current-to-voltage converter. For most gains, levels, and frequencies, THD is well under 0.1%. Overload-to-noise ratio (noise measured in a 20-20,000Hz band) is typically 90dB, and is constant with respect to gain and level.

2.d) Compressor Control Circuitry: (on Card #5)

Each compressor (left and right "Master" and left and right "Bass") feeds its own rectifier with threshold. The drive to the clippers following the compressors and preemphasis/hf limiters is determined by the setting of the CLIPPING control, which simultaneously adjusts all rectifier thresholds (and thus the average compressor output level). Left and right rectifier pairs (which have current-mode high-impedance outputs) are "OR"ed into individual timing circuitry for "Master" and "Bass" channels.

This timing circuitry is proprietary, and is located within sealed modules. The "Master" timing circuitry is most critical to achieving natural sound. It performs the following functions:

1) A peak limiting function with very fast recovery time for transient material;

2) A slower compression function whose recovery time is a function of gain reduction; and,

3) A recovery-delay function which provides extra smoothing of the gain control voltage to avoid low frequency distortion even with fast release times.

The recovery time of the compression function is adjustable in the "Master" band only by means of the RELEASE TIME control. In addition, a gating circuit radically slows the recovery time of the compression function if the input program level drops below a threshold adjustable by the GATE THRESHOLD control, thus preventing "noise swish-up" during pauses or low-level material.

The gain eventually recovers to maximum over a period of about two minutes. This prevents the unit from "going to sleep" permanently on, say, a quiet musical passage (below gating threshold) which follows a loud musical sforzando which has forced considerable gain reduction. (This effect, incidentally, is why we chose not to incorporate an expander into the AGC system -- such circuits tend to make musical mistakes like the one just described.)

The "Bass" timing circuitry is similar to the "Master" timing circuitry, and performs all of the same functions. Its time constants are optimized for most natural, dynamic sound.

Both timing circuits process the signal in logarithmic form, and have low-impedance outputs. The timing circuits drive exponential converters which provide control-current outputs for the "Master" and "Bass" VCA's. The BASS COUPLING control provides the ability to sum a controlled amount of the "Master" timing circuit output into the "Bass" exponential converter, where it sums with the output of the "Bass" timing circuit in a dB-linear manner.

Extensive gain reduction metering is provided. Since the outputs of the timing circuits are dB-linear, the gain reduction meters are provided with dB-linear scales.

The output of the "Master" timing module is applied to a peak detector which "holds" the fast-limiting component of the control voltage until the gain reduction meter ballistics have a chance to "catch up". The output of this peak detector drives the "Master" gain reduction meter, which shows the true peak value of the gain reduction.

The output of the "Master" timing module also drives a slewrate-limited amplifier which removes the fast limiting spikes from the voltage, and which drives the "compressor" meter to show the amount of slow compression occurring.

By subtracting the output of the slewrate-limited amplifier from the peak detector, the fast peaks only are derived. This difference signal feeds the "limiting" meter.

The output of the "Bass" timing circuitry contains a much smaller fast peak limiting component than does the output of the "Master" timing circuitry. No peak detection is necessary to assure accurate metering, and the output of the "Bass" timing circuitry thus drives its gain reduction meter directly.

3) Phase-Corrected Lowpass Filter/Preemphasis: (on Card #6)

After the outputs of the "Master" and "Bass" channels have been summed, they are passed to a filter which performs three functions:

1) It lowpass-filters the signal at 15kHz and 24dB/octave to prevent frequencies beyond the bandwidth of the system from unnecessarily activating the high frequency limiter or causing unnecessary IM distortion in the clipper;

2) It provides a standard FM preemphasis (75us or 50us, depending on Region -- see **Appendix G** if you wish to change the 8100A/1's preemphasis); and,

3) It provides phase correction to make the delay of the lowpass filter plus preemphasis approximately constant, thus minimizing the unavoidable increases in peak level resulting from the preemphasis and filtering functions.

The lowpass filter is designed to partially equalize the frequency response variations in the main 15kHz lowpass filter following in the FM Smart Clipper, thus providing flatter overall frequency response. The preemphasis is created by summing a second-order bandpass filter with the flat signal. The rising side of the filter slope provides the preemphasis; the falling side provides part of the lowpass filter function. The phase corrector is a fourth-order allpass filter, and is physically placed before the lowpass filter and preemphasis.

4) High Frequency Limiter: (on Card #6)

In order to perform the hf limiter function, a variable-gain stage is placed between the output of the bandpass filter creating the preemphasis (see 3 immediately above) and the amplifier which sums the bandpass filter output with the main signal. Thus high frequency limiting is effected by dynamically reducing the preemphasis as required.

The variable gain stage is realized by a junction FET operating as a voltage-controlled resistor, instead of by a VCA as in other processing functions within the 8100A/1. This simplification is possible because the high frequency limiters in the left and right channels are entirely independent, and need not track accurately together.

Each channel has its own rectifier and timing module. The timing in the hf limiter is considerably simpler than in the compressor sections because only fast dynamic filtering occurs; there is no "compression" function.

It should be noted that the 8100A/1 hf limiter is activated by high frequency energy only, as opposed to the old 8000A, whose hf limiter was sensitive to the peak level of the entire preemphasized signal. The 8100A/1's hf limiter therefore cannot be activated by, for example, low frequency overshoot components from the previous multiband compressor. This design improvement is possible because the 8100A/1's FM Smart Clipper permits considerably greater amounts of clipping than the clipping scheme in the 8000A without introducing audible distortion, thus rendering the hf limiter function far less critical and permitting substantial increases in perceived high frequency power output. The 8100A/1 hf limiter need not "know" about the actual peak level of the preemphasized signal -- only approximately how much hf energy is present.

A HIGH FREQUENCY LIMITING control available to the user adjusts the threshold of hf limiting over a range of approximately 3dB. The lowest threshold results in very little clipping on sinusoidal tone; the +3dB threshold results in moderate clipping of tone above approximately 2kHz. In most cases, users prefer operating this control in full "hard", which moves the threshold to the "+3dB" point and results in minimum hf limiting and maximum hf control by clipping, while still limiting hf energy which would otherwise cause disturbing distortion if it were clipped.

Operation of the hf limiter is metered by a simple comparator circuit which lights the appropriate front-panel HF LIMIT lamp if any hf limiting at all occurs. It is primarily useful to verify that the hf limiter circuit is operating properly.

5) **FM Smart Clipper:** (on Cards #8 and #9)

The 8100A/1 FM Smart Clipper (U.S. Patent #4,208,548) is a clipper circuit in which the threshold of clipping is varied dynamically as a function of the high frequency energy in the program, and in which a distortion-cancelling sidechain permits all clipping distortion below 2.2kHz to be reduced by at least 30dB. This results in cleanest sound on both high and low frequencies, and controls the classic distortion problems in preemphasized clippers, the most significant one being sibilance splatter.

It should be noted that it is normal for sinewaves to modulate less than 100% when applied to the 8100A/1 in its normal OPERATE mode. There are two principal reasons for this:

1) Some headroom is left between the threshold of the FM Smart Clipper and the threshold of the subsequent overshoot corrector in order to accommodate the distortion corrector signal. With sinewaves, no distortion corrector signal is produced. Thus, the headroom is not used, and full 100% modulation does not occur.

2) Sinewaves have a very low peak-to-average ratio and high loudness potential compared to program material of identical peak levels. The audio processing, in order to maintain natural sound quality, pushes sinewaves down in level as it would any other similar program material with low peak-to-average ratio. In general, any audio processor which produces 100% modulation on sinewaves tends to sound somewhat unnatural because this psychoacoustic factor has not been accounted for.

5.a) Clipper With Dynamic Threshold: (on Cards #8 and #9)

The clipper is a straightforward shunt clipper which is ordinarily biased with ±1.5 volts, thus providing a somewhat "soft" characteristic (but not nearly as soft as a pair of back-to-back unbiased diodes). The characteristic was chosen to obtain the best compromise between harmonic and IM distortion induced by clipping, when the IM-cancelling circuitry is considered.

The output of the bandpass filter in the high frequency limiter (see 4 above) feeds a rectifier with threshold. When high frequency energy exceeds this threshold, the clipper bias voltage is reduced to reduce the clipping threshold by approximately 1.0dB. The purpose of this threshold reduction is to provide headroom between the clipper threshold and the subsequent overshoot corrector threshold. This headroom accommodates the distortion corrector signal (see 5.c below) which is needed to correct the IM distortion produced when large amounts of hf energy are clipped. If this headroom were not provided, the overshoot corrector would clip off the distortion corrector signal, thus negating its effect. On the other hand, when the input signal to the clipper contains predominantly low frequency energy, the distortion corrector loop is essentially ineffective. In this case, the absolute amount of clipping is minimized by raising the clipping threshold to approximately the threshold of the overshoot corrector.

5.b) 15kHz Phase-Corrected Lowpass Filter: (on Cards #8 and #9)

The main clipped signal is applied to a 15kHz phase-corrected fifth-order Cauer lowpass filter to remove energy above 15kHz, particularly that induced by clipping. (Bear in mind that the unclipped signal was already filtered in the preemphasis lowpass filter.) The signal is also passed through a shelving deemphasis to help match the frequency response of this main signal path to the frequency response of the distortion-cancelling sidechain.

The delay of the 15kHz filter is equalized by means of a fourth-order allpass network to achieve minimum filter overshoot, and also to add sufficient delay to the main path so that the delay of the distortion-correcting sidechain can be matched.

5.c) Distortion-Cancelling Sidechain: (on Cards #8 and #9)

The clipper's output is subtracted from its input to derive the distortion introduced by the clipper. This distortion signal is applied to a 2.2kHz lowpass filter which matches both the frequency response and time delay of the main-path filter extremely closely throughout the 2.2kHz passband.

The output of this distortion-cancelling sidechain filter is summed into the output of the main-path filter. Distortion below 2.2kHz is cancelled by adding the filtered distortion signal to the main path. (Because the clipper's output was subtracted from its input, the actual distortion component -- which appears at the clipper's output -- is out-of-phase and thus cancels properly). The output of the 2.2kHz filter can be considered a smoothing signal. This signal increases the peak level of the summed signal somewhat. If the level increases beyond the 100% point, it is clipped by the subsequent overshoot corrector. In most cases, the level buildup is not very large because the distortion signal contains mostly high frequencies, and these are filtered out by the 2.2kHz lowpass filter, thus greatly reducing the level of the distortion signal before it is summed with the main signal.

The sum of the main-path and distortion signals is passed through a shelving preemphasis which is precisely complementary to the shelving deemphasis introduced in the main path, thus restoring overall flat response.

6) Frequency-Contoured Sidechain (FCS) Overshoot Compensator:
(on Cards #8 and #9)

The FCS overshoot compensator (U.S. Patent #4,460,871) is best thought of as a "bandlimited safety clipper". That is, it performs the function of clipping off overshoots from the earlier FM Smart Clipper, but does not produce out-of-band frequency components as a simple clipper would. If such components were produced, they would produce "aliasing distortion" when applied to the stereo generator and then decoded in a receiver. Simultaneously, the FCS overshoot compensator does not significantly increase low frequency IM products when compared to a simple clipper performing the same function -- a problem particularly characteristic of competing overshoot compensation circuits.

Briefly, the FCS overshoot compensator functions by deriving the overshoots from its input with a "center clipper" circuit, lowpass-filtering the overshoots with a fifth-order passive LC filter to remove out-of-band frequency components, and then mixing the filtered overshoots out-of-phase with a delayed version of the input signal. This delay, created by an encapsulated active allpass network, is identical to the delay in the overshoot filter, thus assuring that the input signal and filtered overshoots arrive in the same place at the same time.

If no filter were used, this process would be identical to clipping the input signal, and would create a "differential clipper". However, the overshoot filtering process reduces the peak level of the high frequency components of the overshoot by removing harmonics. To compensate for this loss of peak level (which would cause less than full cancellation of overshoot), the frequency response of the overshoot filter rises at 15kHz -- thus increasing the level of the high-frequency fundamental to compensate for the loss of harmonics. This is why the system is called "Frequency-Contoured Sidechain".

The final sum of input and out-of-phase filtered overshoot is passed through a third-order lowpass filter to provide further attenuation of unwanted high frequency energy. Phase correction is applied to the combination of this filter and the overshoot filter. (The phase response of the overshoot filter is identical to its matched main-path delay network -- thus the phase correction also corrects the response of the main-path delay network.) This phase correction makes the overall time delay through the entire FCS overshoot compensator approximately constant, and assures that the various filters within the compensator do not upset carefully controlled peak levels in unpredictable ways.

If any very unusual waveforms cause residual overshoots, these are dealt with in a safety clipper at the output of the FCS overshoot compensator system. However, the basic FCS overshoot compensator is so effective that this safety clipper is hardly ever active.

7.a) Stereo Generator -- General Principles: (on Card #7)

The 8100A/1 stereo generator uses the "matrix" approach to the generation of the composite baseband signal. That is, L+R (sum) and L-R (difference) are derived from the input signals, which have been bandlimited to 15kHz by the earlier circuitry. The L+R signal appears directly at the baseband output and occupies the frequency range from 30 to 15,000Hz; the L-R is first multiplied by a 38kHz sinewave to form a double-sideband suppressed-carrier subchannel occupying 23 to 53kHz, and is then summed into the baseband output. The final signal summed into the output is the 19kHz pilot subcarrier, which is phase-locked to the 38kHz subcarrier.

The two principal advantages to this approach (as opposed to the more common "switching" approach) are:

1) The L+R (which is usually the dominant component) is not subject to any switching or modulation process which may introduce distortion.

2) The baseband is derived by a multiplication process which ideally introduces no out-of-band frequency components. In the 8100A/1, spurious components are typically below 0.02% modulation. The baseband thus requires no lowpass filtering. Such a filter is often the most costly part of a conventional switching stereo generator, and necessary compromises in its design tend to degrade separation.

Stereo/mono mode switching is provided by FET switches driven by CMOS logic. In addition, there are two special "test" modes available. Each accepts a right-channel signal. The first creates an L+R signal to test main-channel-to-subchannel crosstalk; the second creates an L-R signal to test subchannel-to-main-channel crosstalk.

The pilot subcarrier and 38kHz subcarrier are generated by precise circuitry which incorporates several feedback loops to assure stability of the stereo generator parameters with time and temperature.

The block diagram (p. J-21) depicts the major subsystems in the stereo generator and should help clarify the discussion below.

7.b) 19kHz Oscillator: (on Card #7)

The 19kHz oscillator employs a 19kHz crystal operated in series-mode to provide positive feedback around an opamp. The oscillator produces a sinewave with well under 0.1% distortion using no other tuned circuits than the crystal itself. To avoid oscillator saturation and distortion, the loop gain of the oscillator is adjusted by a FET whose resistance is controlled by an AGC feedback loop which senses the output level of the oscillator and corrects it as necessary to sustain linear oscillation.

The oscillator can be squelched in mono mode by shorting out the positive feedback through the crystal with an FET switch.

7.c) 19kHz Doubler: (on Card #7)

The output of the 19kHz oscillator is turned into a 38kHz square wave by means of a biased dual comparator. The comparator is biased so that it changes state every time the 19kHz sine wave goes through 45, 135, 225, and 315 degrees. The amplitude of the 38kHz square wave is adjusted by varying the voltage on the "strobe" inputs of the comparator; the 38kHz AGC servo adjusts this level, and thus the level of the filtered 38kHz sinewave.

7.d) 38kHz Filter and Phase Shifter: (on Card #7)

The 38kHz square wave output from the comparator is filtered into a sinewave with less than 0.02% distortion by means of an active filter cascaded with a passive filter. The active filter is a high-"Q" resonator whose center frequency is controlled by a FET activated by the phase-locked-loop circuitry (see **7.f**). By tuning this resonator slightly around the resonance point, its phase shift is significantly affected. This provides the phase control to lock the 19kHz and 38kHz together with correct phase as required for proper operation of the stereo decoder.

A passive third-order Cauer filter attenuates residual distortion in the output of the 38kHz active resonator to achieve the final distortion specification. It is very important that the 38kHz applied to the multiplier be as clean as possible, as every remaining harmonic component will be multiplied by the L-R audio to create out-of-band suppressed-carrier subchannels around each of the harmonics.

7.e) 38kHz AGC Loop: (on Card #7)

Maintaining high separation depends upon precisely controlling the ratio between L+R and L-R gain. Since the L-R is multiplied by the 38kHz sine, the gain of the L-R is directly proportional to the amplitude of the 38kHz sine. This amplitude must be highly stable, or separation will drift with time and temperature.

The 38kHz sine is derived by hi-"Q" filters which can drift. In addition, the PLL circuitry will cause gain variations in the 38kHz active bandpass filter. It is clear, therefore, that a closed-loop servo system must be used to stabilize the 38kHz sine level.

To do this, the filtered 38kHz sine is applied to a high-gain comparator. The other comparator input is connected to a reference voltage derived from the SEPARATION control. The servo forces the peak level of the 38kHz sine to be identical to this reference voltage. The output of the comparator is filtered, buffered, and then connected to the "strobe" inputs of the 38kHz doubler dual comparator, which has the effect of controlling the output level of the comparator, thus closing the AGC loop.

7.f) 38kHz Phase-Locked Loop (PLL): (on Card #7)

The relative phase between the 19kHz pilot and the 38kHz subcarrier must be precisely controlled to achieve optimum separation. This is achieved by detecting the phase between the 19kHz output of the pilot oscillator and the filtered 38kHz sine, and by activating a tuning FET in the 38kHz active resonator which tunes this resonator slightly to either side of resonance, thus adjusting the filter's phase shift to achieve the exact phase desired at the output of the passive filter.

The phase detector is a biased dual comparator which is essentially insensitive to the level of its 38kHz input, but which is quite sensitive to the (highly controlled) level of its 19kHz input. The output of the phase detector eventually feeds a true integrator. This makes the PLL a "type-1" servo system, and results in zero steady-state phase error regardless of loop gain.

7.g) **Stereo Modulator:** (on Card #7)

The stereo modulator derives the L-R in a differential amplifier whose common-mode rejection can be precisely trimmed to null main-channel-to-subchannel crosstalk. The L-R output of the differential amplifier is applied to the input of a VCA which is almost identical to the VCA's used in the audio processing. (See **2.c** for further information.)

The gain control port of the VCA is fed by the sum of the 38kHz sine and a DC current which biases the gain control port to prevent the control current from ever becoming negative. (The VCA can only accommodate single-polarity control currents.) This DC bias means that the output of the VCA contains two components: the desired product of the input L-R and the 38kHz, and an undesired product of the DC and the input L-R, which is simply L-R. This latter component (which is subchannel-to-main-channel crosstalk) is cancelled by feeding some of the L-R around the VCA out-of-phase. Final sub-to-main null is achieved by slightly adjusting the gain of the VCA.

The VCA produces a small second harmonic (76kHz) feedthrough component (typically 70dB below 100% modulation without correction). This is cancelled by applying a small amount of 38kHz to the input of the VCA, where the 38kHz is multiplied by itself to produce a second-harmonic component which is out-of-phase with the 76kHz VCA feedthrough. When the 76 KHZ NULL trimmer is properly adjusted, stable cancellation of more than 80dB is possible.

The pilot, and a DC component to cancel DC offsets at the output of the generator (thus rendering an output coupling capacitor unnecessary) are also summed into the output of the VCA. This mix, which is desired only in stereo mode, is switched into the summing junction of the output opamp through a JFET switch.

L+R is derived by passively mixing the L and R inputs, and then by introducing the sum into the summing junction of the output opamp of the stereo generator. FET switches are provided to turn each channel individually ON or OFF, and also to change gain in mono mode. In mono, an individual channel (either L or R) feeds the output opamp by itself. Its gain into the output must therefore be increased 6.84dB so it can produce 100% peak modulation.

7.h) (not used)

7.i) (not used)

7.j) (not used)

7.k) Mode Switching Logic: (on Card #7)

The 8100A/1 stereo generator has three switch-selected modes of operation. Two are special test modes, and are used only when making stereo performance verification measurements. The first of these facilitates measurement of main-to-sub crosstalk by switching the signal applied to the right stereo generator input into both left and right stereo generator inputs. Since this test signal is also applied to the L-R differential amplifier, this mode can also be used to null the stereo generator's internal MAIN-TO-SUB CROSSTALK NULL.

The second mode facilitates the measurement of sub-to-main crosstalk by injecting the signal on the right channel input line into the L-R differential amplifier. The stereo generator is arranged such that this test mode can also be used when adjusting the stereo generator's internal SUB-TO-MAIN CROSSTALK NULL trimmer.

The normal mode is operate. In this mode, three sub-modes are available by remote control or by switching the front-panel STEREO/MONO switch.

The first is stereo, and simply generates a normal stereo baseband output from the signals on the left and right stereo generator input lines.

The second and third sub-modes are mono left and mono right. These take the inputs on the left and right stereo generator input lines respectively, and apply them directly to the stereo generator output amplifier with sufficient gain to scale peaks to 100% modulation.

A CMOS three-state latch using three NAND gates "remembers" the logic state. The three hard-switched modes are also integrated into the logic system, principally by means of two CMOS exclusive-OR gates. Note also that in order to permit stereo crosstalk measurements, the sub-mode logic is forced into stereo when either of the crosstalk test modes is entered.

The schematic diagram for the stereo generator (Card #7) contains a truth table that shows the state of each JFET switch in each of five operating conditions: main-to-sub test, sub-to-main test, stereo operate, mono left operate, and mono right operate. Further understanding of the operation of the logic and switching system in the 8100A/1 stereo generator may be obtained by studying this table.

8) Power Supplies:

Primary power for the 8100A/1 circuitry comes from a highly regulated ±15 volt power supply. The main supply is +15 volts. This is created by means of a 723C IC regulator with current-boosted output, current limiting, and overvoltage protection using a zener diode and fast-blo fuse.

The -15 volt supply is essentially a current-boosted opamp in a unity-gain inverting configuration which "amplifies" and inverts the +15 V supply, thus "tracking" it. The -15 volt supply is also current-limited and overvoltage protected. Both +15 and -15 supplies are located on a non-plug-in card mounted on the inside of the rear chassis apron. This apron is also used as a heat sink for the regulator power transistors.

The 711C comparators used in the stereo generator require +12/-6 volt supplies. These are created by locally dropping the main +15 volt supply through three forward-biased silicon diode junctions (to create the +12), and by means of local resistive voltage dividers from -15 to ground (to create the -6). In addition, a number of local ±14 volt supplies exist. These are created by means of single diode drops from the ±15 volt main power supply, with local capacitive decoupling. Their purpose is to decouple noise from the main power supply.

Bias supplies are also required for the diode clippers in the audio processing. There are two such supplies; the first creates approximately ±1.2 volts (for Cards #3 and #4), while the second creates ±4.2 volts (for Cards #8 and #9). Both supplies use a pair of opamps. The first is a unity-gain voltage-follower whose input is a temperature-compensated voltage created by a resistor/diode network; the second is a unity-gain inverter which creates the complementary negative voltage.

APPENDIX B:

Circuit Description

The following section provides an extremely detailed description of the circuitry used in OPTIMOD-FM on the component level.

It may be wise to read **Appendix A** first, and to consult the block diagram on p. J-21. Referring to the appropriate Schematics and Parts Locator Drawings in **Appendix J** will help you to follow the text and will aid component-level troubleshooting.

In those cases where the circuitry is duplicated in the left and right channels, only the left channel circuit and component designators will be discussed.

1.a) Input Amplifier: (on Cards #3 and #4)

The input is applied to the RF filter chamber, and there encounters an RF filter and 10K bridging pad R303, R304, R305. Strapping R305 into the pad introduces 20dB loss, which is the normal condition of the pad.

The output of the pad is connected to a low-noise true instrumentation amplifier consisting of IC301A, IC301B, IC302A, and associated resistors. R306, R307 provide bias current for IC301, which is a low-noise bipolar-input dual IC opamp. R308, R311 are feedback resistors for the two sections of IC301. The differential gain is controlled by the series resistance of R310 and GAIN control R309. The common-mode gain of the IC301 pair is 1.

The differential output of IC301A and IC301B is converted to a single-ended output and the common mode component of the output is nulled by means of differential amplifier IC302A and associated resistors. R316 adjusts the balance of the resistor network to assure maximum common-mode 60Hz rejection.

NOTE

Nearby lightning strikes may induce sufficient energy into the 8100A/1's audio input wiring to pass through the RFI protective networks and destroy IC301 or IC401. If the 8100A/1 is installed in a lightning-prone location, it is advisable to keep spare NE5532's in stock. Installation of varistors between each side of the audio input lead and earth may help prevent such problems. IC301 is socketed, and is thus easily replaced.

1.b) 30Hz Highpass Filter: (on Cards #3 and #4)

The non-bypassable 30Hz highpass filter IC302B, C303, C304, C305, R317, R318, R319 is a third-order Chebychev filter with 0.5dB passband ripple (nominal) and a ripple bandwidth (i.e., -0.5dB frequency) of 30Hz. It is realized as a unity gain Voltage-Controlled Voltage-Source (VCVS) active filter. This filter is non-inverting, has a gain of exactly 0dB in the passband, and uses positive feedback to "sharpen up" the response around the cutoff frequency. Most modern books on active filters extensively discuss this type of filter. (See for example -- Wong and Ott: Function Circuits. New York, McGraw-Hill, 1976, pp. 230-231.)

1.c) Allpass Phase Scrambler and Preemphasis/Deemphasis: (on Cards #3 and #4)

This filter contains a single-order allpass filter IC303A, R320, R321, R322, C306 followed by a second-order non-minimum-phase peaking equalizer IC303B, R323, R324, R325, R326, R327, C307, C308. The phase response of the first section varies from 0 to 180 degrees as a function of frequency, while the phase response of the second section varies from 0 to 360 degrees as a function of frequency. The amplitude response of the first section is flat; the amplitude response of the second section is broadly peaked at approximately 200Hz.

To restore flat response, a complementary dipping-equalizer section is inserted after the two bands of the dual-band compressor have been recombined. This circuit consists of IC307A, R359, R360, R361, R362, R363, C319, C320. Its gain far from 200Hz is -1.76dB, and it exhibits a second-order dip centered at 200Hz.

2.a) Dual-Band "Master/Bass" Compressor (General): (on Cards #3, #4, and #5)

The dual-band compressor consists of an audio path and control circuitry. We will first discuss the audio path generally. Details of the VCA operation and control circuitry operation are found immediately below in sections **2.c, 2.d,** and **2.e.**

2.b) Crossover And Bass Clipper (on Cards #3 and #4)

The crossover consists of 12dB/octave sections. The first 6dB/octave filter is located before the VCA, the second after. Since the input to the control rectifiers is taken from the VCA outputs, the control-voltage crossover is 6dB/octave.

The first 200Hz highpass section for the "Master" compressor is filter C309, R328. The second 200Hz highpass section is C318, R357. The first 200Hz lowpass section for the "Bass" channel is R342, R343, C314. The second lowpass section for the "Bass" channel is R367, R366, R365, C321.

A clipper, consisting of biased diodes CR303, CR304, and resistors R367, R366, is located before the second lowpass section. Thus the second lowpass section rolls off harmonics created by clipping.

In order to force the "Master" and "Bass" channels to add correctly, the polarity of the "Bass" VCA is inverted by using the appropriate inputs of IC309B (compare with IC305B).

2.c) Voltage-Controlled Amplifier VCA Operation:

NOTE

> This section contains a general description of the voltage-controlled amplifier circuitry used through the 8100A/1, including the multiplier in the stereo generator. The "Master" VCA will be specifically described.

The basic operation of the VCA depends on a precisely-matched pair of gain-control blocks with differential voltage inputs and current-source outputs. The gain of each block is controlled by means of a control current.

If used alone, one such gain-control block would introduce considerable distortion. Therefore, the first of the two matched blocks IC305A is used as the feedback element in a high-quality operational amplifier, IC304. The second of the matched blocks IC305B is then driven by the predistorted output of IC304. To provide more detail: The output of IC304 is first attenuated by R334, R335, C311, and then applied to the input of the feedback element IC305A. The output of IC304 is predistorted as necessary to force the current output of IC305A to precisely and linearly cancel the audio input into the "virtual ground" summing junction of IC304. This same predistorted voltage is also connected to the input of IC305B. Thus the output of IC305B is an undistorted current, which is converted to a voltage in current-to-voltage converter IC306A, R341, R376, C312. The output of IC306A is the output of the VCA.

Because IC305A is in the feedback loop of IC304, the gain of the VCA is inversely proportional to the gain of IC305A. Thus if the control current is applied to the control port of IC305A (through R333), then the VCA behaves like a two-quadrant analog divider. However, if the control current is applied to the control port of IC305B, then the gain of the VCA is directly proportional to the gain of IC305B, and the VCA behaves like a two-quadrant multiplier. The VCA is used in the "divider" mode in the "Master" and "Bass" VCA's, and in the "multiplier" mode in the stereo generator.

In the case of the "Master" VCA, a fixed current is applied to the control port of IC305B through R339, R340, CR301 to fix the gain of IC305B. CR301 provides temperature compensation.

Second-harmonic distortion is introduced by differential offsets in either IC305A or IC305B. This distortion is cancelled by applying a nulling voltage directly to the input of IC305B by means of resistor network R336, R337, R338.

If the VCA is not perfectly balanced, "thumps" due to control current feedthrough can appear at the output. These are equivalent to multiplying the control current by DC. If a correct DC offset is applied to the VCA input, then this equivalent DC multiplication can be nulled to zero and the "thumps" eliminated. Such an adjustable DC offset is provided by R331, R332.

R329, R330, C310 are frequency-compensation components to prevent the VCA from oscillating supersonically.

2.d) **Compressor Control Circuitry:** (on Card #5)

The output of the "Master" VCA is applied to a voltage in/current out fullwave rectifier-with-threshold, IC503A, IC504, R505, R506, R507, C502, CR502. This is essentially an opamp with discrete class-B output stage. A bias voltage of -12V on the "+" input of IC503A holds the voltage at this opamp's "-" input at -12V by feedback and provides appropriate bias conditions for the rectifier to prevent saturation. R507 determines the rectifier's transconductance. C502 provides DC blocking between the nominal ground potential of the input side of R507 and the -12 volts at IC503A's "-" input.

A negative-going voltage at the input side of R507 pulls current away from the "AC virtual ground" at IC503A's "-" input. An equal current must therefore flow into "-" input by turning on the NPN transistor whose base is connected to the output of IC503A. Because of the class-B biasing, this assures that the PNP whose base is connnected in parallel to the NPN is off.

A collector current essentially equal to the NPN's emitter current flows into the NPN from the output terminal of the rectifier. Part of this current comes from the rectifier load; part comes from the fixed collector current of the top PNP transistor. This PNP creates the threshold of limiting by saturating and diverting all class-B output stage collector current away from the load until the output stage current exceeds the nominal PNP collector current. When the output stage current exceeds the PNP collector current, the PNP comes out of saturation, and the difference between the PNP collector current and the rectifier output current is delivered to the rectifier's load. The PNP transistor's collector current is fixed by its emitter resistor R505, and by its base voltage (determined by the setting of the CLIPPING control, R542). The CLIPPING control thus varies the collector current of the PNP, and therefore the threshold of limiting. In PROOF mode, CR502 and R506 parallel R505, increasing the collector current, and raising the limiting threshold by approximately 14dB.

If the voltage at the input side of R507 goes positive, then the bottom PNP transistor turns on, and its neighboring NPN turns off. The collector current of the PNP is inverted by the dual NPN current mirror, and the current mirror output is summed into the output port of the rectifier, where it is also subject to the action of the PNP threshold transistor as described immediately above.

The output of the left "Master" rectifier is "OR"ed with the output of the right "Master" rectifier by means of two diodes: CR501 & CR504. Because the output of the each rectifier is in the form of a current, voltage drops across the "ORing" diodes do not affect the accuracy of the rectifier.

The output of the "OR" circuit is applied to a proprietary circuit which computes the VCA control voltage. Release time control for the slow "compressor" function is provided by R508, R509. A JFET switch Q501 is provided to radically slow the compression release, thus essentially freezing the gain when ordered to do so by the gating circuitry (described immediately below). 22-megohm resistors across the gating FET'S create a slow "leak" in the gating function which permits to gain to fully recover over a period of approximately two minutes.

The output of the control voltage module varies between 0 volts (maximum gain) and approximately -10 volts (minimum gain=maximum gain reduction). Thus release is inhibited by applying a voltage of greater than +10 volts to the anode of CR503 which forces Q501 off. Release is enabled by applying a voltage of less than -10 volts to the anode of CR503. This reverse-biases CR503, and Q501 is forced on by R510's forcing Q501's gate to be at the same potential as its source.

The output of the release time module is a low impedance voltage source. It is applied to exponential converter circuit IC501, IC502, R501, R502, C501 through pad R503, R504. The collector current of either matched transistor in IC502 is an almost perfect exponential function of its base-emitter voltage. The scaling factor of the converter is stabilized by forcing a constant current through the left-hand transistor by means of IC501. This current is determined by the current injected into IC501's "-" input through R502. The base of the left-hand transistor is grounded; the emitters of the matched transistors are connected. Thus, assuming a perfect match between transistors, the collector currents of the two transistors will be equal if the base of the right transistor is grounded. Varying the base voltage on the right-hand transistor varies its collector current exponentially about the nominal current in the left-hand transistor. This nominal current determines the quiescent (no-gain-reduction) gain in the VCA's. The current output at the collector of the right-hand transistor is connected to a matched pair of resistors, one of which feeds the gain control port of the left VCA, and the other of which feeds the gain control port of the right VCA. This is a "current divider" and is analogous to the familiar resistive voltage divider.

The operation of the "Bass" control loop is essentially identical to the operation of the "Master" control loop. The only essential difference is that provision is made to mix "Master" control voltage into the input of the "Bass" exponential converter through BASS COUPLING control R521, and R518. When R521 is fully clockwise, the "Bass" exponential converter is being fed as much "Master" control signal as the "Master" exponential converter. In the absence of output from the "Bass" release time module, "Bass" and "Master" VCA's will thus track exactly.

Because the "Bass" rectifier is always connected to the output of the "Bass" VCA, exceptionally strong bass will exceed the threshold of bass limiting and cause an output from the "Bass" release time module, thus momentarily decreasing the gain of the "Bass" VCA below that of the "Master" VCA. This is the low-frequency equivalent of familiar high frequency limiting.

(**NOTE:** The "multiband feedback compressor with crosscoupling into dB-linear VCA's" concept is protected by U.S. patent # 4,249,042.)

2.e) Gain Reduction Metering:

Gain reduction metering in the "Master" band is provided by three meters.

The first, TOTAL G/R, is driven by a peak detector IC512, R530, C508, CR511. C508 captures negative-going peaks and discharges slowly through R530. To avoid being loaded by the meter, C508 is buffered by voltage-follower IC512B. The discharge time of C508 is sufficiently slow to permit the mechanical movement of the TOTAL G/R meter to rise to the actual peak level of the gain control voltage, thus accurately displaying it.

COMPRESSION is indicated by passing the output of the release time module through a grossly overcompensated 301A opamp IC511 connected as a voltage follower. The 2.2uF compensation capacitor C507 so limits the slew rate of IC511 that only the slow component of gain reduction is permitted to drive the COMPRESSION meter.

The LIMITING meter is connected differentially between the outputs of IC511 and IC512B. It thus indicates the fast component of gain reduction as the difference between the slow component and the peak-held TOTAL component.

"Bass" band gain reduction metering reads the sum of the "Master" and "Bass" control voltages through R519, R520, in the same proportions that are applied to the input of the "Bass" exponential converter. In the interests of simplicity, the "Bass" TOTAL G/R metering signal is not electronically conditioned.

2.f) Gating Circuitry: (on Card #5)

The gating detector consists of a peak detector followed by a comparator. IC opamps are employed for both functions.

The left and right input signals are summed in R538, R539, and lowpass-filtered at 3kHz by means of C510. The lowpass-filtered sum is amplified by means of non-inverting amplifier IC513B, whose gain is variable from 0 to approximately 40dB by means of R537, the GATE THRESHOLD control. Low frequency response of IC513B is rolled off with C511 to prevent low-frequency noise from inhibiting the gate.

The positive peak output of IC513B is detected by CR514 and C503. R536 determines the recovery time of this peak detector.

The output of the peak detector is applied to comparator IC513A. R533, R534 create a reference voltage of +1.9 volts. If the output of the peak detector exceeds this value, then the output of IC513A is driven towards the negative power supply, and the gate is inhibited. Otherwise, the output of IC513A rests close to the positive power supply, and the gate is enabled. In this condition, the GATE LED is lit by current supplied through R531, CR512.

Hysteresis to assure clean switching is provided by positive feedback through R532.

In PROOF mode, CR513 applies +15 volts to the "-" input of IC513A to inhibit the gate, thus permitting all VCA's to recover to full gain.

3) Phase-Corrected Lowpass Filter/Preemphasis: (on Card #6)

Phase correction for the preemphasis and fourth-order lowpass filter associated with it is provided by a fourth-order allpass filter IC601, R601, R602, R603, R604, R605, R606, R607, R608, R609, C601, C602, C603, C604. The overall magnitude response of the filter between the card input and the filter output at IC601B is flat, gain is 0dB, and the phase response varies from 0 to 720 degrees. The operation of the filter is difficult to explain in words, and is best left to a mathematical analysis.

The fourth-order lowpass filter is in fact quasi-fourth-order. The first section of the filter is generated by a conventional second-order multiple-negative feedback active lowpass filter IC602A, R610, R611, R612, C605, C606. (See, for example, Wong and Ott: Function Circuits, op. cit.). However, the second section has been combined with the preemphasis, and transformed from a purely lowpass form to a peaking bandpass equalizer.

To understand this, imagine first a preemphasis cascaded with a 12dB/octave lowpass filter. As frequency is increased, the response will first rise at 6dB/octave, following the preemphasis. However, when the cutoff frequency of the lowpass filter is encountered, the response will reverse itself and fall at 6dB/octave indefinitely.

This is similar to the response of a peaking equalizer. However, when the response of the peaking equalizer falls, it does not fall indefinitely, but only until it reaches unity gain again. Nevertheless, we can choose a peaking equalizer whose rising side matches the rising side of our original preemphasis-plus-lowpass-filter to very close tolerances.

The falling side, after deemphasis, represents the stopband of the filter. Thus, when considered as a totality, the response of the entire fourth-order filter will, instead of falling indefinitely at 24dB/octave, fall for approximately 20dB after cutoff at 24dB/octave, and at 12dB/octave thereafter.

4.a) Differential Preemphasis and HF Limiter: (on Card #6)

The advantage of transforming the lowpass filter as described in the previous section is that it permits us to create the preemphasis differentially, by summing the output of bandpass filter IC602B, R614, R615, R617, C607, C608 with the filter's input. The summation occurs in IC605A. The output of IC602B is passed through a variable-gain stage, realized with FET IC603A, and low-noise non-inverting amplifier IC604. By varying the gain with which the output of IC602B is summed with its input, a high frequency limiter is realized.

Ordinarily, IC603A is pinched off, thus producing maximum gain and full preemphasis. However, as the gate voltage on IC603A is reduced toward ground, the resistance of IC603A decreases, thus decreasing the gain of voltage divider R619, R620, IC603A and reducing the preemphasis.

The polarity reversal in IC602B requires that a compensating polarity reversal occur in the summing process. IC605A is thus non-inverting for the bandpass signal, and inverting for the main signal. In addition, R616 feeds some of the output of IC602B around the variable-gain stage out-of-phase. This permits complete cancellation of the preemphasis despite the inability of the FET variable-gain stage to achieve total cutoff.

4.b) HF Limiter Control Circuitry: (on Card #6)

The high frequency limiter control circuitry is very similar to the compressor control circuitry described in **2.d** above. The output of the bandpass filter only is applied to the rectifier-with-threshold, which is identical to the ones used in the compressor control circuitry. Similarly, the output of the rectifier is connected to a proprietary release-time module, and the threshold-of-limiting adjustment and PROOF mode G/R defeat are also substantially identical to previously described circuits.

The output of the module, unlike the outputs of the release time modules in the compressor control circuitry, is high impedance. It drives the high impedance gate of FET IC603A through R625.

Gain reduction is indicated by a simple ON/OFF LED indicator, driven by IC606A. FET BIAS adjustment R626 determines the quiescent gate voltage of IC603A, assuring pinchoff under conditions of no limiting. This voltage is applied through release time resistor R627 to the "-" input of IC606A. This input will be pulled in the negative direction when gain reduction occurs.

The output of R626 is also applied to the "+" input of IC606A through R628. R629 pulls this "+" input slightly more negative than the output of R626 to hold the output of IC606A negative during conditions of no gain reduction. However, as soon as the "-" input of IC606A is pulled slightly less positive by the occurence of gain reduction, the output of IC606A goes positive and lights the HF LIMIT lamp through R630 and Q601, used as a zener diode.

5.a) Clipper With Dynamic Threshold: (on Cards #8 and #9)

The threshold of the first clipper CR801, CR802 is varied dynamically to make best use of the distortion-cancelling circuitry. When the clipper input signal contains substantial high frequency energy, then the threshold of clipping is lowered approximately 1.0dB to permit the difference-frequency IM distortion-cancelling signal to sum with the output of the main 15kHz lowpass filter without excessive clipping in the subsequent overshoot corrector. However, when the clipper input signal contains predominantly low frequencies, then the clipper threshold is raised to minimize the amount of low frequency clipping which occurs.

This is achieved by using the rectified output of IC602B (the high frequency bandpass filter employed differentially in the preemphasis filter) to control the clipper threshold. The output of IC602B feeds a rectifier-with-threshold IC806B, IC807 (and associated circuitry) whose operation is identical to the rectifier-with-threshold already described in **2.d.** This rectifier feeds a RC filter R844, R847, R848, CR807, C826. R844 determines the attack time of the circuit in conjunction with C826. The recovery time is determined by the series combination of R847 and R848.

If the high frequency energy present at the input to the rectifier is insufficient to overcome its threshold, then the "+" input of IC808A is held at ground by R847, R848. If output current flows into the rectifier, then C826 is pulled negative through CR807. If the voltage across C826 attempts to go more negative than approximately -13 volts, the rectifier will saturate and limit the voltage swing to this value. The voltage divider R847, R848 attenuates this 13 volt variation such that it causes a voltage variation of -0.2v at the output of IC808A, thus changing the threshold of clipping by approximately 1.0dB.

A 1.5 volt quiescent bias for the clipper diodes is provided by passing the output of a voltage divider through CR808 to R846. IC808A then acts as a unity-gain inverting amplifier for the voltage at CR808's anode. CR808 temperature-compensates the threshold of clipping by reducing the clipper bias voltage as the voltage drop across the diodes increases (with temperature variation). The final diode bias voltage at the output of IC808A is thus the sum of the quiescent voltage contributed by the circuitry connected to the "-" input of IC808A, and the voltage variation contributed by the circuitry connected to the "+" input of IC808A.

IC808B is connected as a unity-gain inverting amplifier, and provides a complementary negative bias for CR802.

It is important to understand that this scheme results in sinewaves not hitting 100% modulation in <u>OPERATE</u> mode. This occurs for two reasons. First, the dynamic response of the previous multiband compressor section is such that steady-state 1kHz sinewaves reach about 60% modulation if the CLIPPING control is adjusted to 12:00. This is a direct consequence of the natural loudness balances produced by this processing. Because sinewaves have a very low peak-to-average ratio compared with speech or music, their peak level must be reduced to prevent them (or similar program material) from being unnaturally loud and giving the processing an artificial, strained quality.

Second, a certain amount of headroom is left between the threshold of the first clipper and the threshold of the overshoot compensator to accommodate the distortion corrector signal and overshoots in the 15kHz lowpass filter without performing excessive, non-distortion-cancelling clipping in the overshoot corrector.

Because of the previously mentioned characteristics of the multiband compressor, sinewaves below about 2.5kHz are not ordinarily clipped; thus, no distortion-corrector signal is produced. However, varying the clip threshold does make better use of available headroom than would be the case if the clipper were always left at the "-1.0dB" threshold to which it switches in the presence of substantial high frequency energy.

It is important to note that despite (in reality, because of) this behavior with sinewaves, extremely high loudness is obtainable with speech or music because the processor's behavior is optimized for these signals.

5.b) 15kHz Phase-Corrected Lowpass Filter: (on Cards #8 and #9)

The signal from the hf limiter is applied through R801 to clipper CR801, CR802. The clipper output is applied to inverting amplifier IC801A through R802.

The output of IC801A (the clipped signal) is filtered by fifth-order Cauer lowpass filter R805, R807, C801, C802, C803, C804, C805, C806, C807, L801, L802, to remove high frequency energy generated by clipping which would otherwise induce "aliasing" distortion in the stereo generator. This lowpass filter is realized as a "passive ladder" for maximum stability. The filter's response is nominally +0, -0.6dB from 0 to 15.4kHz, with a sharp rolloff thereafter. There are two notches, the first of which occurs at 19kHz to provide extra protection for the pilot tone.

The load resistor for the filter, R807, is connected not to true ground, but to the "virtual ground" of the summing junction of IC801B. IC801B is an inverting, frequency selective amplifier, whose feedback network R808, R809, C808 provides a 2dB shelving rolloff. The purpose of this rolloff is to match the gentle rolloff of the 2.2kHz sidechain filter (see **5.c)** so that the distortion is correctly cancelled.

The output of IC801B feeds a differential sidechain which creates a fourth-order allpass function when its output is correctly summed with the output of IC801B (i.e., the main signal). This allpass function does not change the frequency response of the 15.4kHz lowpass filter, but does add phase shift as necessary to make the overall time delay of the 15.4kHz filter plus allpass network more constant than the time delay of the 15.4kHz filter alone.

Basically, this phase corrector sidechain consists of a pair of active inverting bandpass filters built around IC802A, IC802B. The IC802A filter is driven by the output of IC801B through R815. Its output is summed into summing amplifier IC803B through R817.

The second bandpass filter (associated with IC802B) is driven by both the main signal (through R814) and the output of the first bandpass filter IC802A (through R818). The output of IC802B sums into IC803B through R820.

The third input to IC803B is the main signal (through R810). The fourth (and final) input is the output of the 2.2kHz sidechain distortion-cancelling filter (through R822).

To restore flat response, the frequency response of this summation is boosted by passing the signal through a shelving filter complementary to the one realized in the feedback network of IC801B. The shelving boost is created by the R811, R812, R813, C809 in the feedback network of IC803B.

The output of IC803B is the output of the Smart Clipper. Because it contains substantial overshoots, it is connected to the FCS overshoot compensator (see **6** below).

5.c) 2.2kHz Distortion-Cancelling Filter Sidechain: (on Cards #8 and #9)

The distortion-cancelling sidechain takes the difference between the input and output of the clipper (which is the distortion added by the clipper). It then lowpass-filters this difference at 2.2kHz (thus substantially reducing the peak level), and finally sums the difference back into the main signal (at IC803B) to cancel clipping-induced distortion below 2.2kHz.

If no clipping occurs, the signal at the input side of R801 is equal and opposite to the signal at the output of inverting amplifier IC801A and no differential signal is produced. Because of the polarity reversal in IC801A, the difference between the input signal and the clipped output is derived by a simple summation through R804, R805.

This signal (our desired difference signal) feeds filter C814, L803, A1, whose magnitude and phase, when cascaded with additional rolloff R821, R822, C815, match the magnitude and phase of the phase-corrected 15.4kHz lowpass filter (see **5.b** above) from 0 to 2.2kHz. IC803A is a unity-gain buffer to drive this final rolloff network, the output of which is directly summed into IC803B.

6) Frequency-Contoured Sidechain (FCS) Overshoot Compensator:
(on Cards #8 and #9)

Overshoots are derived from the input signal by center clipper IC804A, R823, R824, R825, R826, CR803, CR804. This circuit is a differential amplifier which subtracts the output of a clipper from the clipper's input. This clipper consists of Schottky diodes biased with approximately ±4.2v, and is therefore substantially "harder" than the first clipper (associated with IC801A).

If the output of IC804A were simply added to its input, the sum would be a clipped signal; a "differential clipper" would be created. However, the output of IC804A contains clipper-induced high frequencies which could cause "aliasing" in the stereo generator. The output of IC804A is therefore lowpass-filtered by passive 15kHz ladder filter R828, R829, C816, C817, C818, L804, L805, before being added back into the input signal to cancel overshoots. This filter has a response that rises 4dB at 15kHz. This makes up for the loss of high frequencies which would otherwise reduce the peak level of the overshoots emerging from the filter. To compensate, the fundamental levels around 15kHz are increased by the frequency-contouring.

(**NOTE:** This "Frequency-Contoured Sidechain" overshoot compensation scheme is protected by U.S. Patent #4,460,871.)

The filter has phase shift. To assure correct addition of the filtered overshoot, the input signal is delayed in a modular phase shift network, A1, which whose amplitude response is flat, but which accurately matches the phase response of the sidechain filter throughout its passband. A1 is also equipped with a summing input for the overshoot signal, which appears at the output of IC804B.

The time delay of this network is not constant at all frequencies. The output of A1 is thus passed through allpass network IC805A, R831, R832, R833, R834, C819, C820 to create constant time delay from the input of the overshoot compensator system to its output. The allpass network has a flat amplitude response, but frequency-dependent phase response. (This network is designed to also compensate for the non-constant group delay of the following third-order lowpass filter).

The output of IC805A is passed through a composite capacitor (consisting of two aluminum electrolytics back-to-back, bypassed by a polycarbonate) to remove accumulated DC offsets. Recent research has indicated that this sort of composite structure minimizes the audible degradation caused by passing audio signals through polar capacitors with high dielectric absorption.

This capacitor is necessary because a differential DC offset between the left and right channels translates into a DC L-R component which, if not nulled, results in a fixed 38kHz component in the stereo generator output. This is equivalent to lack of 38kHz subcarrier suppression. Whatever fixed DC offset that is found at the output of the left and right audio processing channels is ordinarily nulled out by stereo generator alignment control R714. However, drifts in the differential offset in the order of 40mV can reduce the 38kHz suppression to less than the -40dB minimum specified by most government broadcast authorities.

The signal then passes through a third-order active 15kHz lowpass filter IC805B, R836, R837, R838, R839, C822, C823, C824 to provide further reduction of any remaining out-of-band energy above 19kHz.

Finally, to catch any slight errors made by the overshoot compensator, the signal is applied to safety clipper R840, R841, CR805, CR806. The basic overshoot compensator is extremely effective; thus, the safety clipper is virtually never active and no additional filtering is included after its output.

The output of the safety clipper is buffered by IC806A, which ordinarily drives the stereo generator. The output of IC806A (which is the audio processing output) also appears at the rear-panel TEST jacks when the rear-panel switch is in NORM.

7.a) Stereo Generator (General): (on Card #7)

The principles of operation of the stereo generator have already been discussed in **Appendix A.** The detailed operation of each of the sub-systems in the stereo generator will be discussed below.

7.b) 19kHz Oscillator: (on Card #7)

The crystal oscillator employs an AGC loop to enable linear oscillation. The output of the oscillator, IC705, is a 19kHz sinewave with typically less than 0.07% harmonic distortion.

The output of IC705 is reduced in level by voltage divider R747, R748 to avoid overdriving crystal Y701. Oscillation can be suppressed by turning JFET Q709 ON (by applying 0 volts to its gate), thus shorting R747 and virtually eliminating drive to Y701. Y701 operates in its series-resonant mode as a positive feedback element into the "+" input of IC705. Simultaneously, negative feedback is taken around IC705 through R742. The effective gain of IC705 is determined by the resistance between the "-" input of IC705 and ground. This resistance is the sum of R741 and the drain-source resistance of JFET Q708, operated here as a voltage-controlled resistor whose resistance is controlled by its source-gate voltage. As will be seen below, this voltage is determined by the 19kHz crystal oscillator AGC loop.

C709 prevents significant DC offsets from being developed at the output of IC705.

Approximately one-half of the AC source-drain voltage of Q708 is fed back into its gate through R743, R744, C710 to cancel second-harmonic distortion which would otherwise be introduced by Q708.

The positive peak output level of IC705 is sampled by comparator IC706, whose "+" input is biased to a DC reference voltage of +2.92 volts. If the peak output of IC705 attempts to exceed this level, IC706 turns on and charges C711 (towards -15 volts) through R749 and CR701. This voltage is coupled to the gate of gain-control FET Q708 through R744, thus closing the AGC loop.

Recovery time for the AGC loop is determined by R745. CR702 internally clamps IC706 to reduce saturation effects and speed its response time. C712 is a power-supply bypass capacitor.

7.c) 19kHz Doubler: (on Card #7)

The 19kHz output of IC705 is applied to both halves of dual comparator IC707. A -2.09 volt bias voltage is applied to pin 5 through R752, R753; a +2.09 volt bias is applied to pin 2 through R754, R755. The voltages and polarities are arranged so that each time the 19kHz sine passes through 45 degrees + (n x 90 degrees) (n=1,2,3...), the output of IC707 switches from high to low, or vice-versa. Its output is thus a 38kHz square wave 90 degrees out-of-phase with its 19kHz input. The low level at IC707's output is approximately -0.5 volts; the high level is controlled by the voltage applied from the output of IC709A to strobe terminals 9 and 13 of IC707. As will be seen below (see **7.e**), this closes the AGC loop which controls the amplitude of the 38kHz sinewave derived by filtering the output of IC707.

7.d) 38kHz Filter And Phase Shifter: (on Card #7)

The squarewave output of IC707 is connected to a high-"Q" "Q-enhanced" active bandpass resonator IC711, R775, R776, R777, R778, R779, R780, C718, C719, Q711, employing both positive and multiple-negative feedback to minimize sensitivity to component drifts and tolerances. The nominal "Q" of the resonator is 40.

This resonator is tuned by a control voltage applied to the gate of Q711, a JFET operating as a voltage-controlled resistor. Varying the resistance of Q711 varies both the center frequency and "Q" of the resonator. As will be seen below (see **7.f**), voltage control is derived from the Phase-Locked-Loop (PLL) circuit.

If the sum of the resistance of R777 and Q711 becomes too low (below about 780 ohms, nominally), then IC711 will oscillate at approximately 38kHz and produce a reasonably pure sinewave output. The circuit has been designed with considerable safety margin to prevent this from happening as long as all passive components in the circuit approach their specifications for temperature stability and long-term drift. However, oscillation might occur if, for example, one of the components in the circuit were to exhibit severe drift. If this oscillation occurs, it can cause considerable confusion because the oscillation is easily mistaken by a service technician for the desired 38kHz component. Yet the oscillation is not precisely locked to the 19kHz pilot, is not controllable by the 38kHz AGC loop, and cannot be suppressed by turning off the 19kHz oscillator.

The output of IC711 contains perhaps 0.5% THD. To create a 38kHz sinewave in which all spurii are down at least 80dB, the output of IC711 is connected to a passive third-order elliptical lowpass filter R781, C720, C721, C722, L701. This filter is essentially flat to 38kHz, and provides substantial rejection for all harmonics. It exhibits a notch at 114kHz (third harmonic).

The output of the passive filter is buffered by IC712. Note that not all opamps of the generic type specified will work satisfactorily in this socket, and that the factory checks each opamp to make sure that its second harmonic (76kHz) distortion remains greater than 80dB below the 38kHz level.

Recalling that the 38kHz output of IC707 is 90 degrees out-of-phase with the 19kHz pilot, note that the total phase shift through both active and passive filters has been designed to create the necessary additional 90 degree shift to bring the 38kHz back into the proper phase relationship with the 19kHz pilot. This relationship is <u>precisely</u> maintained despite drifts in the filters by means of the 38kHz PLL circuit.

7.e) 38kHz AGC Loop: (on Card #7)

The output of the 38kHz passive filter is buffered by IC712. IC712's output is connected to the "-" input of IC710 .

IC710 is used as a comparator. Its "+" input is connected to the wiper of R772 (L-R GAIN), which provides a DC reference voltage for IC710. If the peak voltage of the 38kHz sine at the output of IC712 attempts to rise above the DC reference voltage at the "+" input of IC710, then IC710 will produce a negative-going output which discharges C715 through CR703 and R765. Under conditions where IC710 is non-conductive, the quiescent voltage on C715 is set at approximately +6VDC by means of voltage divider R763, R764. These resistors also determine the recovery time of the AGC loop.

C715 is buffered by means of unity-gain voltage follower IC709A. The output of IC709A is applied to the "strobe" inputs of IC707, thus adjusting the amplitude of the 38kHz square wave output from IC707 and completing the feedback loop.

7.f) 38kHz Phase-Locked Loop (PLL): (on Card #7)

The 38kHz output of IC712 is applied to the "+" input of one-half of dual comparator IC708 (used as a phase detector). The "-" input is grounded. The output of this "38kHz" half of IC708 is therefore a symmetrical 38kHz squarewave.

The output of the 19kHz oscillator IC705 is applied to the "-" input of the other half of IC708. The "+" input of this "19kHz" half is biased at +1.69VDC by means of voltage divider R759, R760. The output of the 19kHz half of IC708 is thus an asymmetrical 19kHz square wave.

The outputs of the two halves of IC708 are "OR"ed together. The output of IC708 is thus a 38kHz pulse whose duty cycle depends on the phase relationship between the 19kHz and 38kHz.

This pulse's baseline is approximately -0.5VDC; its peak is approximately +4.5VDC. The pulse is level-shifted by passage through C714, and then coupled to switching transistor Q710 to turn it on and off.

When Q710 is on, it saturates and essentially connects R767 to the +15V supply. The other side of R767 is connected to the "-" input of inverting integrator IC709B. Feedback forces this "-" input to be within a few millivolts of ground. Thus, when Q710 is on, exactly 1 mA flows into the summing junction of IC709B and is integrated by IC709B in conjunction with feedback capacitor C716. When Q710 is off, essentially no current flows. CR706 prevents the output of IC709B from going more than 0.7V positive, thus protecting C716 and Q711.

The feedback loop is closed by connecting the output of IC709B to the gate of Q711. Varying the voltage at the gate of Q711 varies its resistance, thus retuning 38kHz resonator IC711 and associated components (see 7.d above) and changing the phase shift through IC711.

Current is removed from the IC709B summing junction through R768 and R769 (PILOT PHASE). Because of the integrator, the average current into the summing junction of IC709B must be zero. Otherwise, the output of IC709B would eventually go to the "+" or "-" power supply rails and saturate.

Feedback from the output of IC709B thus adjusts the phase shift in IC711 (and thus the duty cycle of the current pulses through R767) until the average current into the "-" input of IC709B is zero. The amount of current removed from the "-" input of IC709B is adjusted by the operator (by means of R769) until the desired phase relationship between the 19kHz pilot and 38kHz output of IC712 is achieved.

Once adjusted, the stability of this relationship is primarily dependent upon the stability of phase detector IC708. The 38kHz side is referenced to ground. Therefore, changes in the amplitude of the 38kHz input will have virtually no effect upon the duty cycle of the output of the 38kHz side of IC708. However, the duty cycle of the 19kHz side of IC708 is highly dependent upon maintaining the amplitude of the 19kHz constant. Ordinarily, this is effectively done by means of the 19kHz oscillator AGC loop (see 7.b). A failure in this loop can therefore cause drifts both in pilot level and pilot phase.

In addition, leakage in Q710 can cause instability problems, particularly if the leakage changes with time and/or temperature.

7.g) L-R Amplifier: (on Card #7)

The difference signal is derived by means of differential amplifier IC701 and associated components. The left input is applied to R707, which is the inverting (-) input of the differential amplifier. The right channel is applied to R708, which is the non-inverting (+) input of the differential amplifier.

Trimpot R709 (MAIN-SUB CROSSTALK) adjusts the non-inverting gain of the amplifier without affecting the inverting gain. This permits precise cancellation of common-mode (L+R) input components, and thus allows the linear main-channel to subchannel crosstalk (which represents L+R components which have leaked into the subchannel) to be nulled.

The non-inverting gain of the differential amplifier is switched from +1 to +2 by turning off Q705 (by placing -15V on its gate). Ordinarily, the amplifier requires a gain of +1. However, in the SUB-TO-MAIN crosstalk test mode, the amplifier must take a gain of +2 to create the same peak modulation level that a given input level would create in the NORMAL operating mode.

7.h) 38kHz Doubly-Balanced Modulator: (on Card #7)

The operation of the modulator is very similar to the operation of the voltage-controlled amplifiers used in the audio processing section, and the reader should first refer to **2.c** above.

The essential difference between operation of the audio VCA's and operation of the 38kHz modulator is that the audio VCA's are operated in a divider mode, and the 38kHz modulator is operated in a multiplier mode. Thus control current in IC703B is varied to perform the multiplication function.

Control current in IC703B is the sum of a bias current (through R727) and a 38kHz modulating current which is coupled through R726 and DC blocking capacitor C704. (Pin 3 of IC703B ordinarily sits at approximately -13.5V). The bias current is necessary because pin 3 can only accept control currents of one polarity. However, it also means that the 38kHz modulator multiplies its L-R input times a constant, thus passing some L-R directly without modulating it by 38kHz. This feedthrough component (which undergoes a phase reversal when passing through the modulator) is cancelled by feeding in-phase L-R around the modulator through R734. Final adjustment of the feedthrough cancellation (which is equivalent to nulling subchannel to main-channel crosstalk) is effected by slightly trimming the bias current in IC703A by means of R717 (SUB-MAIN CROSSTALK), thus varying the gain of the modulator. (**NOTE:** This adjustment will also affect L-R gain and thus separation. If for some reason R717 is adjusted, R772 will also have to be slightly trimmed.)

Cancellation of the residual 76kHz second-harmonic output of the modulator (which results from unavoidable non-linearities) is effected by passing a very small amount of 38kHz from R725 (76KHZ NULL) through R723 into the "-" input of IC703B. This component is multiplied in IC703B by the main 38kHz component to create a 76kHz component of correct amplitude and phase to cancel the basic 76kHz feedthrough in the modulator.

The basic feedthrough is in the order of -70dB; this circuit typically reduces it to slightly below -80dB. It can therefore be seen that the circuit is not terribly critical, and moderate drifts with time and temperature will not result in difficulties.

The circuit affects only the 76kHz component, and not sidebands which appear around it due to L-R modulation. These sidebands are a function of (1) basic modulator non-linearity, and (2) any residual second harmonic appearing in the 38kHz modulator input (from the output of IC712).

If the 76kHz sidebands have excessive energy, they can interfere with SCA service. In most production units, these sidebands are down -75dB or more below 100% with pure L or R modulation, and will of course disappear with pure L+R, since no audio is applied to the modulator in this case. This performance is far better than required to fully protect an SCA. If interference to SCA suddenly develops, and a spectrum analyzer connected directly to the composite output of the 8100A/1 reveals high levels of 76kHz sidebands, then the most likely sources of difficulty are IC703 itself, or else substantial second-harmonic distortion at the output of IC712. IC703 is a specially selected and graded device, and replacements must be obtained from the Orban factory. (**NOTE:** Replacement of IC703 requires a complete stereo generator realignment as detailed in **Appendix E**).

The output of IC703B is in the form of a current. This is converted to a voltage by means of output summing amplifier IC704.

Q706 switches the "stereo" components onto IC704's summing junction depending on whether the 8100A/1 is in "stereo" or "mono" mode. These "stereo" components include the output of IC703B, the L-R cancellation component from R734, the 19kHz pilot, and a DC offset null current.

The pilot is summed through R733 from the wiper of PILOT INJECTION trimmer R731. The DC offset component is summed through R729, R730 from the wiper of DC OFFSET trimmer R726. C705 bypasses any power supply noise.

The purpose of the DC offset adjust is to render an output coupling capacitor unnecessary. Such a capacitor would have to be a very large polar electrolytic (approximately 470uF or greater). Not only would such a capacitor compromise low frequency separation slightly, but its existence would also tend to compromise the extreme transparency of the basic 8100A/1 signal path, since recent research has indicated that such capacitors audibly degrade sound quality.

7.i) L+R/Mono Path: (on Card #7)

The L+R component at the output of the 8100A/1 is created by passively mixing L and R, and then by summing the mixture into IC704. L and R pass through precision-matched resistors R701 and R702 and then through switching FET's Q701, Q702. When driver transistors Q703, Q704 are off, the gates of Q701, Q702 are connected to their drains through R703, R704, thus holding Q701, Q702 on. When Q703, Q704 turn on, they turn off the left or right channels respectively by pulling the gates of Q701, Q702 close to -15 volts and turning Q701, Q702 off.

The sum is attenuated by virtue of R701 and R702's being the top of a voltage divider, the bottom of which is the resistor network R735, R736, R737. R735 is simply the summing resistor into IC704. The attenuation of the overall voltage divider is controlled by adjusting the resistance of its lower leg. This is done by turning Q707 on or off. In stereo mode, Q707 is on and results in highest attenuation. In mono mode, the attenuation must be reduced by 6.84dB to permit a single channel (without pilot) to modulate the transmitter to 100%. This is achieved by replacing the very low on-resistance of Q707 with the higher resistance of R737.

7.j) Output Summing Amplifier: (on Card #7)

Little need be said about this amplifier which has not been stated above. IC704 is a conventional inverting summing amplifier. Its gain is adjusted by means of its feedback resistor, OUTPUT ATTENUATOR R738. IC704 is a fast 518-type amplifier, and is somewhat prone to instability when loaded with capacitive loads such as coax. To isolate the amplifier from such loads, R739 is placed between the output of IC704 and the 8100A/1's composite output. The source impedance of the composite output is thus 470 ohms regardless of the setting of R738.

7.k) Mode Switching Logic: (on Card #7)

The three basic logic states -- stereo, mono left, and mono right -- are "remembered" by three CMOS NAND gates in IC714. The output of each gate is connected to an "output bus" through a 47K isolation resistor R783, R784, R785 to permit pulling a given output bus down without damaging its associated gate. "Negative logic" is used; a sub-mode is ON when its output bus is at -15V, and OFF when its output bus is at ground. The two inputs of each gate are connected to the two output busses of the other two gates. Thus, if either of the other gates is ON, the gate in question is held OFF. However, if the output bus associated with the gate in question is externally pulled ON, the other two gates are pulled OFF; thus the gate in question is latched ON. Logic switching is effected by momentarily switching any of the gate output busses to -15V; this is done by forcing the transistors in optoisolators IC715, IC716, IC717 to conduct by passing current through their LED's (remote control), by operating the momentary front-panel STEREO/MONO switch, or by the power-up circuit, which uses R786, C723, and CR712 to hold a selected output line at -15 for a fraction of a second after powerup.

Failures in the logic will almost certainly be due to failures of IC713-IC717, or to failures in the JFET switching transistors. All of these components can be freely replaced as necessary without readjustments.

Note that the phototransistors in IC715-IC717 have had their base leads (pin 6) cut off flush with the IC package. This is because the base lead is extremely sensitive to leakage currents, and condensing moisture is quite sufficient to cause false switching. It is therefore extremely important that the base lead be cut off if any of these optoisolators are ever replaced.

Essentially no further circuit description is required, as most of the circuitry is integrated onto the CMOS logic chips. The discussion of operational principles supplied in **7.k** of **Appendix A,** plus examination of the truth table found on the schematic for Card #7, should suffice to permit full understanding of the operation of the logic. (Some readers may be unfamiliar with the Exclusive-OR gate, IC713. This logic element operates such that its output is at -15 volts when its inputs are the same, and at 0 volts when its inputs are different.)

8.a) Unregulated Power Supply: (on chassis outside RF-tight enclosure)

The unregulated power supply is wholly conventional. It consists of a dual-primary transformer T101, two full-wave rectifiers CR101, CR102 and CR103, CR104, and two energy storage capacitors C101, C102.

T101's primary may be switched for 115 volt operation by paralleling its two primaries, or for 230 volt operation by connecting its two primaries in series. RF filtering is provided on the AC line by means of FL101. In addition, VHF and UHF RF is filtered from the unregulated DC supply lines as they enter the main chassis by means of C103, C104, C105, C106, C107, L101, L102. The RF suppression scheme divides the chassis into three major sections. The section to the left contains the AC wiring and the unregulated power supply, and is assumed to contain some RF. The main chassis, to the right, uses RF suppression on each line entering or leaving the area, and is thus RF-free. The RF filter box, on the rear panel, interfaces the audio input and output lines with the outside world. It contains the input pads. Its connections to the main RF-tight compartment are all RF-filtered.

8.b) +15 Volt Regulator: (on Card #1 -- rear chassis apron)

The +15 volt regulator is the main reference for all other voltages in the OPTIMOD-FM system. It employs a 723C IC voltage regulator IC101 in conjunction with an external series-pass transistor Q101. This transistor is mounted on the rear apron of the chassis, which serves as a heat sink.

The 723C contains a reference voltage source, an opamp (externally compensated by means of C109 to prevent oscillation), and a current limiting transistor. The reference voltage (nominally +7.15 volts) is developed at pin 6. C108 filters high frequency noise from the reference voltage. The reference voltage is directly connected to the non-inverting input of the internal opamp, pin 5. Voltage divider R105, R106, R107 develops a precise fraction of the output voltage of the regulator at the wiper of R106. R106 adjusts this fraction. The wiper of R106 is connected to the inverting input of IC101's internal opamp. Negative feedback thus forces the voltage at the wiper of R106 to be equal to the reference voltage. Thus the output voltage of the regulator is always the reference voltage divided by the voltage divider gain.

The output current flowing through Q101 develops a voltage drop across R103. When the current exceeds approximately 3/4 amp, said voltage drop is sufficient to turn on the current-limit transistor inside IC101, whose base-emitter junction is connected to pins 2 and 3 of IC101. The current-limit transistor then shunts base drive current from the external series-pass transistor Q101 and prevents damage due to overheating.

If a catastrophic failure in the +15 volt regulator causes it to lose control over its output voltage, the rest of the circuitry must be protected against the full unregulated voltage, or the entire system will be severely damaged. This protection is provided by zener diode VR101, CR105, and 1 amp fast-blo fuse F102.

In the event that the regulator loses control of the output voltage, VR101 will conduct and limit the output voltage to approximately 16.5 volts, which will not damage the system. Extremely large amounts of current will flow in VR101. However, before VR101 is damaged, this current will blow F102, thus disconnecting the circuitry from the unregulated supply. VR101's clamping action will also prevent the negative tracking supply from going any higher than -16.5 volts. If the regulator is operating properly, the current limiting circuitry will prevent F102 from blowing even if the regulator output is short-circuited.

Under certain unusual circumstances, the regulator may lose control of its output voltage, yet the current limiting circuit may still work. If this occurs, F102 will not blow, and VR101 will overheat and burn out. Fortunately, its failure mode is a short-circuit. It will therefore still protect the OPTIMOD-FM circuitry even in this exceptional circumstance.

8.c) -15 Volt Regulator: (on Card #1 -- on rear chassis apron)

The -15 volt regulator is an operational amplifier containing a discrete power-booster output stage with current limiting. It "amplifies" the output of the +15 volt regulator by -1, thus producing a -15 volt tracking supply. Shutdown of the +15 volt supply (due to current limit conditions or to a fault which blows F102) will also result in the -15 volt supply's shutting down.

The basic opamp is IC102; its input resistor R109 and feedback resistor R108 are equal-valued, resulting in a gain of -1 ±2%. IC102's negative supply comes from the unregulated -22 volt supply. The common-mode range of the 301A opamp includes the positive power supply, thus permitting operation with IC102's positive supply at ground. Under normal operating conditions, the "+" input of IC102 is grounded, and its "-" input is within 10mV of ground.

Q103 and Q102 form a conjugate emitter follower which can boost the output current of IC102 to more than 3/4 amp. The basic emitter follower is Q103; Q102 is connected in a 100% negative feedback configuration to boost the current output capability of Q103.

Q104 is a current-limit transistor. If the -15 volt supply is called upon to deliver more than 3/4 amp, sufficient voltage drop (approximately 0.6 volts) will occur across R104 to turn on Q104, thus shunting drive current away from Q103 into the load and protecting Q102/Q103 from burnout. Under these conditions, IC102 is protected by internal current limiting circuitry.

C113 frequency-compensates the -15 volt supply to protect it against high frequency oscillations. R102 increases the circuit's immunity to leakage in Q103.

The rest of the circuitry is protected against a catastrophic failure of the -15 volt regulator by means of zener clamp VR102, CR106, and fuse F103. The operation of this circuit is identical to the operation of the corresponding circuit in the +15 volt regulator (see 8.b).

8.d) Miscellaneous Voltage Supplies:

The operation of these supplies is extremely straightforward. No further explanation beyond that given in **Appendix A** is required.

APPENDIX C:
User Access

ROUTINE ACCESS

The first part of this Appendix describes how to access those parts of OPTIMOD-FM ordinarily involved in setup, adjustment, or alignment. (The second part of the Appendix provides information on the disassembly techniques necessary to access the balance of the circuitry.)

a) User Adjustments: To access the user adjustments, open the small access door using the key furnished. This will reveal all user-adjustable controls.

b) Line Fuse, Power Switch, and Line Voltage Selector: These are accessed by swinging down the entire front panel, which is hinged at the bottom. To avoid damage, this should be done only with the small access door locked. Using the 5/64" hex wrench supplied, remove the three hex-socket screws at the top of the front panel and carefully swing the panel out and down.

c) Circuit Cards: First, swing the front panel down (see **b**). You must then remove the subpanel by first loosening four DZUS fasteners by turning each one-quarter turn counterclockwise with a long 3/16" or 1/4" slotted-blade screwdriver. Taking care not to stress the flat cable beneath it, tilt the top of the subpanel outward and leftward to clear the upper chassis lip and the door support bail at the right. The PC cards may now be removed from their slots.

**** This procedure is directly reversible with cautions:

-- The subpanel should always be replaced to protect the cards from RFI.

-- DZUS fasteners turn only 1/4-turn. Don't force them, lest they be damaged in a way that is very time-consuming to repair.

NOTE

OPTIMOD-FM Model 8100A/1 shares chassis components with other Orban products. In the 8100A/1, the first available card slots are not used, and there is no Card #2. The power supply regulator card is Card #1 (which is mounted on the rear panel).

SERVICE ACCESS

General Cautions: These apply to all the procedures described below.

-- For best RFI protection, replace all screws and tighten normally to achieve firm contact.

-- If screws are lost, replace them with screws of the same length, since longer screws may cause mechanical interference or internal short circuits.

-- Most screws used in OPTIMOD-FM are binding head to achieve secure fastening without lockwashers. If a pan head screw is substituted, use an internal star lockwasher to retain this security.

-- Plating on all screws is Cadmium type II. Almost any other plating is acceptable unless corrosive atmosphere is present.

a) Cover Removal: Removing the top or bottom covers is tedious because thirty screws must be removed. (The large number of screws is necessary to achieve an RF-tight seal.) Luckily, most service access can be achieved without removing either cover! Specific instructions for doing this are found below.

If you wish to remove either cover, simply remove all thirty screws.

**** This procedure is directly reversible with cautions:

-- When replacing a cover, align it as closely as possible with the corresponding holes, and start all screws. After all screws have been started, tighten all screws to normal tightness, "inland" screws first.

b) Access To Area Behind Rear Panel: If the covers are still in place, they needn't be removed.

Remove eight screws holding the top cover to the flange of the rear panel. Remove the corresponding eight screws from the bottom cover. The rear panel will remain solidly in place.

Set the chassis, bottom cover down, on a pad on a table. Allow 6" (15cm) between the rear panel of the chassis and the table edge. Unplug the power cord.

Now remove three groups of three screws which are circled in black on the rear panel.

VERY carefully and slowly, pull the rear panel about 3/4" (2cm) toward you, and tilt the top edge down until the rear panel is horizontal and resting on the table.

CAUTION

Watch for snags in the internal wiring, and for any stress on the ceramic feedthrough capacitors on the divider wall or input filter box. These capacitors are very fragile and difficult to replace.

**** This procedure is directly reversible with cautions:

-- When positioning the rear panel over the corresponding holes, make sure that no wires are pinched under the flanges. Start, but do not tighten all nine screws. Observe the areas where the flanges on the rear panel meet the flanges on the side panels. Adjust the rear panel so that the flanges line up in order to provide a flat mounting surface for the cover when tightened.

c) Access To RF Filter Card: First open the rear panel (procedure **b** above).

Remove the four screws holding the Input Filter Box to the rear panel. **VERY** carefully and slowly, tilt the metal box back to vertical, taking care to avoid snagging the internal wiring and stressing the ceramic feedthrough capacitors.

This will reveal the internal circuit card, which is attached to the rear panel by four #4-40 screws. While this card can be removed for component replacement, it is easier (though less workmanly) to clip out the defective component from the topside and to install its replacement by tack-soldering to the old leads.

**** This procedure is directly reversible with cautions:

-- If components have been replaced, make sure that reassembly will not result in crushing of the component against the rear panel.

-- Tilt the box back to horizontal (so it rests against the rear panel) very slowly and carefully. Watch for wire snags and dress wires appropriately. Make sure that no wires are crushed under the flange.

d) Access To Unregulated Power Supply Chamber: If the covers are not already removed, remove the five cover screws which attach the top cover to the flange of the side panel. Remove the corresponding five screws from the bottom cover.

Open the front panel.

Remove the shoulder screw that attaches the door-support bail to the left chassis wall. Note that there is a nylon washer between the bail and chassis wall to prevent scraping.

Turn the chassis so that the left wall is facing you. Remove the left rack flange by removing the six unrecessed screws.

Remove the three screws that attach the rear panel to the main (steel) side panel.

Remove the remaining six screws and gently lift off the side panel by pulling outward.

**** This procedure is directly reversible with cautions:

-- Position the steel side panel and start, but do not tighten, all nine screws. Observe the areas where the flanges meet the rear panel and internal bulkhead, and align the flanges so that the covers will seat on a flat mounting surface.

e) Removal Of Card #1 (The DC Regulator) From Rear Panel And Power Transistor Replacement: Because the removal procedure is complex, this card was designed to permit many servicing operations to be performed without removing the card from the chassis.

The plastic transistors and some capacitors are socketed in very tiny sockets pressed into the card. IC's are conventionally socketed. Many unsocketed components can be replaced from the topside by tack-soldering the new component to the lead stubs of an old clipped-out component.

If the card must be removed, do it as follows:

CAUTION

The rear panel serves as a heat dissipator for the power transistors. Proper contact is necessary to insure sufficient transistor cooling. Please follow instructions carefully.

Remove the four press-fit plastic plugs on the power transistor covers with a pair of chain-nose pliers. This will reveal the transistor mounting screws. Remove the four screws holding the power transistors.

VERY carefully and slowly pull each transistor from its socket. If, as you do this, the silicone rubber insulator tends to stick to the panel, release it from the panel such that it sticks to the bottom of the transistor instead. After you remove each transistor, press its insulator back in close contact with it pending reinstallation.

NOTE

These insulators form themselves to the bottom surface of each transistor. Since they take a "set", they should not be interchanged or reversed. If you have to replace a power transistor, you may re-use the insulator if it is in good condition. With care, it will re-form itself as necessary. Otherwise, use a conventional mica insulator and white silicone heat-conducting compound.

Open the rear panel (procedure **b**). With the transistors removed, it is possible to release the circuit card from its plastic post mounts by squeezing the tangs in each of the four corners to permit pulling the card off the posts.

**** This procedure is directly reversible with cautions:

-- See the discussion above regarding heat-conduction insulators.

-- The screws mounting the transistors should be tightened evenly. For best thermal contact, tighten each screw a small amount, alternating between screws. Tighten securely, but not enough to damage the threads in the sockets.

-- Note that there must be a split lockwasher under each screwhead to accommodate thermal cycling.

-- The Thermalloy plastic cover does not attach in a conventional or readily obvious way. It rides on the circumference of the special split lockwasher and does not (and should not) become captured under the head of the screw. Consequently, the cover may be slightly loose even after screws are tightened securely. This is normal, and should not (and cannot) be corrected.

-- Be sure to reinstall the press-fit plugs that cover the screwheads.

APPENDIX D:

Field Audit-of Performance Procedure

GENERAL

This Appendix provides instructions enabling Model 8100A/1 users to check the performance of their units using test equipment likely to be found in a well-equipped radio station. This procedure is a starting point for detecting and diagnosing a problem that you believe is caused by the 8100A/1. It is also useful in routine maintenance, and can be used at Proof time to check routine equipment performance, thus providing more data than the Proof alone provides. By its nature, it is limited in scope to discovering static problems. A dynamic problem in the AGC circuitry (caused by the failure of a timing module on Card #5, for example) would not tend to be discovered by performing these tests.

For this reason, measurements must always be complemented by listening. If you are well-acquainted with the "sound" of the 8100A/1 as adjusted for your format, then faults that develop will ordinarily be readily detectable by ear.

If audio problems develop, many engineers immediately tend to blame their processing. However, as is the case with any processing, faults in the audio equipment preceding OPTIMOD-FM will be magnified by the action of the processing. Program material that is marginally distorted at the 8100A/1 input, for example, is likely to be unlistenable by the time it emerges from the output when aggressive processing is used. In addition, be sensitive to possible defects in the monitoring equipment; verify that a problem can be observed on at least two receivers before pushing the panic button.

REQUIRED EQUIPMENT

a) Audio Oscillator. An ultra-low-distortion type like the Sound Technology 1710B is preferred. However, a Heathkit or Eico-type oscillator (such as Heath IG-72) can be used to obtain approximate results, provided that residual distortion has been verified to be below 0.1%.

b) Noise and Distortion (N&D) Test Set. Once again, a high-performance type like the Sound Technology is preferred, but not required.

c) General-Purpose Oscilloscope. DC-coupled, dual-trace, with at least 5mHz vertical bandwidth. This is used to monitor the output of the N&D Test Set, and also to check the stereo generator.

d) Type-Approved Stereo Monitor. The Stereo Monitor will be directly connected to the 8100A/1 baseband output, and its associated FM monitor is therefore not required.

AUDIO PROCESSING

It is often more convenient to make measurements on the bench, away from high RF fields which might otherwise affect results. For example, in a high RF field, it is very difficult to accurately measure the very low THD produced by a properly-operating 8100A/1 at most frequencies. However, in an emergency situation (is there any other kind?!), it is usually possible to do measurements under high-RF conditions which will reveal many of the grosser faults which could develop in the 8100A/1 circuitry.

The audio processing section of the 8100A/1 can be measured independently of the stereo generator by using the rear-panel TEST JACKS as the point to which measuring equipment is connected. The following procedure assumes that all test excitations are applied to the rear-panel main audio input terminals, and that all responses are measured at the TEST JACKS.

a) **Standard Control Setup:** Record the normal settings of the controls so that they can be reset after the measurements have been completed. Then set the controls as follows:

L and R Input Attenuators:	0
Clipping:	+2
Release Time:	10
Bass Coupling:	10
Gate Threshold:	0
H-F Limiting:	0

b) **Skeleton Proof:** This should be performed for both left and right channels.

1) Place both PROOF/OPERATE switches in PROOF.

2) Connect a low-distortion audio oscillator to the 8100A/1 input. Set the frequency to 15kHz, and adjust the oscillator output level to produce 3.3V rms at the 8100A/1 output.

3) Connect the input of the N&D test set to the 8100A/1 output through a 75us (or 50us, if that is your country's standard) deemphasis network. (If the N&D set has its own deemphasis available, this is obviously not necessary.)

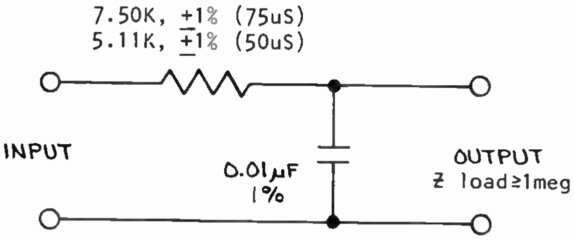
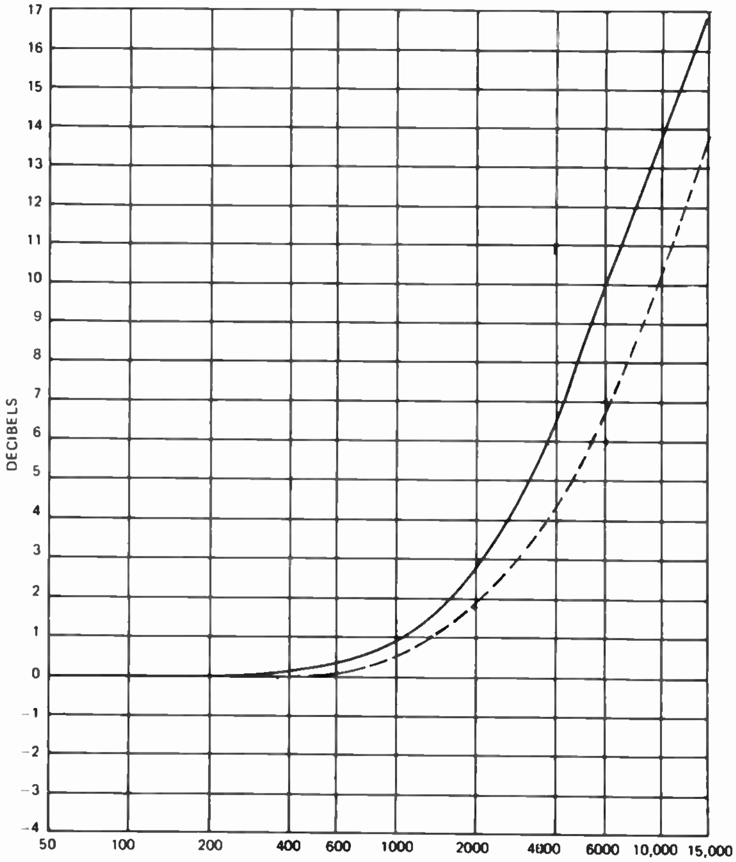


Fig. D-1:

4) Now measure the frequency response by measuring the oscillator output required to produce 3.3V rms (an internal level corresponding to 100% modulation) at the 8100A/1 output at 50, 100, 400, 1000, 5000, 10,000, and 15,000 Hz. If you use the AC voltmeter in the N&D Test Set to measure oscillator and/or 8100A/1 output levels, bear in mind that the 75us deemphasis network must not be used for these measurements.

5) Since the frequency response test requires you to readjust the oscillator output level at each frequency to produce 100% modulation at the 8100A/1 output, it is often convenient to measure the THD @ 100% modulation at the same time that you measure frequency response. If you are using the the N&D Test Set's AC voltmeter in the frequency response test, remember that you must connect the 75us deemphasis network to the N&D meter input for the THD measurements only.

6) Plot the results of the frequency response measurement on a standard 75us (or 50us) preemphasis graph like FCC #73.333 (USA). The points should be within ±0.75dB of the standard preemphasis.



STANDARD 75uS PREEMPHASIS CURVE (Solid Line)
STANDARD 50uS PREEMPHASIS CURVE (Dotted Line)

Fig. D-2: Preemphasis Graph

7) The deemphasized THD should not exceed 0.05% at any frequency. In many cases, results will be determined entirely by the quality of oscillator and distortion analyzer available, and/or by the presence of RF fields which might affect the instruments.

(A more accurate frequency response evaluation can be performed by sweeping the system with a test set like the Tektronix 5L4N Spectrum Analyzer/Tracking Generator. If the station has such equipment, see paragraph 6.c of **Appendix** E for further information.)

8) **Noise:** Short both 8100A/1 inputs, and measure the deemphasized noise at the 8100A/1 output through the 75us deemphasis network. It should not exceed -63dBm. (Note that hum or buzz due to test equipment grounding problems and/or high RF fields may result in falsely high readings. If the output of the N&D set is monitored with a scope, problems like this should be immediately apparent.)

c) **Operate-Mode Measurements:** These measurements evaluate certain static characteristics of the 8100A/1 in its normal OPERATE mode. Normal measurements given herein are provided for service guidance only, and are not guaranteed. As in the PROOF mode measurements above, these measurements should be repeated for both left and right channels.

1) Reconnect the audio oscillator to the 8100A/1 input. Switch both 8100A/1 PROOF/OPERATE switches to OPERATE. Be sure that operating controls are standardized as described in (a) above. Set the oscillator frequency to 1kHz, and adjust the oscillator output level to produce 10dB G/R as read on the MASTER G/R meter on the 8100A/1 front panel.

2) Verify that the 8100A/1 VU meter reads 0 VU ±0.5VU in the COMPRESSOR OUT position.

3) Measure the 8100A/1 output level and THD for each frequency indicated in the table below, and compare your results with the typical readings provided.

Table D-1

50Hz	2.05V rms; less than 0.1% THD
500Hz	2.14V rms; less than 0.1% THD
3kHz	2.50V rms; approx. 2.1% THD
12kHz	2.30V rms; less than 0.1% THD

The increase in THD at 3kHz is caused by a combination of factors, including the "standard" control settings (which are atypical of operating settings), and the fact that this distortion is the result of slight clipping permitted by the Smart Clipper circuitry. The distortion-cancelling clipper in the 8100A/1 exploits the fact that the ear is quite insensitive to very high frequency harmonic distortion (the major distortion component in the 3kHz test is third-harmonic at 9kHz), and is far more disturbed by low-frequency IM distortion. At 12kHz, all clipper-induced harmonic distortion is filtered out by the 15kHz lowpass filters.

Excessive THD at 50Hz not present in the PROOF-mode test is ordinarily caused by problems in the bass timing module on Card #5. Excessive THD at 500Hz is often caused by problems in the master timing module on Card #5. (See paragraph 2.d of **Appendix** B).

Atypical THD at 3kHz can be caused by several factors. Excessively high THD can be caused by a failure in the HF limiter on Card #6 (see paragraphs **4.a** and **4.b** in **Appendix B**). Excessively low THD can be caused by excessive action of the HF limiter, or by failure in the dynamic clipper-threshold adjustment circuitry (see paragraph **5.a** of **Appendix B**.)

4) Now turn the H-F LIMITING control to 10. Sweep the oscillator frequency up from 1kHz, and determine what frequency first turns the front-panel H-F LIMITING lamp ON. This frequency is typically 1.6kHz.

Steps (5) and (6) below provide a first-order test of the dynamics of the timing circuitry in the compressor. However, there are many possible faults which these tests will not detect. These must be diagnosed by more sophisticated tests at the factory.

5) Turn the BASS COUPLING control to "0". Set the oscillator frequency to 5kHz, and adjust the oscillator output level to produce 15dB G/R as indicated on the MASTER G/R meter. Switch the COMPRESSOR PROOF/OPERATE switch to PROOF and measure the time required for the MASTER G/R meter to fall from 15dB to 5dB indicated G/R. This time should be 6 seconds, ±1.5 seconds.

6) Change the oscillator frequency to 50Hz and turn the oscillator's output level control all the way down. Restore the COMPRESSOR PROOF/OPERATE switch to OPERATE, and advance the oscillator output level until the 8100A/1 BASS G/R meter reads 15dB G/R. Switch the COMPRESSOR PROOF/OPERATE switch to PROOF, and measure the time required for the BASS G/R meter to fall from 15dB to 5dB indicated G/R. This time should be 3 seconds, ±0.75 second.

This concludes tests of the audio processing circuitry.

STEREO GENERATOR

This test procedure requires only the use of an audio oscillator, the station's stereo monitor, and an oscilloscope. It has been our experience that the 8100A/1 stereo generator is far more stable than most monitors. Therefore, accurate measurement and adjustment of separation and pilot phase is best done with an oscilloscope. Crosstalk, 38kHz suppression, and pilot parameters cannot be conveniently measured on a scope, and must therefore be measured on the stereo monitor.

a) Stereo Performance Measurements

1) Connect the oscillator to the right-channel 8100A/1 audio input. Connect the stereo monitor to the 8100A/1 baseband (BNC) output.

2) Place both PROOF/OPERATE switches in PROOF. Place the rear-panel TEST/NORMAL switch in NORMAL. Turn the PILOT ON/OFF switch ON. Place the 8100A/1 CROSSTALK TEST switch in OPERATE. Connect an AC VTVM to the right channel TEST JACK on the rear panel. Set the oscillator frequency to 1kHz, and adjust the oscillator output level to produce 3.3V rms at the right TEST JACK.

3) Set the 8100A/1 OUTPUT ATTEN fully CW. Adjust the stereo monitor input sensitivity to produce an indication of 100% modulation on the right channel. (If this cannot be achieved, readjust the 8100A/1 OUTPUT ATTEN appropriately.)

4) **Crosstalk:** Use the stereo monitor to measure main-channel-to-subchannel and subchannel-to-main-channel crosstalk at 50, 400, and 15,000 Hz at 100% modulation. Use the 8100A/1 CROSSTALK TEST switch to produce the appropriate signals. (The CROSSTALK TEST switch takes the right channel input signal and uses this signal to create pure L+R [main-to-sub] or pure L-R [sub-to-main] modulation.) Note that because of preemphasis, substantial adjustment must be made in the oscillator output level to keep the modulation percentage constant as frequency is varied. Verify that crosstalk does not exceed -40dB at any frequency. (**NOTE:** Typically, you will be measuring the performance of the stereo monitor in this test. Only a baseband spectrum analyzer is sufficiently selective to accurately measure the 8100A/1 stereo generator's crosstalk performance.)

5) **38kHz Suppression:** Return the 8100A/1 CROSSTALK TEST switch to OPERATE. Modulate the right channel to 100% at 7.5kHz. Use the stereo monitor to verify that the 38kHz subcarrier suppression exceeds -40dB.

6) **Separation:** Connect the oscilloscope to the 8100A/1 baseband output. DO NOT USE AN ATTENUATOR PROBE; this may compromise the accuracy of the measurement. Trigger the scope externally from the oscillator. Turn the 8100A/1 PILOT switch OFF. Leave the oscillator connected to the right 8100A/1 audio input. Set the scope's vertical sensitivity to 0.5V/div, and DC-couple the input.

Set the oscillator frequency to 1kHz, and adjust the oscillator output level and the scope sweep rate to produce a scope pattern that looks like Fig. E-4 (in **Appendix** E). Separation is measured by determining the flatness of the baseline. If the 8100A/1 has been tweaked to compensate for a given exciter/RF amplifier/antenna system, then the baseline might not be quite flat. You must decide at this point whether to retain the current adjustment (for the sake of expediency) or whether to readjust the SEPARATION control (to determine the amount of separation which the system is capable of providing). If the baseline is almost flat, this implies that no fault has occurred in the stereo generator, and further tests are not required.

If you wish to measure separation, first expand the scope vertical scale to 50mV/div. Next adjust the 8100A/1 SEPARATION control to secure the flattest possible baseline. The separation is then calculated by the formula: S=20log(O/4), where S is the separation in dB, and D is the peak-to-peak deviation of the baseline from flatness (in volts).

Measure the left-into-right and right-into-left separation at 50Hz, 1kHz, 5kHz, and 15kHz, alternately driving the left and right channels and shorting the undriven channel.

The separation should be greater than 45dB, 50-15,000Hz, and is typically better than 60dB. (60dB is the practical limit of resolution for the oscilloscope separation measurement technique.)

7) **Pilot Phase:** Connect the oscillator to the 8100A/1 right audio input. Turn the pilot ON, and place the CROSSTALK TEST switch in SUB-TO-MAIN. Adjust the oscillator frequency to 1kHz (non-critical). You should see a scope pattern like Fig. E-5 (**Appendix E**). Expand the scope display to examine the zero-crossing region (Fig. E-6). The "tips" of the pilot should be exactly horizontal and level if the pilot phase is theoretically perfect. If the pilot phase has been tweaked to accommodate a particular exciter/RF amplifier/antenna system, then slight deviations from perfect flatness may be noted. Gross deviations imply a failure in the Phase-Locked Loop system (see paragraph **7.f** in **Appendix** B for troubleshooting hints).

Return the 8100A/1 CROSSTALK TEST switch to OPERATE.

b) Remote Control And Logic Verification

This procedure tests the operation of the rear-panel remote control terminals, and also verifies that the stereo generator switching logic is producing correct switching. To activate a remote control terminal, ground its "-" terminal to the "circuit ground" terminal on the rear-panel barrier strip, and momentarily jumper its "+" terminal to the "+22V" terminal on the barrier strip.

1) Connect the oscillator to the 8100A/1 right audio input. Switch the front-panel STEREO/MONO switch to STEREO. Verify that the front-panel STEREO light is lit, and that the stereo monitor indicates normal separation. Adjust the 8100A/1 PILOT INJECTION control as necessary to produce 9% pilot. Adjust the oscillator output level until the stereo monitor indicates 100% TOTAL MODULATION.

2) Switch the 8100A/1 into the MONO RIGHT mode by means of the rear-panel remote control terminal. Verify that the STEREO lamp on the front panel goes out, that the pilot disappears, and that the TOTAL MODULATION is 100% ±3%.

3) Disconnect the oscillator from the right channel input, and connect it to the left channel input. Verify that the TOTAL MODULATION reads 0%.

4) Switch the 8100A/1 into STEREO mode by means of the rear-panel remote control terminals. Verify that the front-panel STEREO lamp is on. Readjust the oscillator level as necessary to produce 100% TOTAL MODULATION.

5) Switch the 8100A/1 into MONO LEFT mode by means of the rear-panel remote control terminals. Verify that the STEREO lamp on the front panel goes out, and that the TOTAL MODULATION is 100% ±3%.

6) Disconnect the oscillator from the left channel input, and reconnect to the right channel input. Verify that the TOTAL MODULATION reads 0%.

c) Optional Performance Measurements Using Spectrum Analyzer

If a baseband spectrum analyzer like the Tektronix 5L4N is available, certain other stereo performance specifications can be readily verified.

Modulate the right channel to 100% at 5kHz (use the stereo monitor previously calibrated in step (a) to determine percent modulation). Observe the stereo generator output spectrum by bridging the spectrum analyzer input across the 8100A/1 baseband output. Expected spectrum components in the band from 50-53,000Hz are as follows:

1) 5kHz at 6.8dB below 100% modulation
2) 19kHz at 20.9dB below 100% modulation (for 9% injection)
3) 33kHz at 12.8dB below 100% modulation
4) 43kHz at 12.8dB below 100% modulation

All other components are spurious. Verify that the following conditions are met:

1) All spurious components between 50-53,000Hz should be better than 75dB below 100% modulation, with the exception of 38kHz. 38kHz should be better than 50dB below 100% modulation.

2) 76kHz should be better than 75dB below 100% modulation.

3) The sidebands of 76kHz (71 and 81kHz) should be better than 70dB below 100% modulation.

4) 114kHz and its sidebands should be better than 70dB below 100% modulation.

5) 152kHz and its sidebands should be better than 75dB below 100% modulation.

6) All other spurious should be better than 80dB below 100% modulation.

APPENDIX E:

Field Alignment Procedure and Specification

GENERAL

The following section describes how to align and calibrate OPTIMOD-FM Model 8100A/1 in the field. It is included primarily for purposes of reference, as routine alignment is neither necessary nor desirable due to the high stability of the circuitry.

WARNING!

THE AVERAGE RADIO STATION HAS NEITHER THE NECESSARY EXPERIENCE NOR THE REQUISITE TEST EQUIPMENT TO SUCCESSFULLY COMPLETE THIS PROCEDURE. IF CALIBRATION IS NECESSARY, WE STRONGLY RECOMMEND THAT THE CARD IN QUESTION BE RETURNED TO THE FACTORY FOR CALIBRATION BY OUR EXPERIENCED TECHNICIANS, WHO HAVE ACCESS TO SPECIAL TEST FIXTURES AND A SUPPLY OF EXACT-REPLACEMENT SPARE PARTS. ONLY IN AN EMERGENCY SITUATION SHOULD AN ATTEMPT BE MADE TO ALIGN AND CALIBRATE OPTIMOD-FM MODEL 8100A/1 IN THE FIELD.

The factory aligns each card independently to a standard, so that cards will be completely interchangeable. However, the user does not have access to the special test fixtures necessary to complete independent alignment of the cards. The user thus must use his own OPTIMOD-FM Model 8100A/1 chassis as a test fixture, and align the entire unit as a system.

This section is organized on a card-by-card basis. Cards should be calibrated in the same order as their order in the signal path, from input to output. This will occur naturally if the instructions in this section are followed in order from beginning to end. If a card later in the signal path is aligned while an earlier card is misaligned, the later card may not be correctly aligned, even if the instructions for that card are followed conscientiously.

Before commencing alignment, remove OPTIMOD-FM Model 8100A/1 from its normal rack mounting location and place it on the test bench away from RF fields. Jumper the chassis and circuit grounds together on the rear-panel barrier strip.

REQUIRED TEST EQUIPMENT

The following test equipment (or close equivalents) is required. It is assumed that the technician is thoroughly familiar with the operation of this equipment.

a) Digital Voltmeter, accurate to ±0.1%
b) Oscilloscope, dual-trace, triggered-sweep, with 5mHz or better vertical bandwidth
c) Ultra-Low Distortion Sinewave Oscillator/THD Test Set/AC VTVM (Sound Technology 1700B or 1710B)

d) Low Frequency Spectrum Analyzer with Tracking Generator (Tektronix 5L4N plug-in with 5111 Bistable Storage Mainframe)
e) A 137K 1% resistor
f) A 243K 1% resistor
g) Six 6" alligator-to-alligator jumper leads
h) A 1uF ±20% film capacitor (voltage unimportant)

REFER TO THE FOLD-OUT SCHEMATICS AND PARTS LOCATOR IN **APPENDIX J.**

CARD #1 (POWER SUPPLY)

a) Measure the voltage across C111 (or other convenient point on the +15 volt bus) with the DVM. Adjust R106 until the DVM reads +15.00 volts.

b) Measure the voltage across C112 (or other convenient point on the -15 volt bus). Make sure that the voltage is between -14.85 and -15.15 volts. If it is not, refer to **Appendix B (CIRCUIT DESCRIPTION),** paragraph 8.c for troubleshooting hints.

BEFORE ALIGNING EACH CARD AS DESCRIBED IN THE INSTRUCTIONS BELOW, REMOVE THE CARD OF INTEREST FROM ITS SLOT AND PLUG THE EXTENDER INTO THE EMPTY CARD SLOT. PLUG THE CARD INTO THE CARD EXTENDER. THIS WILL GIVE YOU ACCESS TO THE ALIGNMENT TRIMMERS AND TEST POINTS.

CARDS #3 and #4 (INPUT BUFFER/ INPUT CONDITIONING FILTER/ COMPRESSOR AUDIO PATH)

a) Remove Cards #3, #4 and #5 from their slots. Plug the extender board into the Card #3 slot. Cards #3 and #4 will both be aligned on this extender, one at a time, without moving it from the Card #3 slot.

b) Plug Card #3 into the extender board.

c) Connect one side of a 137K 1% resistor, and one side of a 243K 1% resistor to a convenient ground point (like the chassis) by means of jumper leads. Using two more jumper leads, connect the other side of the 137K resistor to the side of R333 away from IC305, and connect the other side of the 243K resistor to the side of R348 away from IC309. These external resistors now force reference gain-control currents into IC305A and IC309A respectively: 97uA into the "Master" VCA and 55uA into the "Bass" VCA.

d) Connect the chassis ground of the oscillator to the chassis of the 8100A/1. Connect the low side of the oscillator output to the chassis of the 8100A/1. Using a pair of jumper leads, connect the high side of the oscillator output to both the "+" input of IC301A and the "+" input of IC301B. This provides common-mode excitation for the input differential amplifier.

e) Set the oscillator frequency to 60Hz, and the oscillator output to 0 dBm. Observe TP1 (pin D at the card connector) with the AC VTVM adjusted so that the common-mode feedthrough is readily observed. Adjust R316 (CMRR) to null it. The nulled level of the 60Hz should be less than -60dBm.

f) Connect the low side of the oscillator output to the "-" terminal of the left-channel audio input of the 8100A/1. Connect the high side of the oscillator output to the "+" terminal. Set the oscillator output frequency to 5kHz and the oscillator output level to produce -15dBm at the output of IC302B (pin V of the card connector).

g) Observe the output of IC307A (pin K of the card connector) with the AC VTVM, and adjust R376 (MASTER VCA GAIN) to produce +2.0dBm at this point.

h) Readjust the oscillator frequency to 35Hz. If necessary, readjust the output level of the oscillator until it is identical to the oscillator output level produced at 5kHz.

i) Adjust R377 (BASS VCA GAIN) until the AC VTVM indicates +2.0dBm. The gains of both "Master" and "Bass" VCA's are now standardized, assuring card interchangeability.

NOTE: The following distortion and balance adjustments are made without disturbing the resistors jumpered into place in the steps above.

j) Without disturbing the oscillator output level, set its frequency to 5kHz. Switch the AC VTVM into its distortion-measuring mode, and measure the THD. Adjust R336 (MASTER DIST NULL) to null the THD. It should not exceed 0.04% if a noise-limiting 80kHz lowpass filter is employed in the measurement.

CAUTION!

> Any stray audio picked up on the leads of the 137K jumper resistor will cross-multiply with the desired signal in the VCA, and will produce second-harmonic distortion which cannot be nulled with the MASTER DIST NULL control. It may be necessary to bypass the R333 side of the 137K resistor to ground with a tantalum capacitor larger than 5uF and 15VDC. Ground the "+" terminal of the capacitor.

E

k) Set the oscillator frequency to 50Hz. Measure the THD as above, and adjust R351 (BASS DIST NULL) to null it. It should not exceed 0.04%.

l) Remove the oscillator from the 8100A/1 input. Ground the low side of the oscillator output to the 8100A/1 chassis and, using a pair of jumper leads, connect the high side of the oscillator output through a 1uF film capacitor to the side of R333 away from IC305. (The 137K resistor is already connected to this point. Don't disturb it.) Set the output frequency of the oscillator to 100Hz, and its level to produce approximately 0.55V rms at its output. Observe the output of IC307A with the AC VTVM at high gain. You will see a distorted feedthrough component from the oscillator. Adjust R331 (MASTER VCA BALANCE) to null the feedthrough.

m) Move the lead from the 1uF capacitor from R333 to the corresponding side of R348. Do not disturb the resistor already connected to this point. Set the oscillator output to approximately 0.35V rms. Continue to observe the output of IC307A, and adjust R346 (BASS VCA BALANCE) to null the feedthrough component observed.

n) Remove all jumper leads connected to Card #3, and remove Card #3 from the extender.

o) Insert Card #4 in the extender (still in slot #3), and repeat steps **c - n.**

CARD #5 (COMPRESSOR CONTROL/ GATING DETECTOR)

IMPORTANT

Before embarking on this procedure, be sure that Cards #3 and #4 have been standardized according to the alignment procedure above, or are in their original factory-aligned condition.

a) Connect oscillator to the 8100A/1's left input, high side to "+", low side to "-".

b) Pull Card #3 halfway out and connect the AC VTVM to TP1. TP1 may be readily accessed at the end of R323 closest to the edge of the board.

Reinsert Card #3. Make sure that Card #4 is also in its slot. (If only one card is inserted, all gain control current will be diverted to the VCA's in that card, reducing the gain 6dB below its correct value.)

c) Extend Card #5 on the extender.

d) Set the oscillator frequency to 5kHz; set its output level to produce -15dBm at Card #3 TP1.

e) Switch the Compressor PROOF/OPERATE switch (on Card #5) to PROOF and allow the gain to settle for at least one minute.

f) While you are waiting, reconnect the AC VTVM to the output of IC307A on Card #3. This point can be readily accessed at the upper end of R364 by pulling the card halfway out, and reinserting it as above.

g) Adjust R501 (MASTER GAIN CAL) on Card #5 until +2.0dBm is observed on the AC VTVM.

h) Set the oscillator frequency to 35Hz. Be sure that the oscillator output level is the same as it was @5kHz.

i) Adjust R514 (BASS GAIN CAL) on Card #5 until the AC VTVM reads +2.0dBm.

j) Remove Card #5 from the extender, and restore Card #5 to its slot.

CARD #6 (PREEMPHASIS/ HF LIMITER)

This card serves both left and right channels. The procedure below is performed twice; once for the left channel and once for the right. When the reference designator of an alignment trimmer is specified, the reference designators for both left and right channel trimmers will be given in order, with the right in parentheses.

a) Extend Card #6. Place both PROOF/OPERATE switches in PROOF. Turn R626 and R660 (FET BIAS) fully clockwise to guarantee that the FET's in IC603 will be fully pinched-off.

b) Connect the output of the tracking generator in the 5L4N spectrum analyzer to the left (right) audio input of the 8100A/1.

c) The PREEMPHASIS trimmers are used to adjust the entire 8100A/1 for best conformance to the standard FM preemphasis. Place the rear-panel NORMAL/TEST slide switch in NORMAL. This connects the output of the audio processing to the rear-panel TEST JACKS. Now connect the TEST JACK corresponding to the channel which you are aligning to a precision 75us (or 50us) deemphasis network. Connect the output of the 75us deemphasis network to the input of the 5L4N spectrum analyzer.

(To make a 75us network, connect a 7.50K 1% metalfilm resistor between the input and output of the network. Then connect a 0.01mfd 1% capacitor between the output of the network and ground. This network should be loaded by no less than 1 megohm, or its accuracy will be degraded. To make a 50us network, use a 4.99K 1% metalfilm resistor instead of a 7.50K.)

d) Set the 5L4N for a 0-20kHz sweep (2kHz/div). Set its input sensitivity to -10dBV in dB mode. Set the vertical sensitivity to 2dB/division, and set the output level of the tracking generator to obtain an on-screen trace. (You may have to readjust the 8100A/1 INPUT ATTEN if gain is insufficient.)

e) You are now sweeping the entire 8100A/1 system, including the complex filters in Cards #8 and #9. Adjust R618(652) to achieve maximally flat response, similar to Fig. E-1. The response should be ±0.75dB or better, 50-15,000Hz.

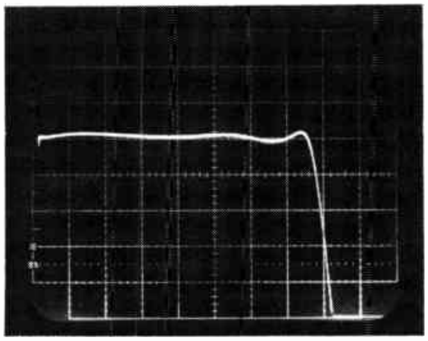


Fig. E-1: Overall Deemphasized System Response

f) Now turn R626(660) (FET BIAS) slowly counterclockwise until the swept response just begins to roll off. Back off until no rolloff is observed, and go a little further for safety.

g) Repeat steps (d) through (f) for the right channel.

h) Connect the 8100A/1 input to the oscillator.

i) Observe the output of IC605A(611A) with the distortion meter. Set the oscillator frequency to 10kHz, and the oscillator level to produce +10dBm at the output of IC605A(611A).

j) Turn R678 (HF LIMIT THRESH--front-panel control) fully CW. Turn the LIMITER PROOF/OPERATE switch (on Card #6) to OPERATE. The 10kHz level should go down to approximately +6dBm. Now adjust R631(665) (DIST NULL) to minimize THD. Bear in mind that you are observing a preemphasized signal, and that THD will be even lower after deemphasis. Even without deemphasis, THD is typically less than 0.1%.

k) Repeat steps (i) and (j) for the right channel.

l) Observe the junction of R669 and R670 with a high-impedance (10 megohm or greater) DC DVM. Adjust R671 (OVERSHOOT COMPENSATION THRESHOLD) until the DVM reads +4.50VDC.

m) Return Card #6 to its slot.

CARD #8 and #9 (FM SMART CLIPPER/ FCS OVERSHOOT COMPENSATOR)

This procedure is performed twice -- once for Card #8, and once for Card #9. Only Card #8 will be referenced.

a) Extend Card #8.

b) Connect the oscillator to the left 8100A/1 input. Place both PROOF/OPERATE switches in PROOF. Connect the AC VTVM to the input of Card #8 (input side of R801). Set the oscillator frequency to 100Hz, and set the oscillator output level until -4.3dBm is observed on the VTVM.

c) Observe the left rear-panel TEST JACK with the AC VTVM with the rear-panel NORMAL/TEST switch in NORMAL. Adjust R841 (SAFETY CLIPPER THRESH) to produce 0dBm at the TEST JACK. This sets a standard gain of +4.3dB through the card.

d) **OPTIONAL PERFORMANCE VERIFICATION** Of Filters In FM Smart Clipper

1) Connect the output of the 5L4N tracking generator to TP1 of Card #8 (pin L on the card connector). Connect the input of the 5L4N to the left rear-panel TEST JACK. Observe the swept response with the 5L4N vertical span at 10dB/div, with 20-20kHz log frequency sweep. The swept response shows the response of the Smart Clipper to clipping-induced distortion components only. Note the high amount of rejection below 2.2kHz, and the very steep slope at 2.2kHz (see Fig. E-2).

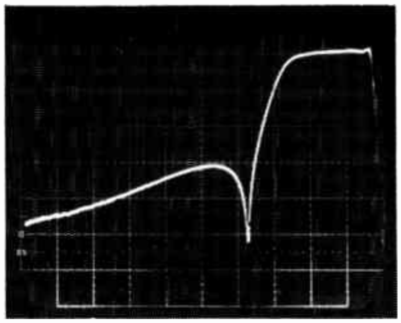


Fig. E-2: Distortion Cancellation Response

If this swept response does not resemble Fig. E-2, then there is a fault in either the filters or phase correctors between the card input and the output of IC803B. This test is both fast and sensitive because accurate cancellation demands accurate matching of the phase and amplitude responses of both the phase-corrected 15kHz lowpass filter and the distortion-cancelling filter. If any circuitry is faulty, then the cancellation will not occur accurately.

2) Measure the clipper bias voltages at the outputs of IC808A and IC808B. These should be approximately ±1.5VDC with no signal. (**NOTE:** The temperature-compensation circuitry will cause this bias voltage to change slightly with temperature to keep the clipping threshold constant.)

e) **OPTIONAL PERFORMANCE VERIFICATION** Of FCS Overshoot Corrector

1) Connect the oscillator to the junction of R806 and L801 (this provides a convenient injection point that bypasses the first clipper).

2) Place the LIMITER PROOF/OPERATE switch (on Card #6) in OPERATE.

3) Observe the left rear-panel TEST JACK with a scope. Set the oscillator frequency to 100Hz, and advance the oscillator output level until clipping just barely occurs. Measure the oscillator output, and verify that it is approximately 0.63V rms. (The "clipping" is the action of the overshoot corrector. If this clipping doesn't occur, then there is a fault in the overshoot corrector sidechain.)

4) Increase the oscillator output 4dB. Substantial clipping should occur. Now sweep the oscillator frequency upward, and verify that the peak level of the output waveform never exceeds the "flat-top" level of the 100Hz clipped sinewave by more than 0.7dB, and that this 0.7dB peak occurs at approximately 4.4kHz. At this frequency, the waveform should resemble a filtered square wave with two equal cycles of ringing on the top and bottom of the wave (Fig. E-3). If substantially more than 0.7dB overshoot occurs, particularly if the ringing is not symmetrical, then suspect problems in the filters or phase-shift networks associated with the FCS circuit.

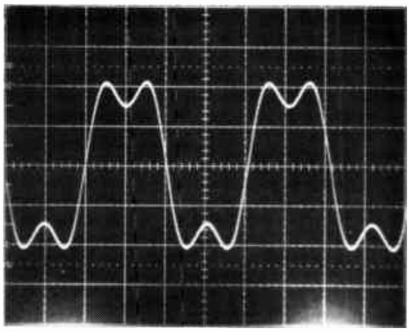


Fig. E-3: 5kHz Overdriven FCS Output

CARD #7 (STEREO GENERATOR)

The stereo generator circuitry is essentially independent of the preceding audio processing circuitry. This alignment procedure can therefore be performed regardless of whether the audio processing circuitry has been correctly aligned.

When the rear-panel NORMAL/TEST switch is in TEST, the two TEST JACKS on the rear panel can serve as audio inputs to the stereo generator. In order to assure correct operation of the stereo generator, these jacks must be driven by a source impedance of less than 10 ohms. This means that a standard 600-ohm audio oscillator cannot be employed without a buffer amplifier. Although such an amplifier can be easily built with a few IC opamps, it is more convenient in the field to use the final buffer amplifiers on the #8 and #9 Cards to provide the correct driving impedance, and to leave the NORMAL/TEST switch in NORMAL. You must also place the LIMITER PROOF/OPERATE switch (on Card #6) in PROOF.

The oscillator is readily interfaced to the cards by clipping an alligator-to-alligator jumper lead to the CR805, CR806 side of R840 (on Card #8), and doing the same to Card #9. These two jumper leads then serve as audio inputs to the stereo generator. Ordinarily, the connection will be made by temporarily removing the cards from their slots, clipping the jumpers on, and then replacing the cards in their slots. This way, they will not interfere with the extended #7 Card.

IMPORTANT

If the test procedure requires that only one channel be driven, ground the other jumper lead to avoid picking up crosstalk which could affect stereo generator adjustments.

WARNING!

IF THE ALIGNMENT CONTROLS ARE ADJUSTED IN THE ORDER SPECIFIED, NO INTERACTION BETWEEN ADJUSTMENTS WILL OCCUR. HOWEVER, IF ADJUSTMENTS ARE MADE IN RANDOM SEQUENCE, SEVERE INTERACTION CAN OCCUR, AND EARLIER ADJUSTMENTS MAY BE DESTROYED! TO AVOID TROUBLE, GO "BY THE BOOK".

a) Extend Card #7, and switch the pilot ON. Connect the input of the 5L4N spectrum analyzer to the 8100A/1 baseband output. Set the 5L4N input attenuator to -10dBV, and display in the 10dB/div log mode. The frequency sweep should be 0-100kHz linear. Set the 8100A/1 OUTPUT ATTEN until the 19kHz pilot reads -30dBV (two divisions from the top of the screen). Then switch the pilot OFF. (100% modulation is now indicated by signals reaching the top of the screen.)

b) **Modulator Distortion Null:** Place the stereo generator CROSSTALK TEST switch in OPERATE. Connect the oscillator to the right channel input (jumper lead from Card #9). Set the oscillator frequency to 5kHz, and advance the oscillator output level until the 38kHz sidebands read 12dB below the top of the screen. You will see several spurious components, including a second-harmonic distortion component at 10kHz, and subchannel second-harmonic components at 28 and 48kHz. Adjust R720 (DIST NULL) to minimize the amplitude of these components. It is ordinarily possible to get them below the bottom of the display (better than 80dB below 100% modulation).

c) **Low Frequency Sub-to-Main Crosstalk Null:** Place the CROSSTALK TEST switch in SUB-TO-MAIN. The 5kHz component you see on the spectrum analyzer is sub-to-main crosstalk. Adjust R717 (SUB:MAIN XTALK) to null the 5kHz as much as possible.

d) **15kHz Sub-to-Main Crosstalk Null:** Change the oscillator frequency to 15kHz. The crosstalk will ordinarily increase. Additional 15kHz crosstalk is nulled with a small piston trimmer capacitor across R712.

e) **76 kHz Null:** Return the CROSSTALK TEST switch to OPERATE. Set the oscillator frequency to 5kHz. Adjust R725 (76 KHZ NULL) to null the 76kHz component. Usually it is possible to get it below the bottom of the screen (better than -80dB below 100% modulation).

f) **76kHz Sideband Specification Verification:** Observe the sidebands ±5kHz about 76kHz, and verify that they are better than 70dB down (-75dB typical). If the sidebands do not meet specification, refer paragraph **7h** in **Appendix** B for troubleshooting hints.

g) **38kHz Null:** Adjust R714 (38 KHZ NULL) to null the 38kHz subcarrier. It is typically possible to achieve better than -70dB short-term suppression, and -60dB long-term suppression.

NOTE

Putting the CROSSTALK TEST switch in either TEST mode will compromise the 38kHz suppression somewhat (although it will never deteriorate above -40dB). Do not be disturbed if you observe this; in this alignment step, you have just nulled the 38kHz in normal operating mode as is correct.

h) **Main-to-Sub Crosstalk Null:** Place the CROSSTALK TEST switch in MAIN-TO-SUB. Adjust R709 (MAIN:SUB XTALK) to null the sidebands observed ±5kHz from 38kHz. Long-term suppression achievable exceeds 70dB.

i) **DC Offset Null:** Connect a DVM to the baseband output of 8100A/1, and observe the DC voltage. (You could also use a DC-coupled scope.) Temporarily suppress the audio oscillator output (or disconnect the oscillator and ground the jumper leads going to Cards #8 and #9.) Switch the CROSSTALK TEST switch to OPERATE, and be sure that the stereo generator is in STEREO mode (i.e., that the front-panel STEREO lamp is lit.) Now adjust R728 (DC OFFSET NULL) until the DC output voltage is 0V.

NOTE

The next part of the Alignment Procedure contains instructions for adjusting the User Controls for an "ideal" stereo output. At the time that the 8100A/1 is installed in a real system, these controls often are slightly readjusted from their "ideal" setting to compensate for variations in exciters, RF amplifiers, and antennas.

j) **Separation:** Connect the audio oscillator to the #8 Card jumper lead, and ground the #9 Card jumper lead as before. Connect the scope to the 8100A/1 baseband output through a wire or coax. DO NOT USE AN ATTENUATOR PROBE; these probes often have midband phase shift (due to slight imperfections in their frequency compensation circuitry) which will compromise the accuracy of the separation adjustment.

Trigger tne scope externally from the oscillator. Adjust the scope sensitivity to 0.5V/div, and input coupling to "DC". Adjust the 8100A/1 OUTPUT ATTEN control for maximum output (fully CW). Set the oscillator frequency to 1kHz, and adjust the oscillator output level until the baseband output is 4V p-p.

Now adjust R772 (SEPARATION) to achieve the flattest possible baseline. Fig. E-4 shows the correct adjustment. To make the final adjustment accurately, expand the vertical scale by a factor of ten by changing the scope vertical sensitivity to 50mV/div.

To approximately measure the separation, use the formula: S=20log(D/4), where S is the separation in dB, and D is the peak-to-peak deviation of the baseline from perfect flatness, in volts. Ordinarily, the separation from 50-15,000 Hz will be essentially unmeasureable (better than -60dB).

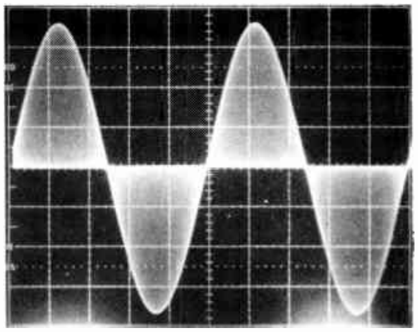


Fig. E-4: Separation

It is desirable to measure both right-into-left and left-into-right separation. If the earlier CROSSTALK adjustments were correctly performed, left-into-right and right-into-left separation should both null at the same setting of R772.

k) **Pilot Phase:** Connect the audio oscillator to the jumper lead going to Card #9, and ground the jumper lead going to Card #8. Place the CROSSTALK TEST switch in SUB-TO-MAIN. Adjust the oscillator frequency to a convenient frequency (non-critical) around 1kHz. Turn the pilot ON.

You will see a scope display similar to Fig. E-5. Expand the scope display by increasing the vertical sensitivity until it looks like Fig. E-6. Now adjust R769 (PILOT PHASE) until the "tips" of the pilot are exactly horizontal and level.

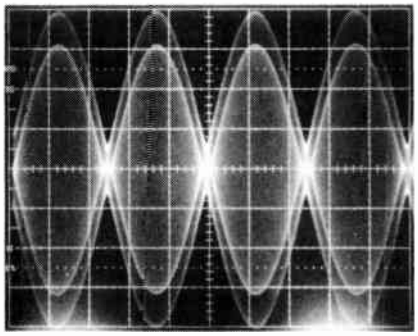


Fig. E-5: Pilot Phase

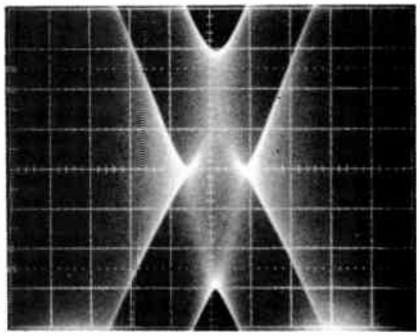


Fig. E-6: Pilot Phase, 10x

l) **Final 38kHz Suppression Check:** Remove the two jumper leads from Cards #8 and #9. Grounding and/or connecting the oscillator to these cards may have slightly affected the DC offset at the points to which the jumper leads were connected. Since the stereo generator is DC-coupled after these points, differential DC offset here translates as a DC-coupled L-R component (i.e., a constant 38kHz component). Now that offset conditions are entirely normal, turn the CROSSTALK TEST switch to OPERATE and adjust R714 (38KHZ NULL) if necessary to renull the 38kHz subcarrier.

m) Remove Card #7 from the extender, and reinstall Card #7 in its slot.

This concludes the Field Alignment Procedure for the entire 8100A/1 system. Insert the extender board in its slot, and replace the subpanel, being sure that all four Dzus fasteners are fully tightened for RF suppression. Close the front door, and fasten with its three screws. Remove the jumper between chassis and circuit grounds, unless it is ordinarily used in your installation.

OPTIMOD-FM Model 8100A/1 is now ready for reinstallation in the station.

APPENDIX F:

Trouble Diagnosis and Correction

This Appendix is the first place you should go to obtain information on what to do if OPTIMOD-FM develops a fault. Many problems experienced in the field can be resolved or conclusively diagnosed with the following diagnostic routines. Even if the repair cannot be done in the field, the information provided by these diagnostic routines can speed the work of the factory service department in making the repair. Please perform these routines and make notes if you observe anything exceptional or unusual.

1) Use systematic troubleshooting techniques to positively determine that the problem is in fact being caused by OPTIMOD-FM, and not by other equipment. If a standby processor/stereo generator chain is available, it should be substituted for the supposedly faulty OPTIMOD-FM to see if the problem vanishes. If a standby processor/stereo generator is not available, audio quality at the OPTIMOD-FM audio input terminals should be checked with a high-quality monitor system. Note that even slight distortion can be seriously exaggerated by "heavy" processing, and that this sort of processing can only be successful if the input audio is extremely clean. A relatively minor problem which develops in the station's audio chain or STL can therefore be magnified by the action of OPTIMOD-FM, even if OPTIMOD-FM is in no way defective.

If the audio is clean going into OPTIMOD-FM, problems can still arise in the exciter. If a standby exciter is available, it should be substituted to see if the problem vanishes. If no standby exciter is available, you can connect the baseband output of the 8100A/1 stereo generator directly into the baseband input of your stereo monitor (bypassing the FM monitor) to see you still observe the problem on the monitor. (Be sure that the problem is not in the stereo monitor by verifying that the problem can be observed on more than one receiver.)

If the problem vanishes when you observe the stereo monitor, the exciter (or composite STL) is strongly suspect. An exciter, for example, may appear to work in mono mode (with OPTIMOD-FM bypassed), yet exhibit noise and/or distortion when asked to pass stereo signals. Some exciters have well-known characteristic problems!

Changes in or deterioration of grounding and/or exterior lead dress can sometimes cause RFI or hum problems to appear in a good OPTIMOD-FM.

If it seems impossible to conclusively isolate the problem to OPTIMOD-FM, yet no other definite cause is found, then performing the Field **Audit-Of-Performance** procedure in **Appendix** D may help diagnose a problem.

2) If the fault has been positively isolated to OPTIMOD-FM, the **Problem Localization Routine** described below should be performed to identify the faulty PC card.

PROBLEM LOCALIZATION ROUTINE

General Principles: The most powerful and general technique for localizing a problem within OPTIMOD-FM is signal tracing. This simply means that the signal is observed at various points as it passes from OPTIMOD-FM's input to its output. If the signal is normal at some point "A" in the circuit, and is abnormal at a point "B" further towards the output, then the problem clearly lies in circuitry between points "A" and "B".

Signal tracing in OPTIMOD-FM is facilitated by the fact that much of the circuitry is duplicated for stereo, and is arranged so that the bad channel can be readily compared with the good one, which serves as a "normal" reference.

Power Supply Tests: Some circuitry is common to both channels, and failures will therefore affect both channels in a symmetrical way. In particular, problems in the power supply may affect many OPTIMOD-FM circuits simultaneously. For this reason, the first step in any troubleshooting procedure is to check the power supply for normal output. Gross changes in power supply voltage can be detected with the "+15VDC" and "-15VDC" positions on the VU meter. Normal readings are 0VU ±0.5VU. If normal readings are obtained, skip to the next section on **VU Meter Technique.**

If either "+" or "-" power supply output is significantly low, it could indicate a defect in the supply itself. But it is more likely to indicate a shorted IC or capacitor somewhere in the circuit that is overloading the supply and causing it to current-limit.

The power supply is electronically protected against excessive current demand by other parts of the circuitry. If a failure causes a high current demand on the power supply, its output voltage will drop as far as necessary to reduce output current to approximately 0.75A. If the power supply voltage is observed to be abnormally low, unplug each circuit card in turn and check if the power supply recovers by observing the "-15VDC" meter position. (The negative regulator tracks the +15V supply. So the -15V supply will go down if the +15V supply does, even if the -15V supply or load is completely normal. A normal "-15VDC" reading thus assures a normal "+15VDC" reading.) If recovery occurs, then troubleshoot the unplugged board. Ordinarily, the defective component will become very hot, and is easily detected by touch. (Wet your finger first to avoid burns!) If all cards are removed and an undervoltage problem does not disappear, examine the meter card, motherboard, and chassis wiring before suspecting the supply itself. (A wiring problem will be indicated by an ohmmeter's indicating very low resistance between the "+15V" or "-15V" power busses with AC power OFF.)

Even if power supply voltages appear normal on the VU meter, subtle problems such as hum, noise, or oscillation may still exist with the supply. To check for this, test the regulated DC with a well-calibrated DVM, scope, and AC VTVM with 20-20k Hz bandpass filter. Voltages should be +15.00V ±0.075V, -15.00V ±0.375V. Ripple must be less than 2mV rms, 20-20,000Hz. There must be no high frequency oscillation.

VU Meter Technique: If one channel goes dead, the VU meter provides a means for fast signal tracing. Note, however, that problems other than gross gain changes or total failure to pass signal may not be detected by the meter alone.

First, switch through the first six VU meter functions (which monitor the audio processing) to see where the signal disappears (or the VU meter pegs, implying that a defective IC opamp has latched up to the power supply rail.) Refer to the block diagram (p. J-21) to locate the exact points in the signal path monitored by the meter.

If the signal is normal at the input terminals and abnormal in either INPUT BUFFER position, then the problem lies with Card #3 (left channel) or Card #4 (right channel), or with the incoming audio circuitry prior to these cards.

If the signal is normal at the INPUT BUFFER positions but abnormal in the COMPRESSOR OUT position, then the problem probably lies with Card #3 (left channel), with Card #4 (right channel), or, if both channels are equally affected, with Card #5.

If the signal is normal at the COMPRESSOR OUT positions, but abnormal in either FILTER OUT position, then the problem may lie with Card #6 (which contains both channels), Card #8 (left channel), or Card #9 (right channel). The **Card Swap Technique** (below) must be used to localize the problem more precisely.

Abnormal readings in the three Stereo Generator positions on the VU meter switch (19KHZ OSC; 38KHZ AGC, 38KHZ PLL) are almost always due to problems with Card #7, or with the power supply. (L-R can read abnormally if the rear-panel NORMAL/TEST switch is left in TEST, and no inputs are provided to the rear-panel TEST jacks.)

The instructions below provide more detailed information on troubleshooting at the "card exchange" level. Servicing on the "component replacement" level requires more profound understanding of OPTIMOD-FM circuit operation, which is provided by **Appendix A (SYSTEM DESCRIPTION)** and **Appendix B (CIRCUIT DESCRIPTION).** If the technician wishes to troubleshoot OPTIMOD-FM 8100A/1 at the component level, he should first use **Appendix A** to help track down the fault to a given subsystem, and then refer to **Appendix B** for an extremely detailed explanation of the circuitry at the component level.

Card Swap Technique: If the defective card has not yet been conclusively identified and if the fault appears on one channel only, the next step involves a card swap technique. The PC cards in OPTIMOD-FM Model 8100A/1 have been specifically configured to aid troubleshooting if a fault appears in one stereo channel only. Cards #3 and #4 are identical, as are Card #8 and #9. Therefore, these card pairs can be interchanged one pair at a time to see if the problem moves from one channel to the other (implying that the fault is with one of the cards just moved), or stays the same (implying that the problem lies elsewhere in the system).

If interchanging these card pairs fails to affect the location of the problem, then Card #6 should be investigated. This card passes both left and right audio. To aid troubleshooting, a jumper is provided at the output of the card to interchange the outputs of the left and right channels (See Fig. F-1). If this jumper is moved and the fault moves from one channel to the other, then Card #6 is probably faulty.

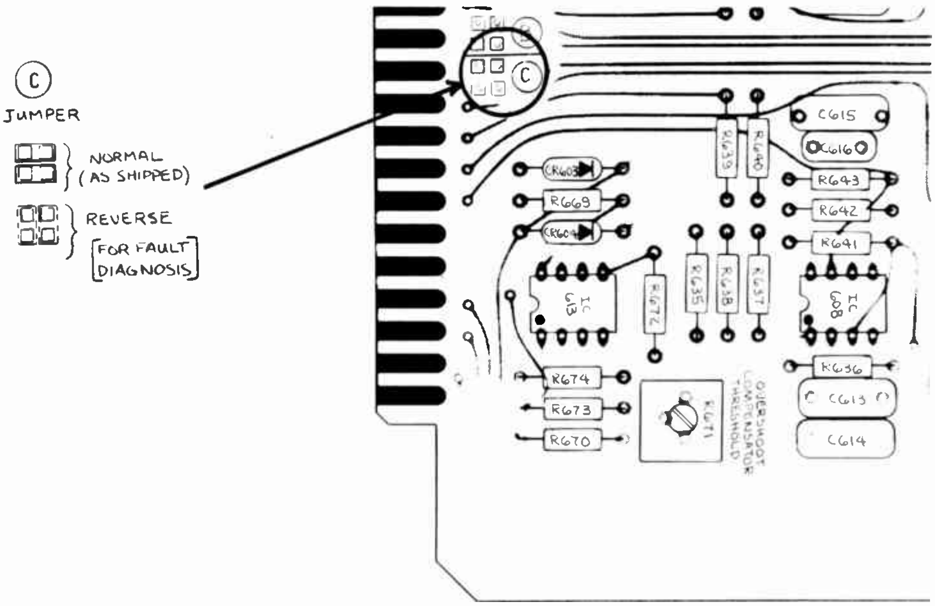


Fig. F-1: Card #6 Left and Right Jumpers

Cards Common To Both Channels: Cards #1 (Power Supply), #5 (Compressor Control Circuitry), and #7 (Stereo Generator) are common to both channels. Card #6 contains the common ±4.2V clipper bias supply used by Cards #8 and #9.

Diagnosis of power supply problems was discussed above.

A failure in Card #5 (the common processing control card that controls both Card #3 and #4) can manifest itself on both channels as distortion (too little gain reduction), low loudness (too much gain reduction), pumping or other dynamic problems (failure in the timing circuitry), or failure of the gating circuitry (which is usually indicated by abnormal behavior of the front-panel GATE lamp). First-order problems in card #5 are often indicated by a failure to produce the "standard level" under "standard control setup" conditions. (See c.1 and **c.2** in **Appendix** D for instructions on how to make this test.)

Problems in Card #7 (Stereo Generator) can be isolated by use of the rear-panel TEST jacks. When the rear-panel TEST/NORMAL switch is in NORMAL, these jacks carry the output of the audio processing. If on-air problems are observed (and you have determined that they are not due to the exciter, monitor, or other external causes), yet audio from these jacks (listened to through standard deemphasis) sounds normal, suspect the stereo generator (or possibly the power supply.)

FAILURES WHICH CANNOT BE DIAGNOSED BY CARD-SWAPPING

Phase Corrector Failures: One possible problem which is difficult to diagnose by means of a card swap is failure of a phase corrector on Cards #6, #8, or #9. Some failures can grossly change the phase response of a given channel without significantly affecting the frequency response. While each channel sounds normal by itself, the mono sum will exhibit gross frequency response aberrations due to phase cancellations. If the 8100A/1 is driven by mono material, the "L-R" meter position will fail to null.

The principal difficulty is determining which channel is abnormal, since phase corrector failures will cause audible problems (most often increased distortion) only with certain types of program material. The following describes listening tests to detect phase corrector failures. If the ear can detect the usually subtle effect of the corrector failure by listening to one channel only, then the card-swap technique can be successfully applied to isolate the problem. In these tests, it is important to drive both channels with identical program material, as the usual differences between the left and right channels can totally mask any differences due to phase corrector failure. The easiest way to assure identical L and R drive is simply to drive both L and R inputs in parallel from a single signal.

A phase corrector failure on Card #6 will cause slightly more high frequency clipping than would otherwise be expected, so the failed channel may sound slightly grittier when program material containing large amounts of high frequency energy is processed.

A phase corrector failure in the Smart Clipper (first part of Cards #8 and #9) will cause the distortion cancellation function to work incorrectly, and will result in sibilance distortion (splattered "ess" sounds on voices).

A phase corrector failure in the FCS Overshoot Compensator (second part of Cards #8 and #9) will result in inaccurate overshoot cancellation. This will result in overdriving the safety clipper when significant high frequency energy is present, which will in turn cause out-of-band frequency components to be generated. These components will cause aliasing distortion when decoded in a stereo receiver.

To test for this, drive one channel at a time with bright program material. Check separation by listening to the undriven channel as decoded by your stereo monitor. If one channel causes notably more "garbage" in the other (undriven) channel, then the channel causing the high amounts of "garbage" is suspect.

CATALOG OF TYPICAL SYMPTOMS AND PROBABLE CAUSES

This troubleshooting guide is a catalog of some possible failure modes in the 8100A/1. It should be used in conjunction with **Appendices A** and B to aid troubleshooting at the component level.

ALWAYS BE SURE THAT THE PROBLEM IS NOT IN THE SOURCE MATERIAL FEEDING OPTIMOD-FM.

Whistle is heard on air, perhaps only in stereo reception.

1. Power supply oscillation. Suspect C111, C112, IC101, IC102.
2. IC711 oscillating due to value shift in passive component. 38kHz no longer phase-locked to 19kHz. 38KHZ PLL and 38KHZ AGC meter positions will read abnormally.
3. Whistle on one stereo channel only probably due to oscillating IC. Use signal tracing techniques to isolate defective IC.

Buzz or hum.

1. Improper grounding. Chassis not properly grounded to rack. Circuit and chassis grounds connected through excessively long path. No direct connection between 8100A/1 circuit ground and circuit ground of exciter with balanced input.
2. RFI. Improve grounding scheme. Relocate 8100A/1 chassis. Change length of baseband output coax to retune it.
3. Low line voltage causing regulator to drop out and pass ripple.
4. C101, C102 in unregulated power supply failed, resulting in extremely high ripple. Power supply regulator drops out on each ripple cycle which instantaneously goes lower than 17.5 volts.

Loss of modulation control.

1. Make sure LIMITER PROOF/OPERATE switch (on Card #6) is in OPERATE.
2. Check for tightly-controlled peak levels at rear-panel TEST jacks. If levels not well-controlled, check ±4.2V supply on Card #6.
3. If levels are well-controlled, check stereo generator. Loss of modulation control will be accompanied by gross failures to meet separation and/or crosstalk specifications. Alternately, there may be a large spurious output, like 38 or 76kHz.

Bass incorrectly balanced.

1. It is normal when operating the 8100A/1 "independent" to have it accentuate bass on many records (particularly older ones). If you want the frequency balance between "Air" and "Program" to be substantially identical, operate the BASS COUPLING control closer to "wideband".
2. Possible misalignment or failure in exponential converter circuitry for either "Master" or "Bass" compressors. This will cause frequency response to be non-flat even in PROOF mode. If this is the case, check circuitry associated with IC501, IC502, IC506, IC507.
3. Failure in Input Conditioning Filter (on Cards #3 and #4). This will be revealed in PROOF mode.
4. Failure in either "Bass" or "Master" VCA, causing gain shift.

Insufficient high frequency response.

1. Due to the FM preemphasis curve, some high frequency loss is inevitable when the 8100A/1 is operated aggressively for maximum loudness (i.e., large amounts of clipping, and fast release time). To obtain more highs, back off both the CLIPPING and RELEASE TIME controls.
2. In "independent" mode, the increase in bass response on certain records may cause an <u>apparent</u> loss of highs. Try operating "wideband" temporarily to see if the highs are then balanced like the input material.
3. R626 (left channel) or R660 (right channel) misadjusted, such that IC603A (left channel) or IC603B (right channel) is always turned ON, thus partially defeating the preemphasis.
4. HF limiter working too hard. Check IC605B, IC607 (left channel); IC611B, IC612 (right channel) for correct rectifier action and correct hf limiting threshold. (These circuits are independent. Thus, the bad channel can be compared to the good channel with a mono source.)

Gross distortion.

1. Power supply voltage low. (Check AC power line voltage first.)
2. IC opamp failure. This must be diagnosed by signal tracing.

3. Failure in clipper-diode bias supplies. Low bias voltage will cause excessive clipping, and will also result in abnormally low modulation. Check IC806B, IC808 and associated circuitry (on Cards #8 and #9) to make sure that the output is approximately ±1.5VDC under no-signal conditions, and approximately ±1.35VDC when a 5kHz sinewave at level sufficient to cause gain reduction is applied to the input of the appropriate channel. Check IC613 and associated circuitry (on Card #6) to make sure that the output is approximately ±4.2VDC under all OPERATE conditions.

4. Gross failure in a sidechain, such as IC latchup. This will either misbias the main signal path, or add distortion to the main signal, without causing the main signal to disappear. IC's in sidechains include IC601A, IC608A, IC602B, IC609B, IC604, IC610 (on Card #6); IC802A, IC802B, IC803A, IC804A, IC804B (on Card #8 and Card #9).

5. Exponential converter(s) IC501, IC502, IC506, IC507, or timing module(s) A1, A2 (on card #5) defective, causing very low (or no) control current to VCA's on Cards #3 and #4, thus causing these VCA's to take very high gain. Timing module failure will be indicated by MASTER COMPRESSOR G/R or BASS G/R meter's pegging at the top of the scale (beyond 0).

Moderate to Subtle Distortion.

1. Distorted program material and/or distortion problems in studio or STL (see **Appendix K** for further discussion).

2. Check points listed in "Gross Distortion" (immediately above), for moderate deviations from normal parameters.

3. CLIPPING control misadjusted.

4. Failure in rectifiers IC503A, IC504, IC503B, IC505, IC508A, IC509, IC508B, IC510, or in timing modules A1, A2 on card #5. These problems will usually be indicated by failure to produce standard level under standard conditions (see c.1 and **c.2** in **Appendix D**).

5. Safety clipper misalignment (R841). This alignment is most unlikely to drift by itself from its factory-adjusted condition. But humans with alignment tools sometimes do strange things. If you are in doubt about this alignment, it can be checked (and readjusted if necessary) by performing Part 7 of **Appendix E.**

6. Phase corrector failure. See "Phase Corrector Failures" earlier in this Appendix for a further discussion.

7. Failure in distortion-cancel sidechain on Cards #8 and #9. This is indicated by a "gritty" high end with severe sibilance splatter.

L-R does not null on mono material.

1. This is caused by gain, frequency response, or phase response differences between the left and right channels. So before assuming that the problem is internal to OPTIMOD-FM, make sure that the feed is really 100% mono. This can be reliably assured by driving both left and right OPTIMOD-FM inputs in parallel from a single signal source.

2. If L-R will not null in PROOF mode, then the problem is static, and is caused by abnormal frequency and/or phase response in one channel. If the frequency response is normal, suspect the phase correctors on Cards #6, #8, and #9 (including A1, the phase delay network module).

3. If L-R will null in PROOF mode, then the left and right VCA's or high frequency limiter circuitry are failing to track dynamically under gain reduction conditions. In the case of the VCA's, the dual gain block (IC305, for example) is suspect. In the case of the HF limiter, the rectifiers or timing modules are suspect.

Lack of 38kHz suppression.

1. Drift in power supply voltage.
2. Excessive offset in IC701. Extreme offset or latchup of this IC will be indicated by a constant deflection of the VU meter in the L-R mode.
3. Failure of IC805A, IC806A or IC905A, IC906A (on Card #8 or #9) such that considerable DC offset appears between the left and right audio processing outputs. If the offset changes by only 40mV, this is sufficient to change the 38kHz from being perfectly nulled to being suppressed only -40dB.
4. Defective L-R modulator IC702, IC703.

Pilot phase unstable.

1. Leaky Q710.
2. Bad IC704B or leaky C716.
3. R769 intermittent.
4. Power supply voltage unstable.
5. Bad phase detector IC708.

Separation unstable.

1. Power supply voltage unstable.
2. R772 intermittent
3. IC710 defective.

SCA interference.

1. Loss of suppression of 76kHz or its sidebands. (See **7.h** in **Appendix B** for a complete discussion.)
2. Out-of-band emissions caused by FCS overshoot compensator failure forcing safety clipper to perform overshoot compensation. This will also cause loss of dynamic separation.
3. Power supply oscillation.

Sibilance Distortion.

1. Source material at OPTIMOD-FM input terminals distorted.
2. Failure of distortion-cancelling sidechain on Cards #8 or #9.
3. Failure of the HF limiter. If the HF limiter isn't working at all, then even a properly-operating distortion-cancelling clipper may generate some audible distortion.

Unit drops out of stereo mode.

1. Logic failure in IC713, IC714.
2. False pulses such as noise or rectified RF on remote terminals.

Unit drops out of mono mode.

1. See "Unit drops out of stereo mode", immediately above.
2. CROSSTALK TEST switch accidentally left in a TEST mode. Both TEST modes will force the logic into STEREO mode.

19kHz frequency out-of-tolerance.

1. It is normal for the frequency to be somewhat "off-center", as long as it is within ±2Hz of 19kHz per specifications and government requirements. If the problem is verified by your monitor service or by a high-precision calibrated frequency counter, then replace crystal Y701. (No frequency trim is provided.)

FACTORY ASSISTANCE

Orban Associates, Inc., maintains a Customer Service Department to help Orban product users who experience difficulties. Orban Customer Service is supplied at two levels. The first is telephone consultation. Often, a problem is due to misunderstanding, or is relatively simple and can be fixed by the customer aided by phone advice from the factory. Telephone consultation should always be the first step in any factory service transaction. Units will be accepted for factory service (the second level) only after consultation, and only after a Return Authorization (RA) code number has been provided by phone or letter. The RA number flags the returned unit for priority treatment when it arrives on our dock, and ties it to the appropriate information file.

The purpose of this formality is to save both the customer and the factory time and trouble by attempting to weed out problems which are caused by equipment other than OPTIMOD-FM, misapplication, or environment, and to identify those problems that lend themselves to quick field repair.

Before calling Customer Service, be prepared to give the model number (8100A/1) and serial number of your unit. If the unit is in its warranty period and the Registration Card was never returned, we will also need the name of the dealer from which the unit was bought, the invoice number, and the invoice date.

Be prepared to accurately describe the the problem. What is the complaint? Is it constant or intermittent? If it is intermittent, can it be correlated to environmental conditions like line voltage, temperature, humidity, electrical storms, vibration, etc? Do problems only occur with certain program material (live voice, very bright music, music with heavy bass transients, etc.)? What about source: cart, disc, reel-to-reel, live microphone?

Be prepared to describe any unusual observations made during the **Problem Localization Routine** you performed using the instructions above.

Then, contact the Customer Service Department by telephone, letter, or Telex (see title page for numbers). A Customer Service Engineer is ordinarily available during local business hours, Monday through Friday. The Customer Service Engineer will do everything practical to help correct the fault and have your OPTIMOD-FM up and running again as quickly as possible.

In many cases, field repairs can be effected by merely exchanging a single circuit card, rather than by returning the entire OPTIMOD-FM chassis for repair. The factory ordinarily maintains a small number of "loaner cards". One of these may be provided as a spare circuit card for use while your card is being repaired at the factory. In most cases, factory service of defective cards is preferable to field service because the factory maintains a supply of exact-replacement spare parts, and has the experienced technicians and special test fixtures necessary to assure that the repaired card meets factory specifications in all respects. Instructions for packing and shipping cards or the complete chassis are found at the end of this Appendix.

DIAGNOSIS AT THE COMPONENT LEVEL

After following the above diagnostic procedure to localize the problem to a single card, you may want to troubleshoot the card on the component level instead of returning the card to the factory for service.

Here are some suggestions....

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 the technician measures more than a few millivolts between these two terminals, the IC is probably bad.

Exceptions are IC's used without feedback (as comparators) and IC's whose outputs have been saturated due to excessive input voltage because of a defect in an earlier stage. However, if an IC's "+" input is more positive than its "-" input, yet the output of the IC is sitting at -14 volts, this almost certainly indicates that it is bad. The same holds if the above polarities are reversed. Because the characteristics of OPTIMOD-FM are essentially independent of IC opamp characteristics, an opamp can usually replaced without need for recalibration.

NOTE

THE DUAL CURRENT-CONTROLLED GAIN BLOCKS EMPLOYED IN THE VCA's AND STEREO GENERATOR L-R MODULATOR (IC 305, 309, 405, 409, & 703) ARE NOT OPAMPS. IF THEY ARE REPLACED, RECALIBRATION IS ABSOLUTELY NECESSARY.

A defective opamp may appear to work, yet it may 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 sparingly, because it can cause resistive short circuits due to moisture condensation on cold surfaces.

SELECTING AND ORDERING REPLACEMENT PARTS

Nearly all parts used in Optimod-FM have been very carefully chosen to make best use of both major and subtle characteristics. For this reason, parts should always be replaced with exact duplicates as indicated on the Parts List. It is very risky to make "close-equivalent" substitutions because of the possibility of materially altering performance and/or compliance with FCC requirements. The Factory is ordinarily able to supply any replacement part rapidly at an uncommonly reasonable price.

Specifically, such parts include all FET's and precision metal-film resistors, almost all capacitors, trimmer resistors, and integrated circuits, most transistors, and certain diodes.

Certain cards contain potted modules which, if diagnosed as defective, must be replaced as a unit. Ordinarily, this requires return of the entire card to the factory.

Certain parts are selected by the factory to tighter-than-normal specifications in order to obtain circuit performance which meets our exacting standards. Such parts are footnoted in the Parts Lists.

Certain parts, if replaced, require partial recalibration which may or may not be practical in the field. Such parts are footnoted in the Parts Lists. The recalibration requirements are outlined in the appropriate section of **Appendix B (Circuit Description)** and/or **Appendix E (Alignment).**

Service in areas involving selected parts or recalibration is best referred to the factory, which, as a result of training, experience, availability of special equipment, and availability of exact replacement parts, is generally far better qualified to perform repairs efficiently and correctly.

Ordering Parts From The Factory: If parts are ordered from the factory, we require all of the following information:

-- The Orban part number, if ascertainable from the Parts List
-- The Reference Designator (e.g., R503)
-- A brief description of the part
-- And, from the serial label on the rear of the unit:
    - the exact Model Number
    - the Serial Number
    - the "M" number, if any

REPLACEMENT OF COMPONENTS ON PRINTED CIRCUIT CARDS

It is important to use the correct technique for replacing components mounted on PC cards. Failure to do so may result in circuit damage and/or intermittent problems.

Many components, if replaced, will cause a change in calibration which will require returning the affected circuit card to the factory for recalibration. Also, some components are selected for characteristics which are not indicated by the manufacturer's part number. Most of these components are listed as "selected" on the parts list, but not all. In addition, the selection criteria are not generally described. It is therefore almost always wiser to return the defective card to the factory for service.

Most circuit cards used in OPTIMOD-FM are of the double-sided plated-through variety. This means that there are traces on both sides of the card, and that the through-holes contain a metallic plating in order to conduct current through the card. Because of the plated-through holes, solder often creeps 1/16" up into the hole, requiring a sophisticated technique of component removal to prevent serious damage to the card.

If the technician has no practical experience with the elegant and demanding technique of removing components from double-sided PC cards without card damage, it is wiser to cut each of the leads of an offending component from its body while the leads are still soldered into the card. The component is then discarded, and each lead is heated independently and pulled out of the card with a pair of long nose pliers. Each hole may then be cleared of solder by carefully heating with a low-wattage soldering iron and sucking out the remaining solder with a spring-activated desoldering tool. THIS METHOD IS THE ONLY SATISFACTORY METHOD OF CLEARING A PLATED-THROUGH HOLE OF SOLDER IN THE FIELD!

The new component may now be installed by following the directions below starting with step (4).

Otherwise, use the following technique to replace a component:

1) Use a 30-watt soldering iron to melt the solder on the solder side (underneath) of the PC card. Do not use a soldering gun or a high-wattage iron! As soon as the solder is molten, vacuum it away with a spring-actuated desoldering tool like the Edsyn Soldapullt^{R}. AVOID OVERHEATING THE CARD; overheating will almost surely damage the card by causing the conductive foil to separate from the card base.

Even with care, you are likely to blister the enamel solder-mask coating on the card, which, in most cases, is no cause for concern. The coating exists mainly to prevent moisture from condensing between the traces and to simplify wave-soldering.

2) Repeat step (1) until each lead to be removed has been cleared of solder and freed.

3) Now release the component by gently wiggling each of the leads to break solder webs. Then lift the component out.

4) Bend the leads of the replacement component until they will fit easily into the appropriate PC card holes. Using a good brand of rosin-core solder, solder each lead to the bottom side of the card with a 30-watt soldering iron. Make sure that the joint is smooth and shiny. If no damage has been done to the plated-through hole, soldering of the topside pad is not necessary. However, if the removal procedure did not progress smoothly, it would be prudent to solder each lead at the topside as well in order to avoid potential intermittent problems.

5) Cut each lead of the replacement component close to the solder (underneath) side of the PC card with a pair of diagonal cutters.

6) Remove all residual flux with a cotton swab moistened with a solvent like 1,1,1 trichloroethane, naptha, or 99% isopropyl alcohol. The first two solvents are usually available in supermarkets under the brand name Energine^{R} Fire-proof Spot Remover and Regular Spot Remover, respectively. The alcohol, which is less effective, is usually available in drug stores. Rubbing alcohol is highly diluted with water and is ineffective.

It is good policy to make sure that this defluxing operation has actually removed the flux and has not just smeared it so that it is less visible. While most rosin fluxes are not corrosive, they can slowly absorb moisture and become sufficiently conductive to cause progressive deterioration of performance.

SHIPPING INSTRUCTIONS

Circuit Cards: A circuit card is best shipped in the special Orban Associates shipping carton used to supply loaner cards. If you wish to ship a card without this carton, cut two pieces of 1" or thicker soft foam to 6.5" x 9" (17cm x 23cm) or larger. Sandwich the card between the two foam pieces, and ship the foam "sandwich" in a rigid cardboard carton.

A "JIFFY-BAG" OR SIMILAR SOFT MAILING BAG DOES NOT PROVIDE SUFFICIENT PROTECTION FOR THE CARD, AND MUST NOT BE USED!

Shipping The Complete Chassis: If the original packing material is available, it should be used. Otherwise, a sturdy, double-wall carton of at least 200 pounds bursting test and no smaller than 22" x 15" x 12" (56 x 38 x 31 cm) should be employed.

OPTIMOD-FM should be packed so that there is at least 2" of packing material protecting every point. A plastic wrap or bag around the chassis will protect the finish. Cushioning material such as Air-Cap, Bubble-Pak, foam "popcorn", or thick fiber blankets are acceptable. Folded newspaper is not suitable. Blanket-type materials should be tightly wrapped around OPTIMOD-FM and taped in place to prevent the unit from shifting out of its packing and contacting the walls of the carton.

The carton should be packed evenly and fully with the packing material filling all voids such that the unit cannot shift in the carton. Test for this by closing but not sealing the carton and shaking vigorously. If the unit can be felt or heard to move, use more packing. The carton should be well-sealed with 3" (8 cm) reinforced fiber glass or polyester sealing tape applied across the top and bottom of the carton in an "H" pattern. Narrower or parcel-post type tapes will not stand the stresses applied to commercial shipments.

The package should be marked with the name of the shipper, and the words in red: DELICATE INSTRUMENTS, FRAGILE!. Even so, the freight people will throw the box around as if it were filled with junk. The survival of the unit depends almost solely on the care taken in packing!

After a formal Return Authorization (RA) number is obtained from the factory, units should be shipped to the Service Manager at the address shown on the title page.

YOUR RETURN AUTHORIZATION NUMBER MUST BE SHOWN ON THE LABEL, OR THE PACKAGE WILL NOT BE ACCEPTED!

INSURE YOUR SHIPMENTS APPROPRIATELY!

SHIP PREPAID -- DO NOT SHIP COLLECT!

DO NOT SHIP PARCEL POST!

(Otherwise, have a nice day.)

APPENDIX G:

Changing Preemphasis

Unless specially ordered with a different preemphasis, the OPTIMOD-FM Model 8100A/1 is normally configured with 75us preemphasis. If your country's standard is 50us and you therefore wish to change the preemphasis on your unit from 75us to 50us, you must replace resistors and capacitors on card #6, as indicated in the following table. Refer to **Appendix** J for location of components.

The parts required may be obtained as kit **OPT-11** from the factory at nominal charge. When ordering, please specify both the model number and the preemphasis desired.

If you have not had much experience reworking double-sided printed circuit boards, see **Replacement of Components on PC Boards** in **Appendix F.** Verify correct operation when the modification has been completed (see **Appendix D** for a suggested method of verification).

COMPONENTS TO BE CHANGED ON CARD #6

		75us		50us	
Component	Notes	Value	ORBAN Part #	Value	ORBAN Part #
R602, R636	1	27.4k	20042.274	26.7k	20042.267
R603, R637		23.7k	20042.237	23.2k	20042.232
R604, R638	2	5.90k	20051.590	1.24k	20051.124
R605, R639		4.99k	20041.499	1.05k	20041.105
R607, R641		4.02k	20041.402	4.32k	20041.432
R608, R642		165.0k	20043.165	261.0k	20043.261
R614, R648		9.09k	20041.909	8.87k	20041.887
R617, R651		51.1k	20042.511	49.9k	20042.499
R619, R653		1.37k	20041.137	2.00k	20041.200
C603, C615	3	0.01uF	21702.310	0.047uF	21702.347
C604, C616	4	150pF	21018.115	100pF	21018.110

NOTES:

1) All resistors are Metal Film, 1/8-Watt, 1%, Style RN55D, except as noted.
2) R604 and R638 must be within 1/2% of their nominal values.
3) 2%, Polypropylene, 50V (Noble CQ15P style).
4) 1%, Mica, 500V (CM05 style CD-15).

APPENDIX H:

Detailed Exciter Interface Instructions

This Appendix provides instructions on interfacing OPTIMOD-FM to certain exciters requiring special wideband interfaces. Most exciters have straightforward wideband inputs, and no special considerations are involved.

Collins 310Z-1(B)

Prior to installing the required Continental 785-1 Wideband Interface card, this exciter must be modified using a kit of parts and instructions provided by Continental. Once this modification has been performed, proceed as in the case of the Continental 510R-1 (immediately below).

Continental 510R-1 (Collins 310Z-2)

1. Obtain a 785E-1 interface card directly from Continental.

2. Remove the 53kHz phase-linear baseband filter (FL-1), Continental Part # 673-1162-020. The filter is located on the opposite side of the chassis under the protective grill in the rear of the exciter. To access tnis filter, first remove the entire rear grill of the exciter. Next, the circuit board that covers the screws that secure the filter in its socket must be removed. The filter is plugged into an octal socket and can be readily unplugged once its hold-down screws are removed.

Despite the inconvenience, it is IMPERATIVE that this filter be removed as it shunts the baseband input to the FM modulator and its continued presence would seriously degrade separation.

3. Replace the hardware and grill removed in step (2).

4. Install the 785E-1 Interface Card in its designated slot in the card cage.

5. Be certain that the Interface Card is not being overloaded by OPTIMOD-FM. This can happen easily if the B/B LEVEL control on the modulator card of the Continental exciter is set excessively low and the OPTIMOD-FM output level is increased to make up the gain. The problem may not be immediately noticeable under test conditions, but will seriously degrade the normal operation of the system.

To avoid this condition, do not change the adjustment of the B/B LEVEL control from the setting appropriate for use with the Continental stereo generator. If there is any reason to suspect that this control has been misadjusted, it is worthwhile to check the input sensitivity. The B/B LEVEL control is correctly adjusted when a sinewave of 1.24V rms (3.5v p-p) applied to the Continental Wideband Input produces 100% modulation at any frequency.

Gates (Harris) TE-1 and TE-3

1. If you do not have a Gates (Harris) Wideband Interface Kit (P/N 994 6672 001), order the Orban ATE-3F Interface Kit (Orban P/N 04014-000-00) directly from Orban.

2. Both the Gates (Harris) and Orban interface kits contain complete instructions for installation. Bear in mind that the Gates (Harris) interface provides a balanced input. This means that the OPTIMOD-FM circuit and chassis grounds will ordinarily be jumpered together on the rear barrier strip. The Orban interface provides an unbalanced input, and the OPTIMOD-FM circuit and chassis grounds will ordinarily be unjumpered.

RCA BTE-15

1. If your exciter is not equipped with an RCA "Monaural Audio Module" (RCA P/N MI-561072), then order Orban Accessory RCA-1 (Orban P/N 05004-000) directly from Orban.

2. Install OPTIMOD-FM directly above the exciter, allowing at least 1 3/4" (1 rack unit) of air space between the units. You may want to switch the OPTIMOD-FM's LINE VOLTAGE selector to "230V" so that it can be operated from the same 230 volt circuit that ordinarily powers the exciter.

3. Using the BNC/BNC cable provided with your OPTIMOD-FM, connect the OPTIMOD-FM baseband output to the WIDEBAND BNC connector (J108) on the right rear apron of the exciter mainframe. The WIDEBAND input is the second BNC connector from the top. Be careful not to connect to the TELEMETRY input.

4. Remove the RCA BTS-1B stereo generator from the BTE-15 mainframe. If the RCA "Monaural Audio Module" is available, install it in place of the RCA stereo generator. S201, which is located on the Monaural Audio Module circuit board, must be in the EXTERNAL position.

If the "Monaural Audio Module" is not available, install the "RCA Jumper Plug" obtained in step (1) in the jack vacated by the RCA stereo generator.

5. If any of the following conditions are noted after installing OPTIMOD-FM, your BTE-15 probably has a defective varactor diode:

   a) The peak modulation level, as indicated on your modulation monitor peak flasher, seems to vary several percent with transmitter room temperature.

   b) Modulation is asymmetric.

   c) OPTIMOD-FM cannot supply enough level to modulate the exciter to 100%.

Any of these conditions should make you suspect RCA modulated oscillator diodes CR2 and/or CR3. Replacement of these diodes and realignment of the modulator is critical, and should probably be left to RCA Service.

APPENDIX J:

Schematics, Parts Locators, and Parts List

The documents in this Appendix reflect the actual construction of your unit as accurately as possible. If changes are made, they will be found in an Addendum inserted in the front of this Manual. If there is a disagreement between these drawings and your actual unit, it more likely reflects an error in documentation than an error in the construction of your unit.

If you intend to replace parts, please consult the section in **Appendix F** on **Selecting And Ordering Replacement Parts.**

Schematic drawings for the major cards face the corresponding Parts Locator Drawing.

Schematic Drawings and Parts Locator Drawings for miscellaneous assemblies and the chassis interwiring follow.

TABLE OF CONTENTS

SCHEMATICS WITH PARTS LOCATORS

Notes

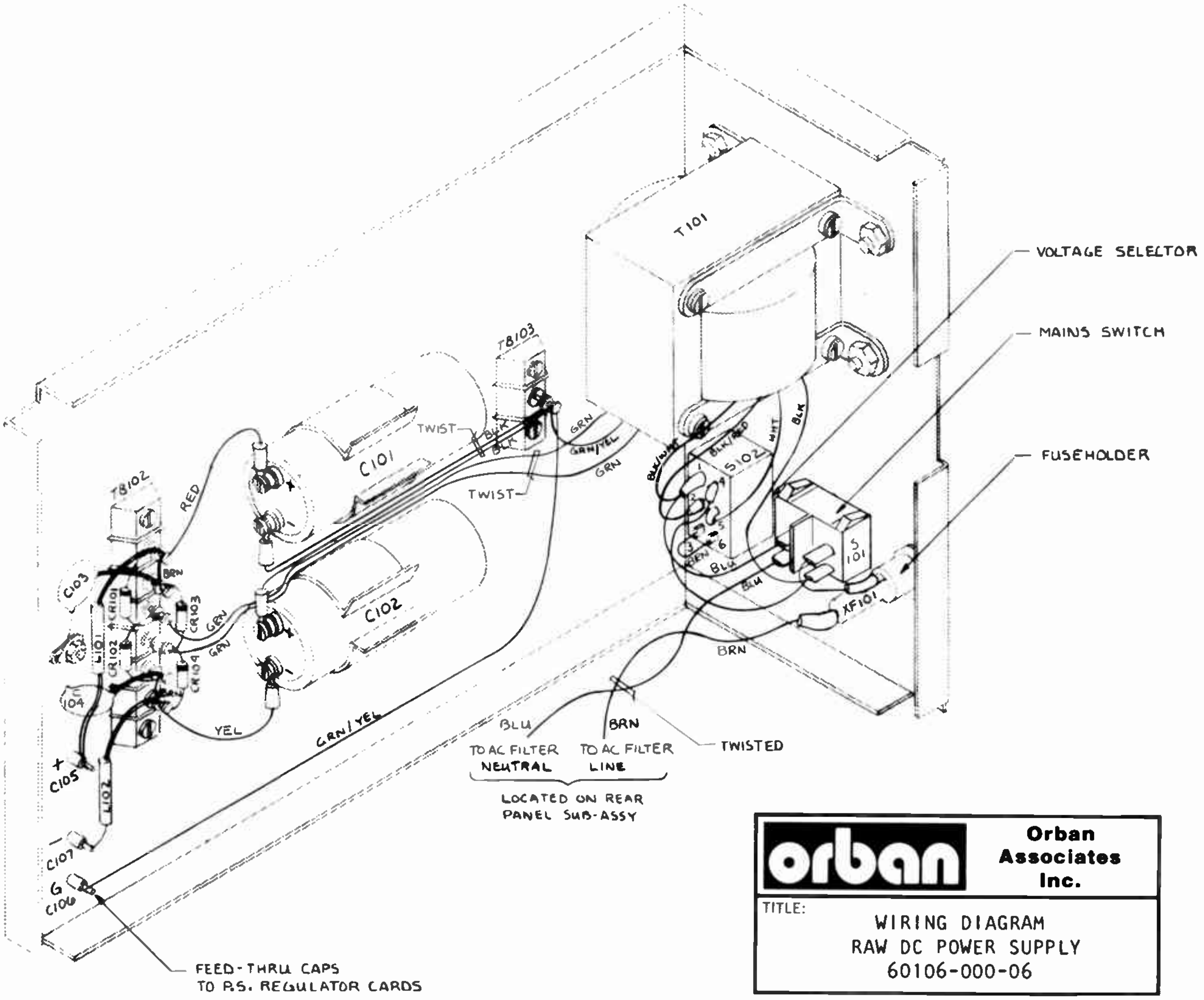
1) Chassis interwiring is indicated on the Schematics for the interconnected cards.

2) Complete information on the Studio Chassis Accessory (including the #3/4TX cards) is found in a separate Supplemental Manual shipped with the Accessory.

3) Connections for the Dolby connector and other such accessories are shown either in an Appendix of this manual or in a separate Supplemental Manual for the accessory.

PARTS LIST Indexed by assembly, by commodity, by Reference Designator. See first page for parts described only generally.

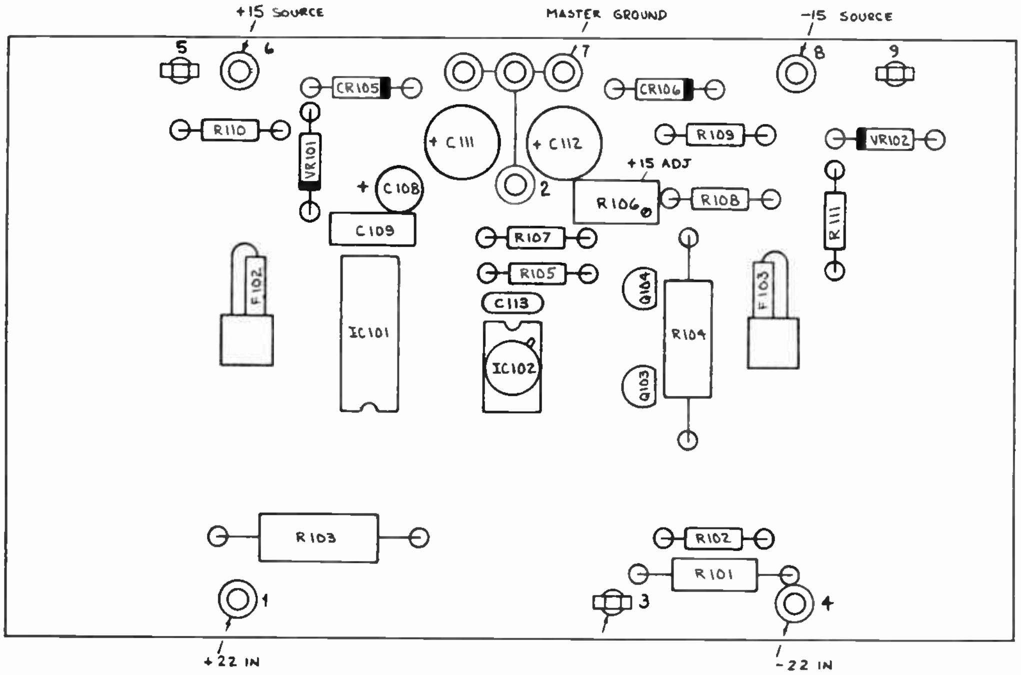
INS



orban

Orban Associates Inc.

TITLE:
WIRING DIAGRAM
RAW DC POWER SUPPLY
60106-000-06

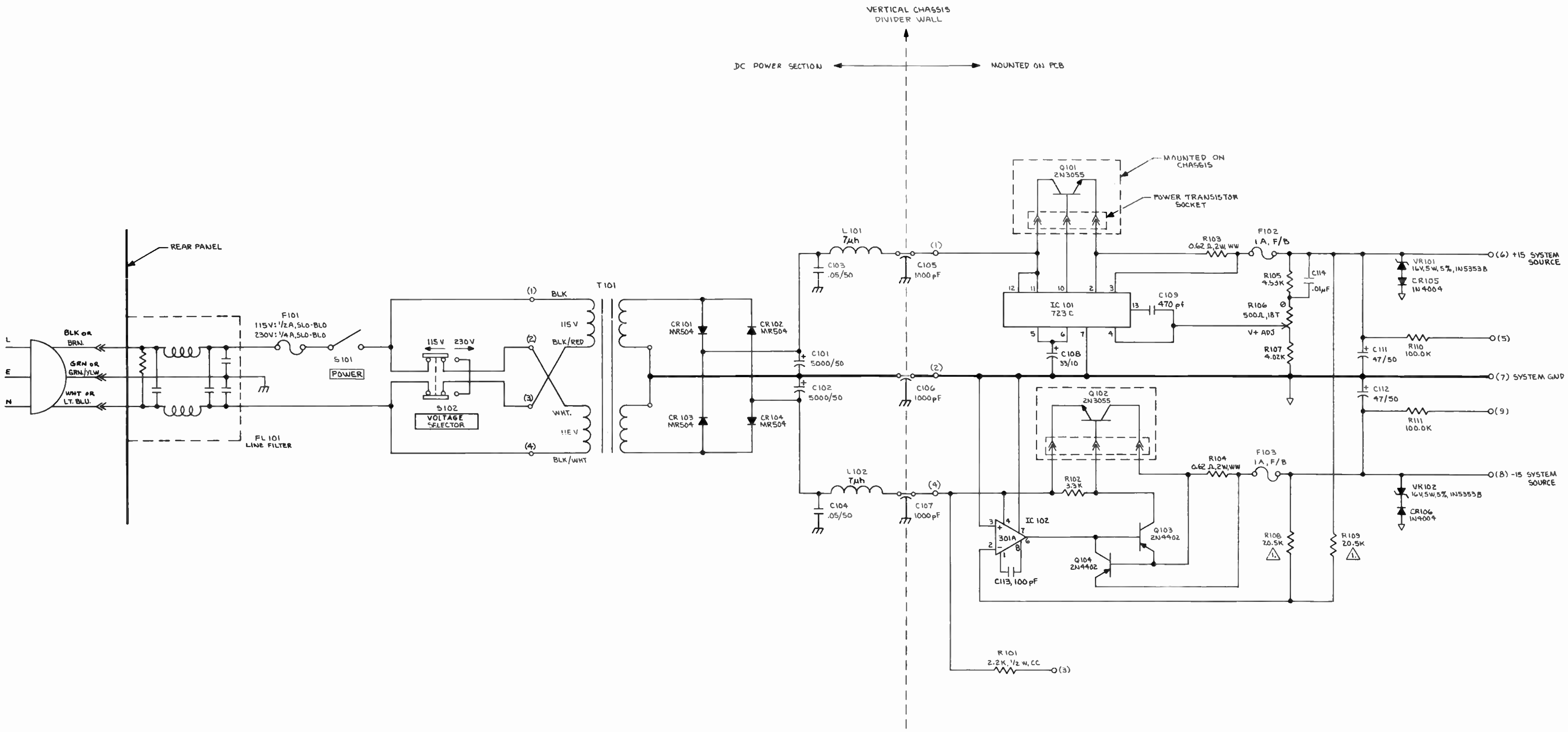


NOTE: Q101 and Q102 are mounted on Rear Panel

orban

Orban Associates Inc.

TITLE:
ASSEMBLY DRAWING
POWER SUPPLY REGULATOR
30310-000-05



REFERENCE DESIGNATORS		
ITEM	LAST USED	NOT USED
C	C114	C110
CR	106	—
IC	102	—
Q	104	—
R	111	—
S	102	—
VR	102	—
FL	101	—
T	101	—
L	102	—
F	103	—

⚠1. SELECTED: MATCHED TO 1/4%

NOTES:

J

orban

Orban
Associates
Inc.

TITLE:

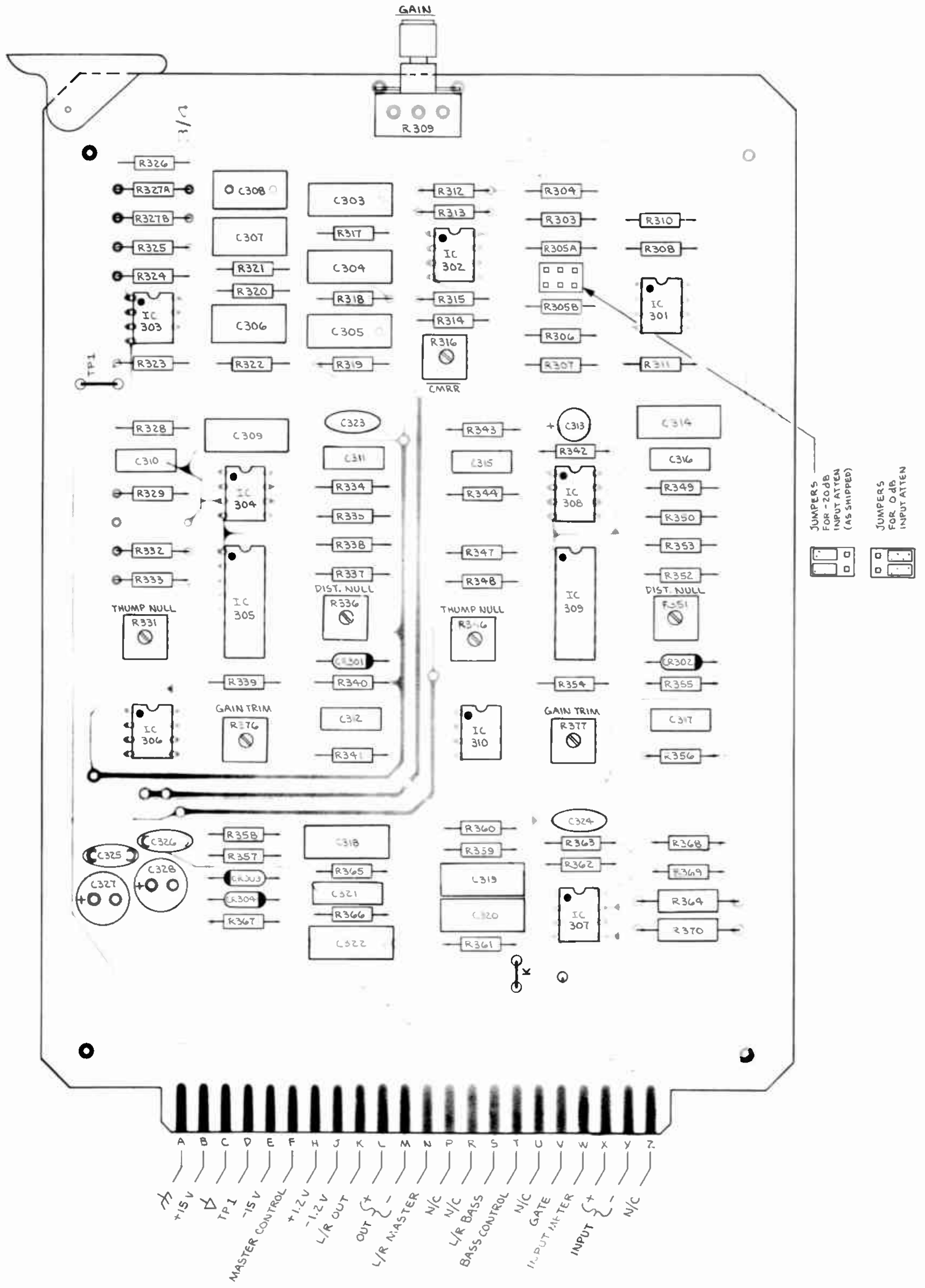
SCHEMATIC
POWER SUPPLY
60021-000-07

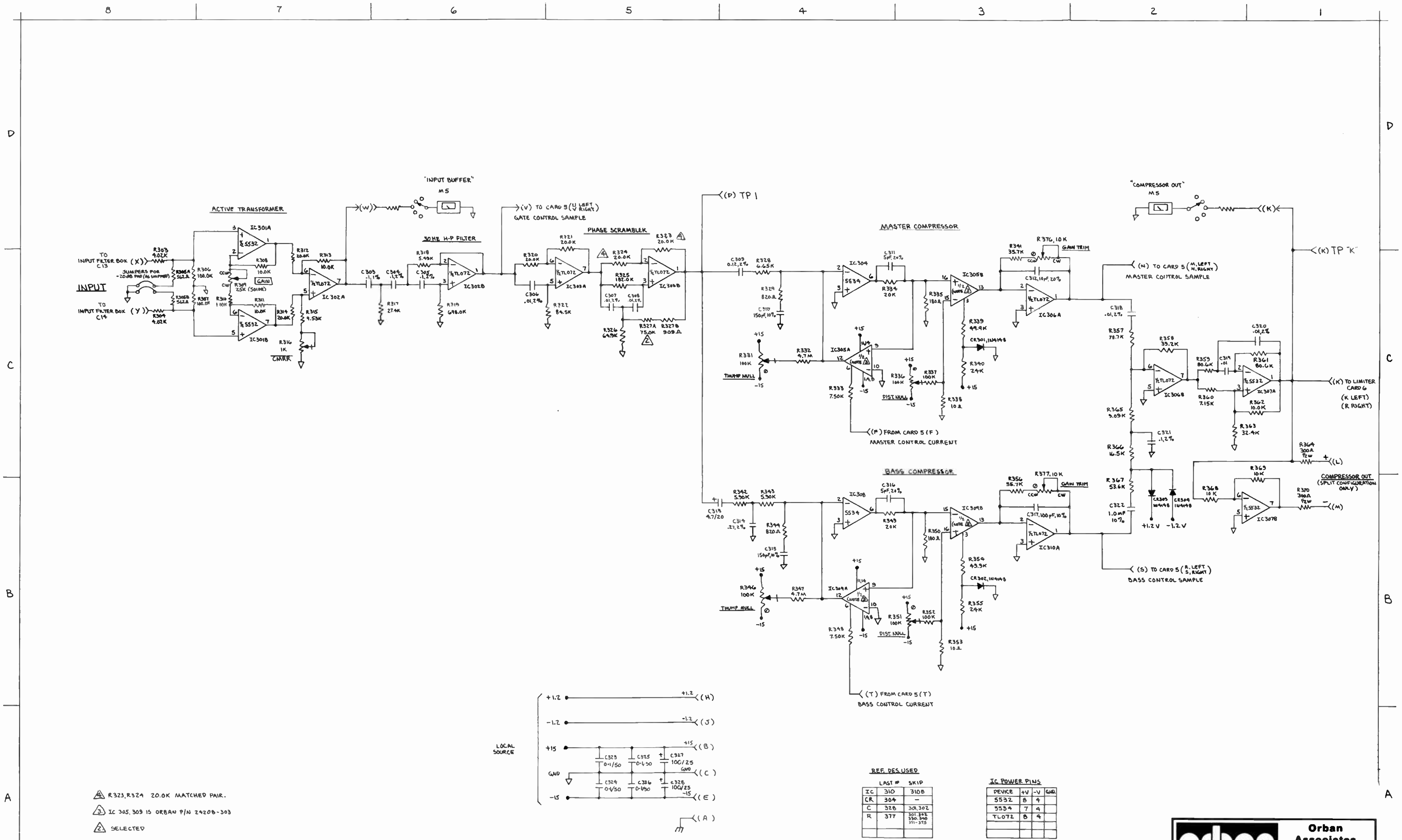
orban

Orban Associates Inc.

TITLE:

ASSEMBLY DRAWING
CARD #3/4
30430-000-05





4. R323,R324 20.0K MATCHED PAIR.

3. IC 305, 309 IS ORBAN P/N 24208-303

2. SELECTED

1. REFERENCE DESIGNATORS SHOWN FOR #3 (LEFT) CARD ONLY. #4 CARD IS 400 SERIES

NOTES:

REF. DES USED

	LAST #	SKIP
IC	310	310B
CR	304	–
C	328	301,302
R	377	301,302 330,340 371-375

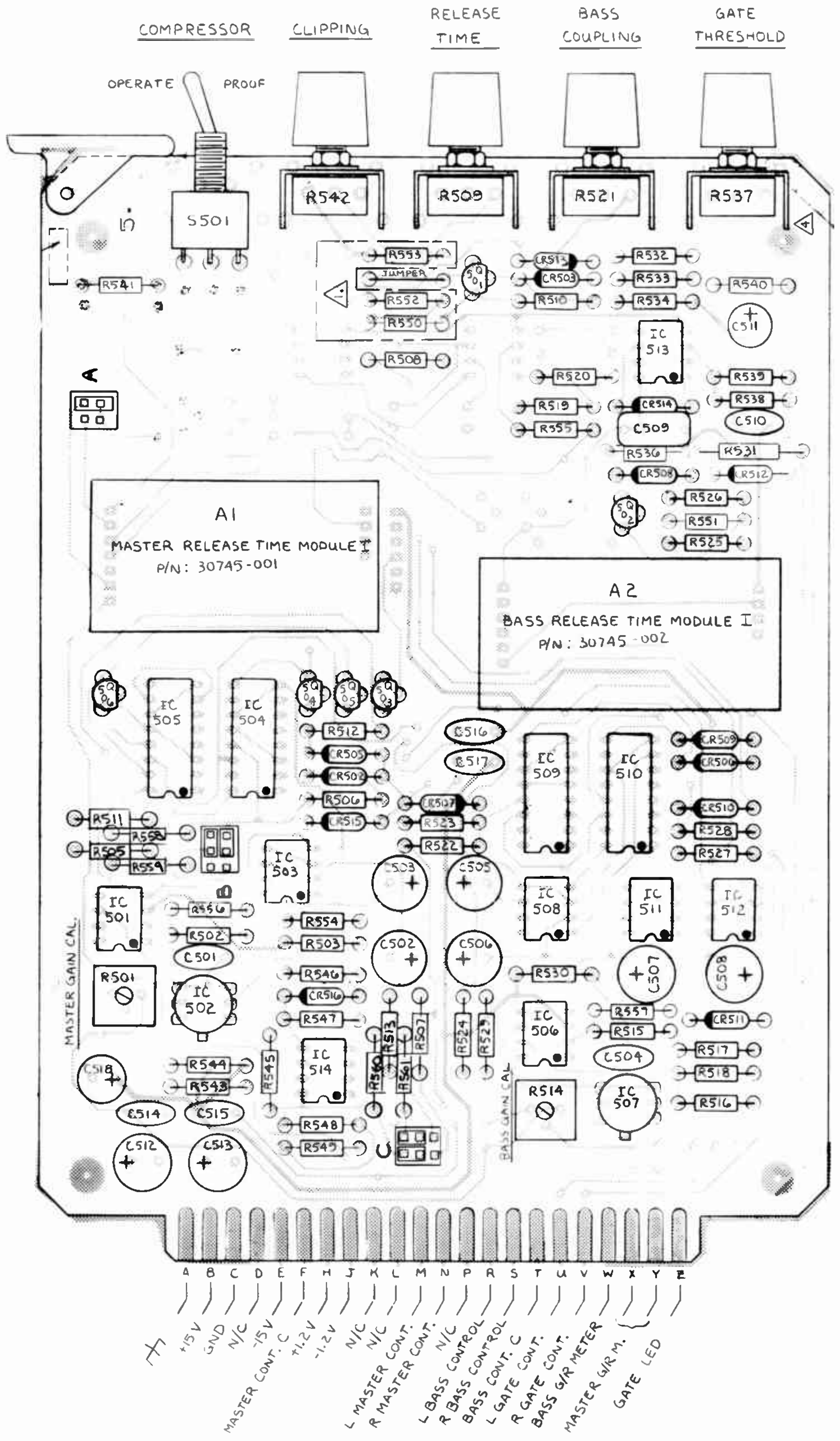
IC POWER PINS

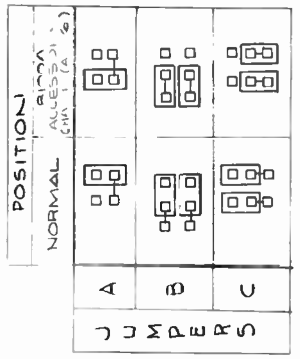
DEVICE	+V	-V	GND
5532	8	4	
5534	7	4	
TL072	8	4	

orban

Orban Associates Inc.

TITLE:
SCHEMATIC
CARD #3/4
60034-000-04

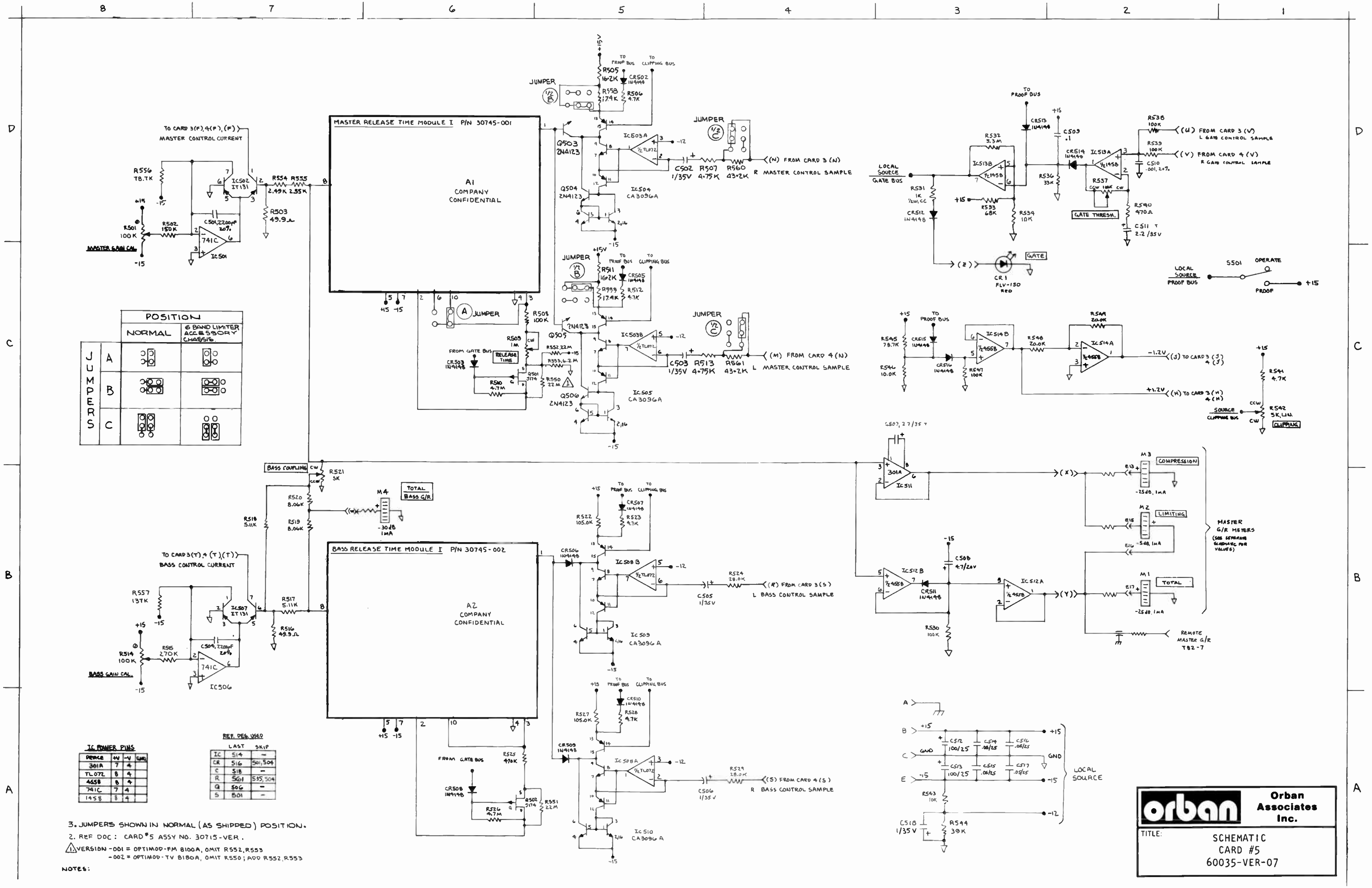




Orban Associates Inc.

orban

TITLE: ASSEMBLY DRAWING
CARD #5
30715-000-03



IC POWER PINS

DEVICE	+V	-V	GND
301A	7	4	
TL072	8	4	
4558	8	4	
741C	7	4	
1458	8	4	

REF. DES. USED

	LAST	SKIP
IC	514	–
CR	516	501,504
C	518	–
R	561	535,504
Q	506	–
S	501	–

3. JUMPERS SHOWN IN NORMAL (AS SHIPPED) POSITION.

2. REF DOC: CARD #5 ASSY NO. 30715-VER.

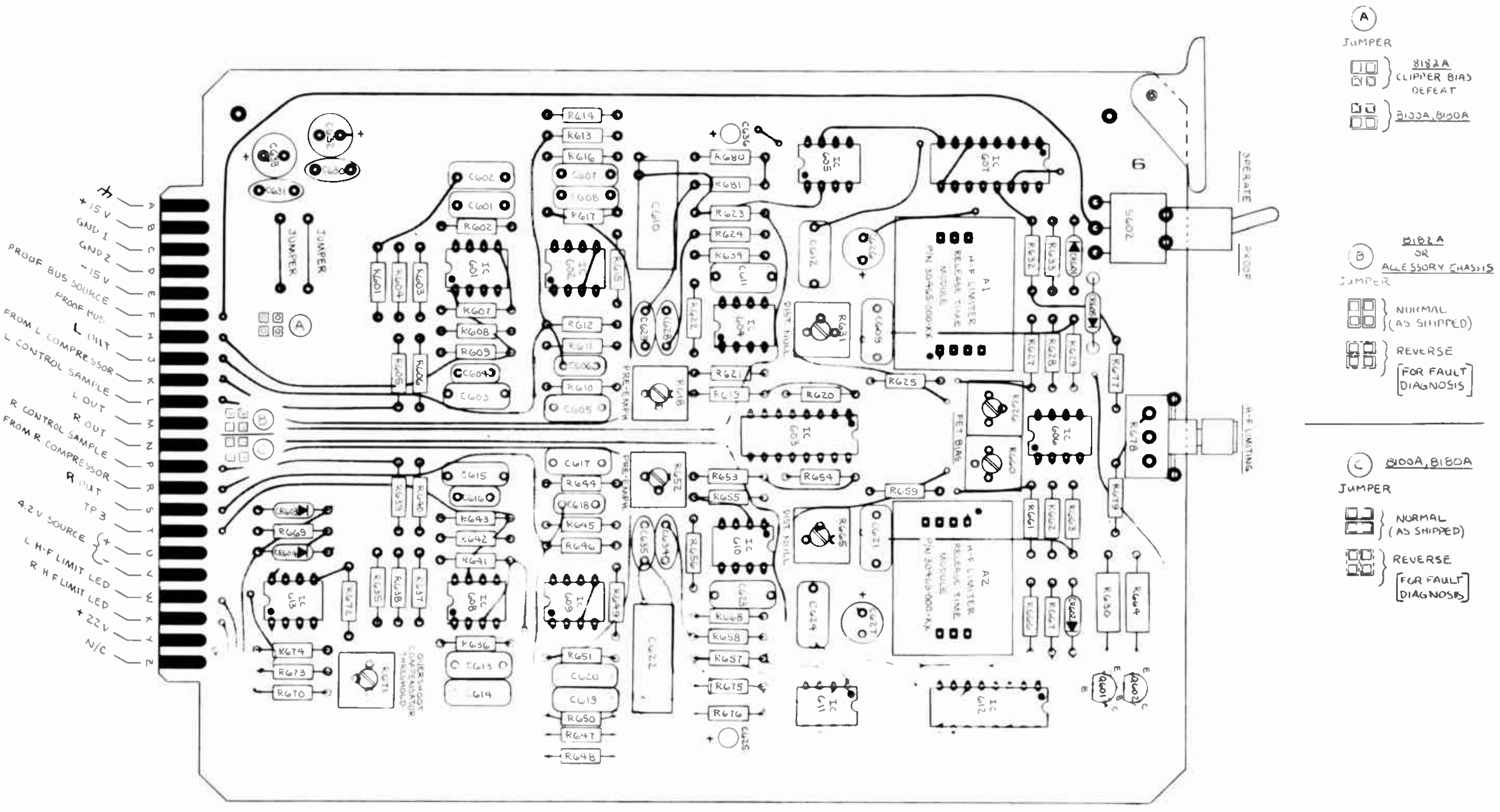
⚠1 VERSION -001 = OPTIMOD-FM 8100A, OMIT R552, R553
-002 = OPTIMOD-TV 8180A, OMIT R550; ADD R552, R553

NOTES:

orban
Orban
Associates
Inc.

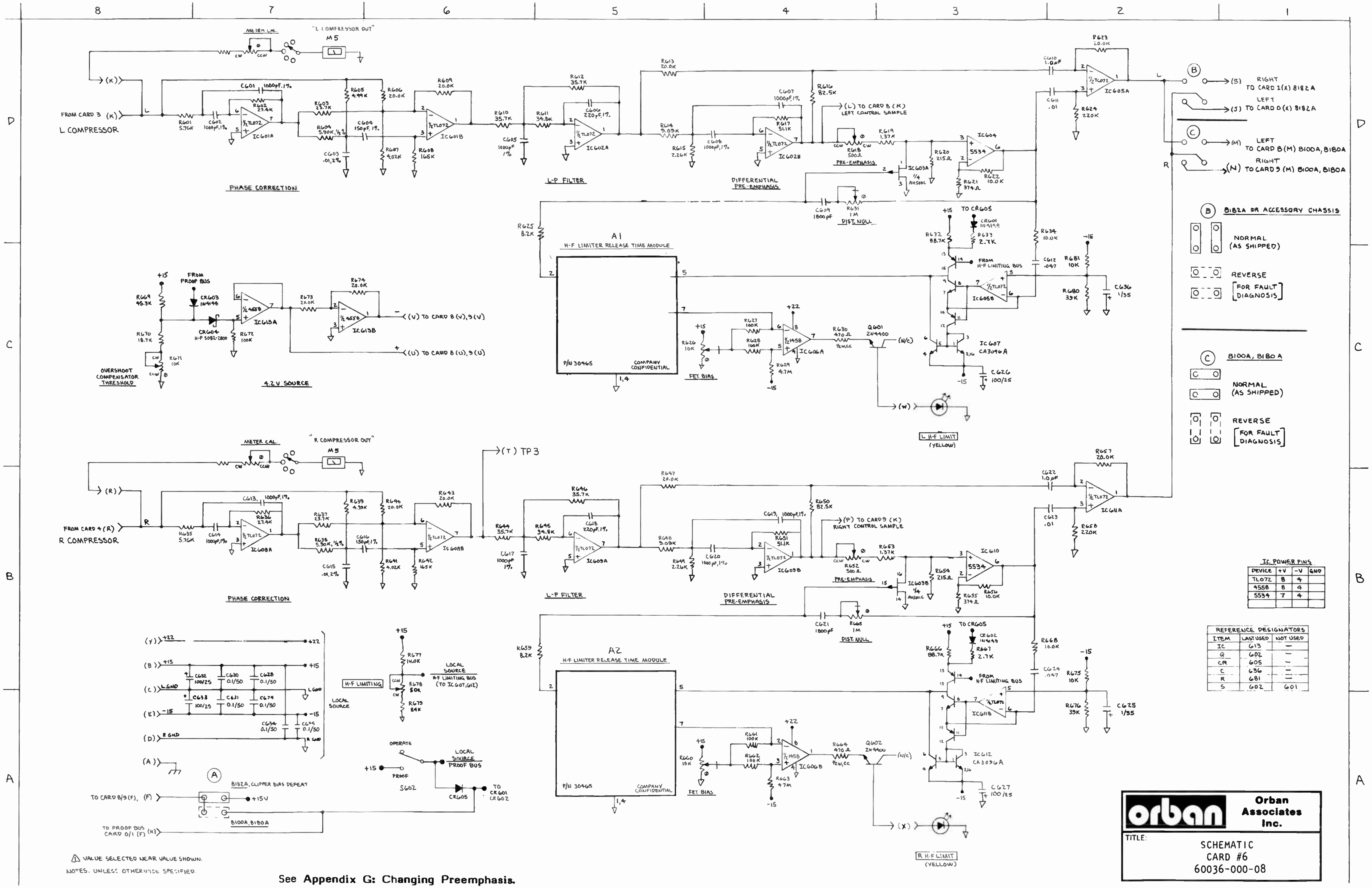
TITLE:
SCHEMATIC
CARD #5
60035-VER-07

J



orban **Orban Associates Inc.**

TITLE: ASSEMBLY DRAWING
CARD #6
30460-000-07



IC POWER PINS

DEVICE	+V	-V	GND
TL072	8	4	
4558	8	4	
5534	7	4	

REFERENCE DESIGNATORS

ITEM	LAST USED	NOT USED
IC	613	—
Q	602	—
CR	605	—
C	636	—
R	681	—
S	602	601

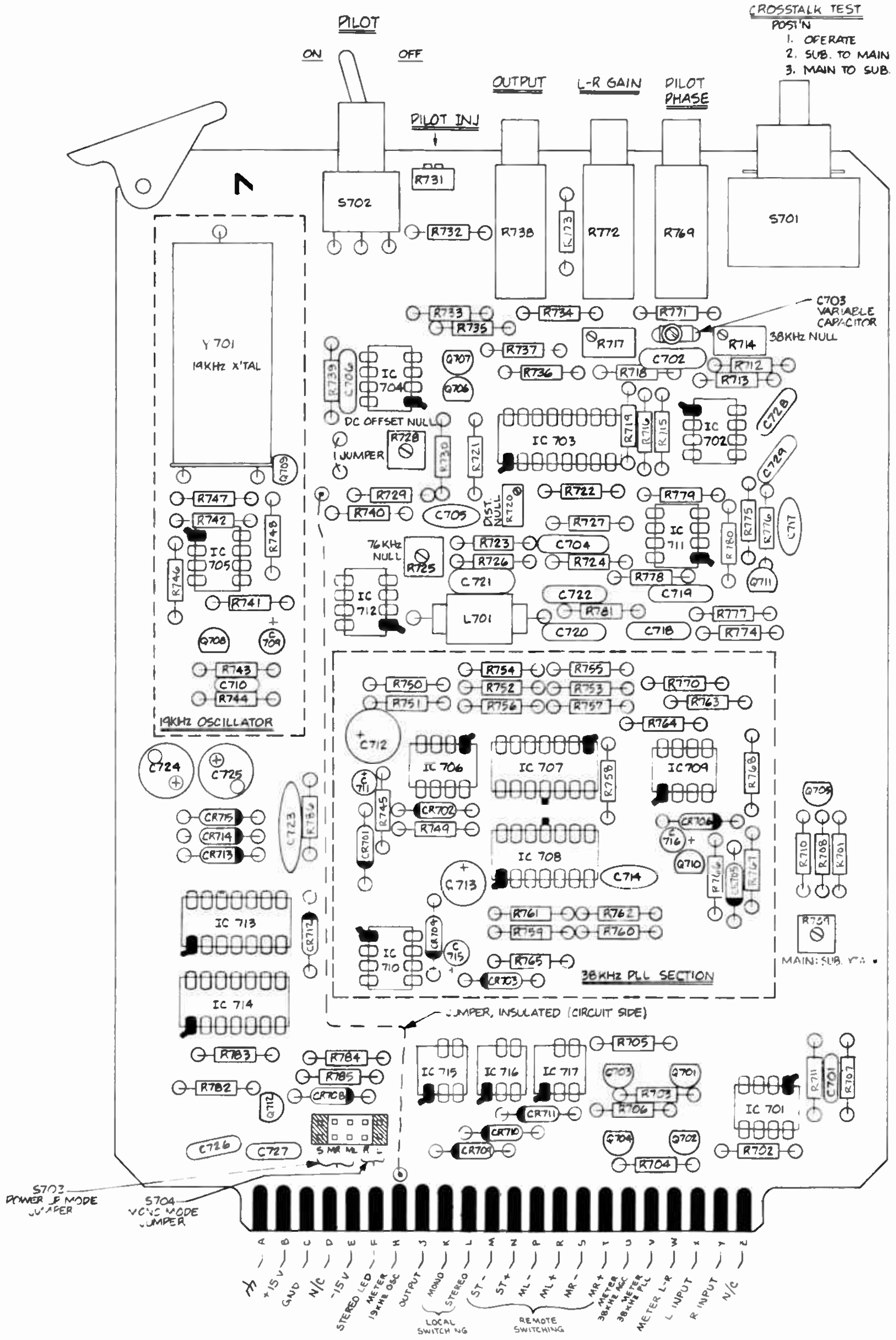
⚠ VALUE SELECTED NEAR VALUE SHOWN.

NOTES. UNLESS OTHERWISE SPECIFIED.

See **Appendix G: Changing Preemphasis.**

orban Orban Associates Inc.

TITLE:
SCHEMATIC
CARD #6
60036-000-08



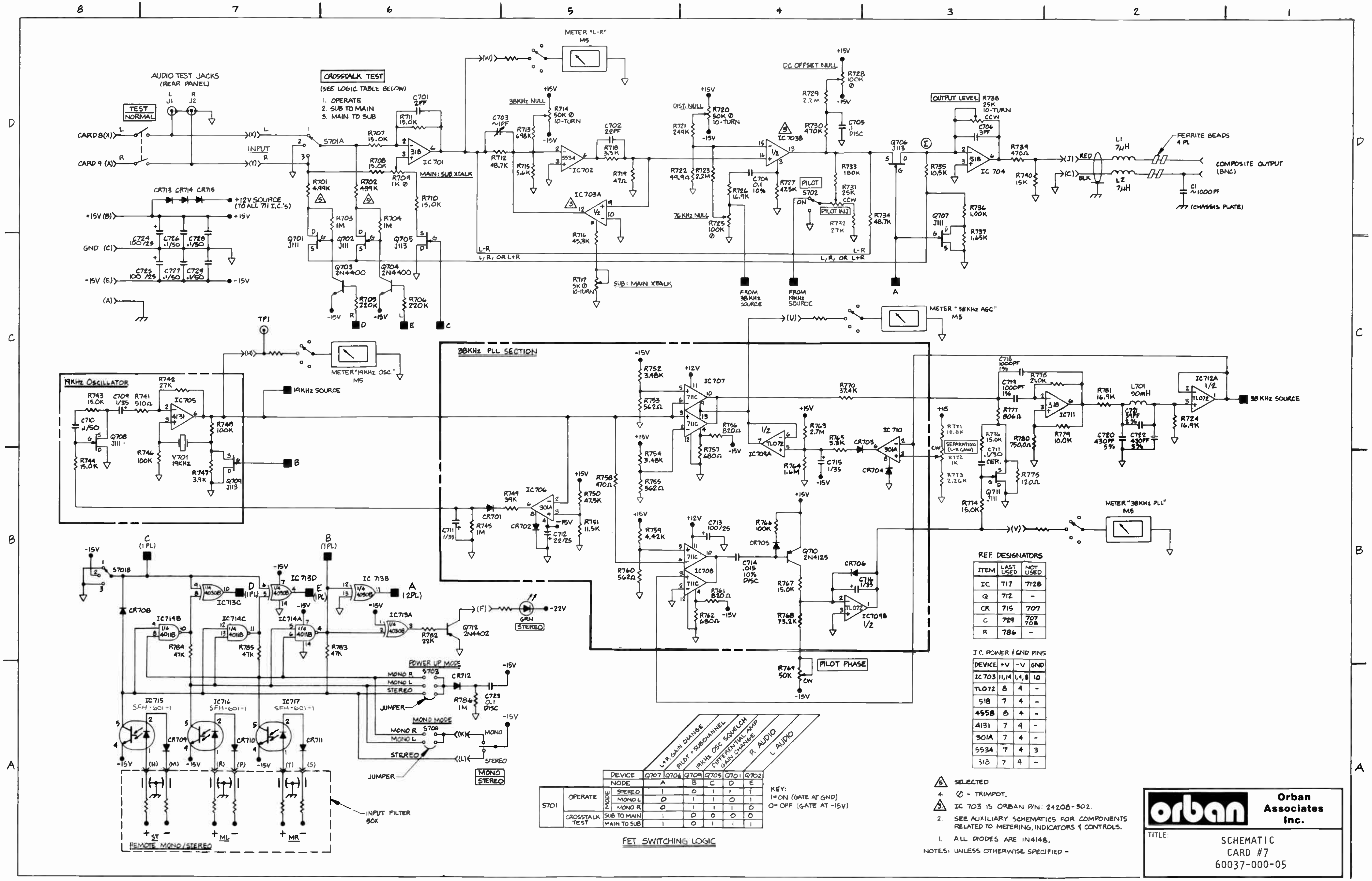
orban

Orban Associates Inc.

TITLE:
ASSEMBLY DRAWING
CARD #7
30470-000-05

NOTES:

ADJUSTMENTS: R709 MAIN TO SUB XTALK
R714 38KHz NULL
R717 SUB. TO MAIN XTALK
R720 DIST NULL
R725 76KHz NULL
R728 DC OFFSET NULL



REF DESIGNATORS

ITEM	LAST USED	NOT USED
IC	717	712B
Q	712	-
CR	715	707
C	729	707 708
R	786	-

I.C. POWER & GND PINS

DEVICE	+V	-V	GND
IC 703	11,14	1,4,8	10
TL072	8	4	-
518	7	4	-
4558	8	4	-
4131	7	4	-
301A	7	4	-
5534	7	4	3
318	7	4	-

				L+R GAIN CHANGE	PILOT + SUBCHANNEL	19KHz OSC SQUELCH	DIFFERENTIAL AMP GAIN CHANGE	R AUDIO	L AUDIO
			DEVICE	Q707	Q706	Q709	Q705	Q701	Q702
			NODE	A		B	C	D	E
S701	OPERATE	MODE	STEREO	1		0	1	1	1
			MONO L	0		1	1	0	1
			MONO R	0		1	1	1	0
	CROSSTALK TEST		SUB TO MAIN	1		0	0	0	0
			MAIN TO SUB	1		0	1	1	1

KEY:
1=ON (GATE AT GND)
0=OFF (GATE AT -15V)

FET SWITCHING LOGIC

5. SELECTED
4. ⊘ = TRIMPOT.
3. IC 703 IS ORBAN P/N: 24208-302.
2. SEE AUXILIARY SCHEMATICS FOR COMPONENTS RELATED TO METERING, INDICATORS & CONTROLS.
1. ALL DIODES ARE IN4148.

NOTES: UNLESS OTHERWISE SPECIFIED -

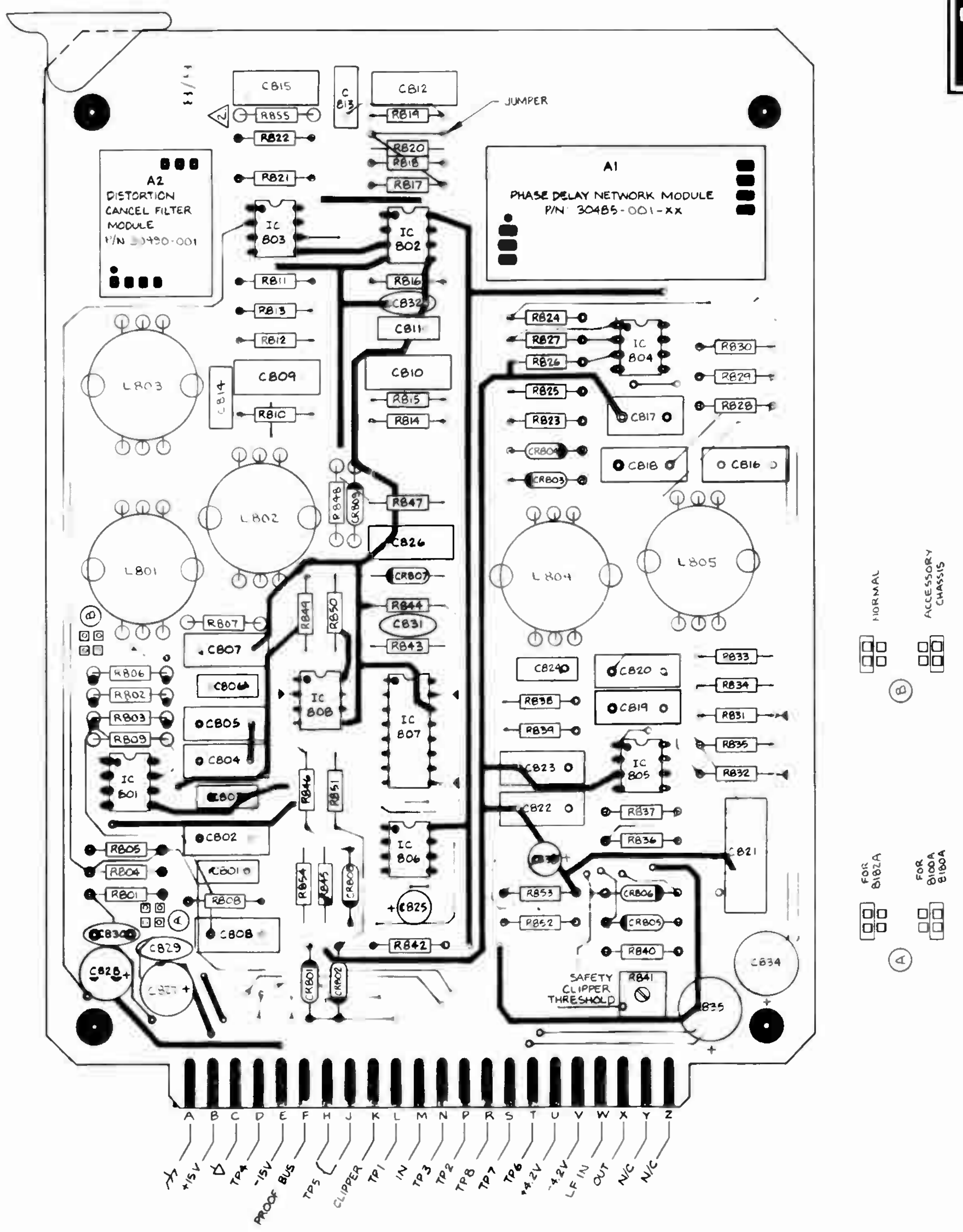
Orban Associates Inc.

TITLE:
SCHEMATIC
CARD #7
60037-000-05

orban

Orban
Associates
Inc.

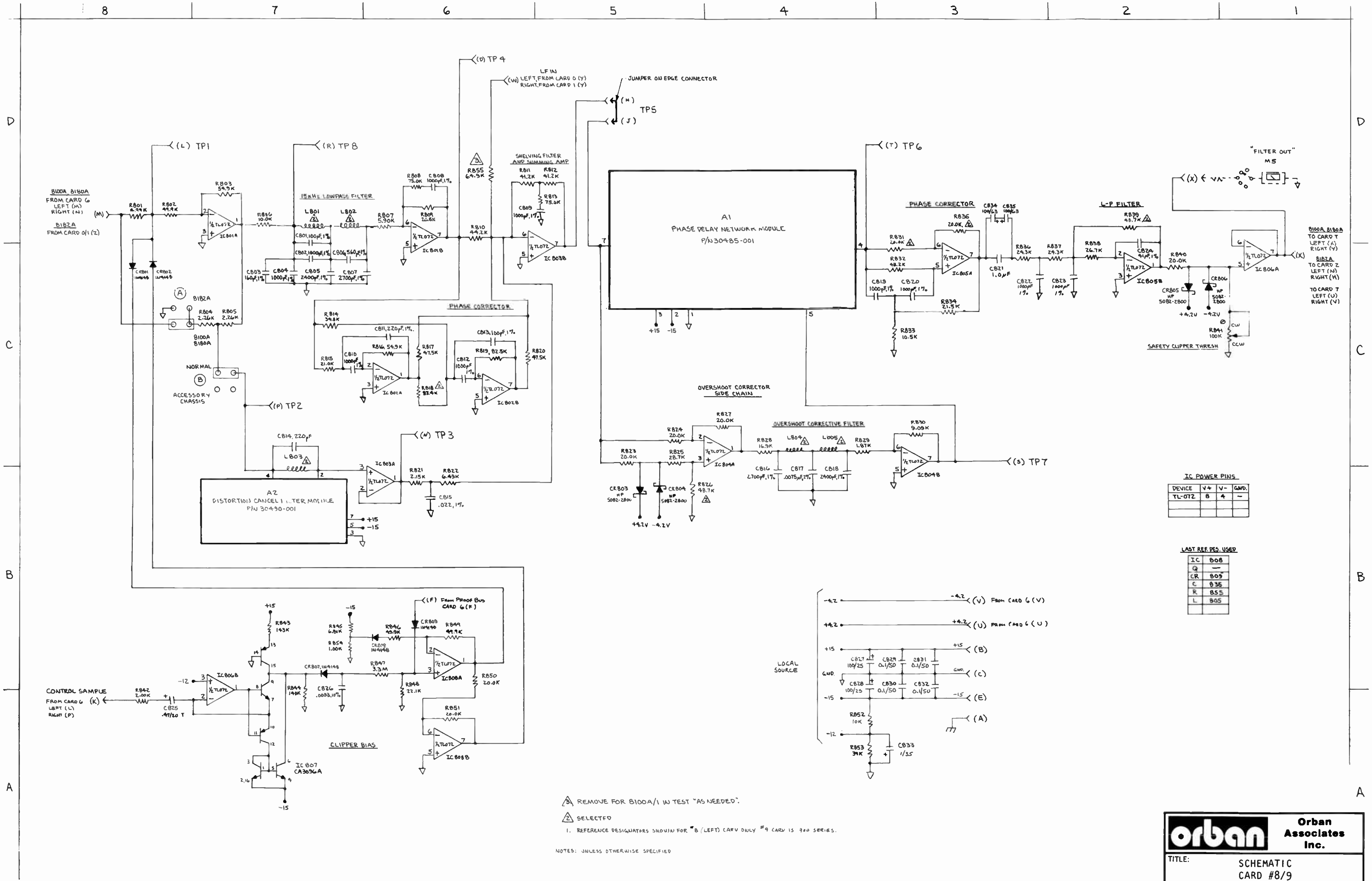
TITLE:
ASSEMBLY DRAWING
CARD #8/9
30480-VER-06



NOTES:

1. VERSIONS: -001 = CARD 8 } 8100A, 8180A, 8182A
-002 = CARD 9

△2 REMOVE FOR 8100A IN TEST "AS NEEDED"



IC POWER PINS

DEVICE	V+	V-	GND.
TL-072	8	4	—

LAST REF. DES. USED

IC	808
Q	—
CR	809
C	835
R	855
L	805

⚠3 REMOVE FOR B100A/1 IN TEST "AS NEEDED".

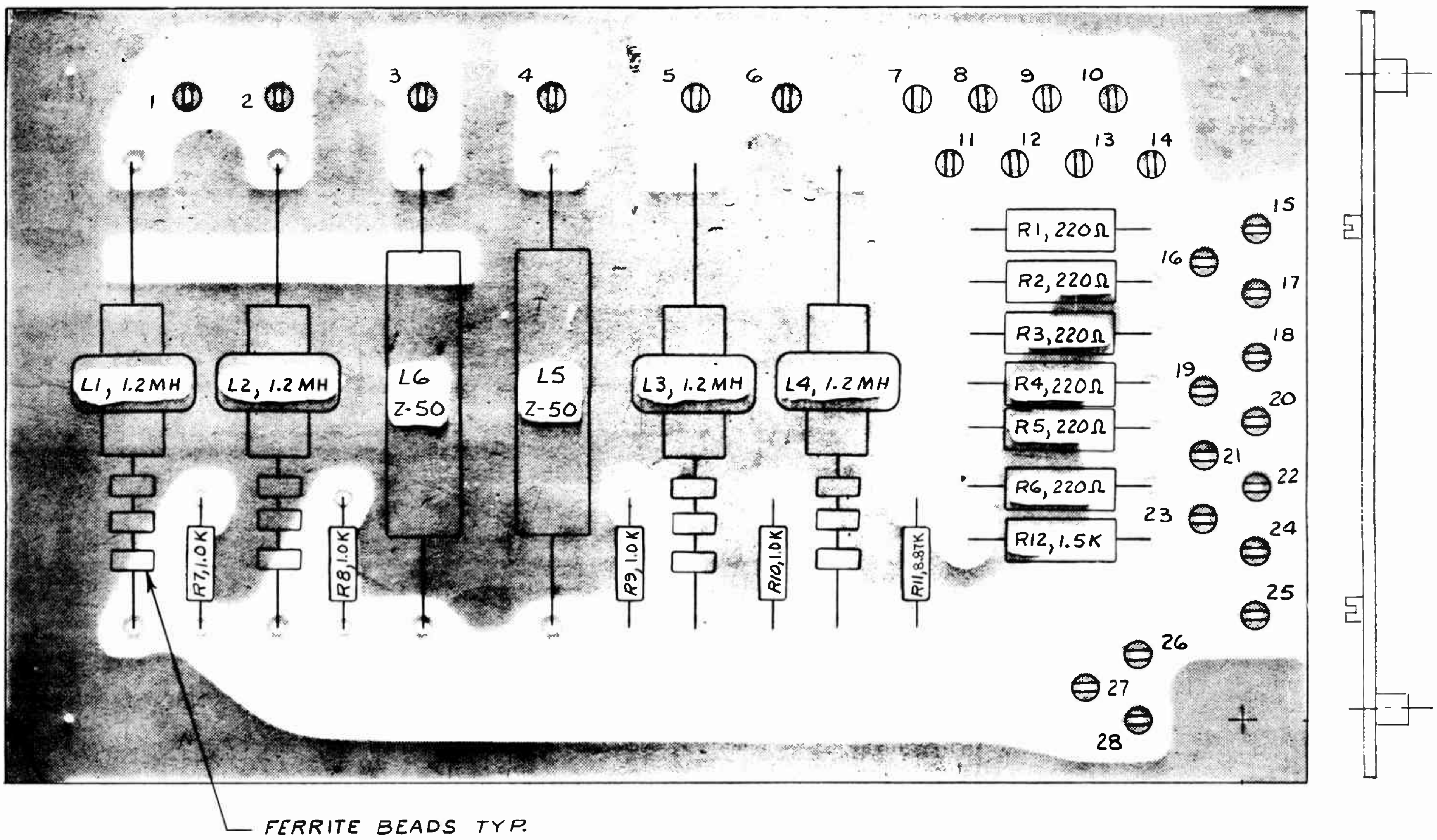
⚠2 SELECTED

1. REFERENCE DESIGNATORS SHOWN FOR #8 (LEFT) CARD ONLY #9 CARD IS 900 SERIES.

NOTES: UNLESS OTHERWISE SPECIFIED

orban Orban Associates Inc.

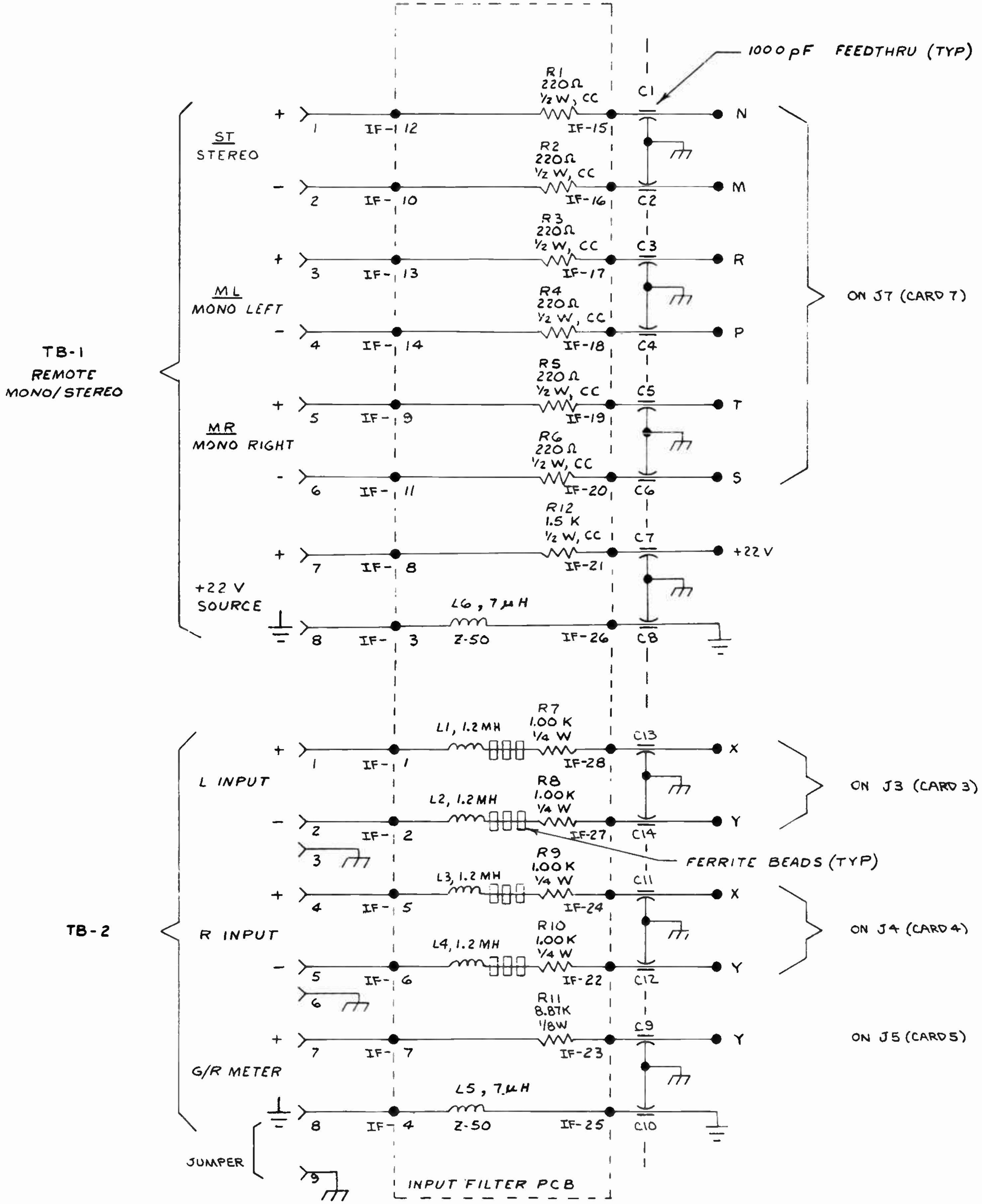
TITLE:
SCHEMATIC
CARD #8/9
60038-000-08



COMPONENT SIDE VIEW

orban Orban Associates Inc.

TITLE: ASSEMBLY DRAWING
INPUT FILTER BOARD
30495-000-02

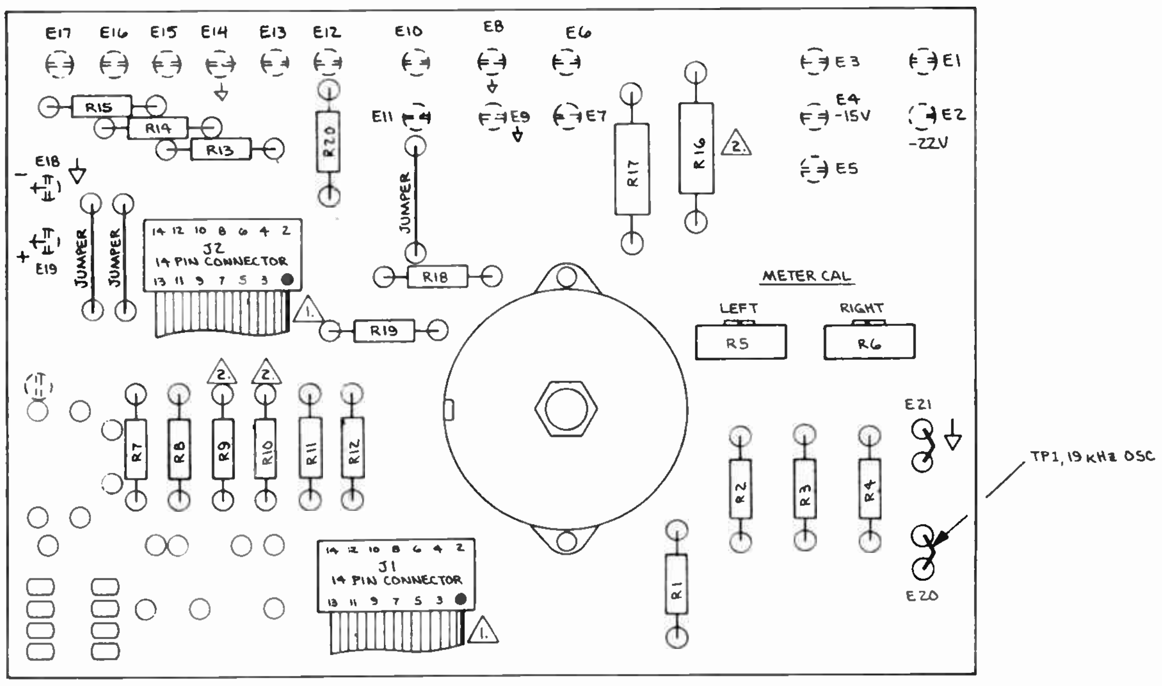


LAST REF. DES.
C 14
R 12
L6

orban Orban Associates Inc.

TITLE:
SCHEMATIC
INPUT FILTER BOARD
60039-000-03

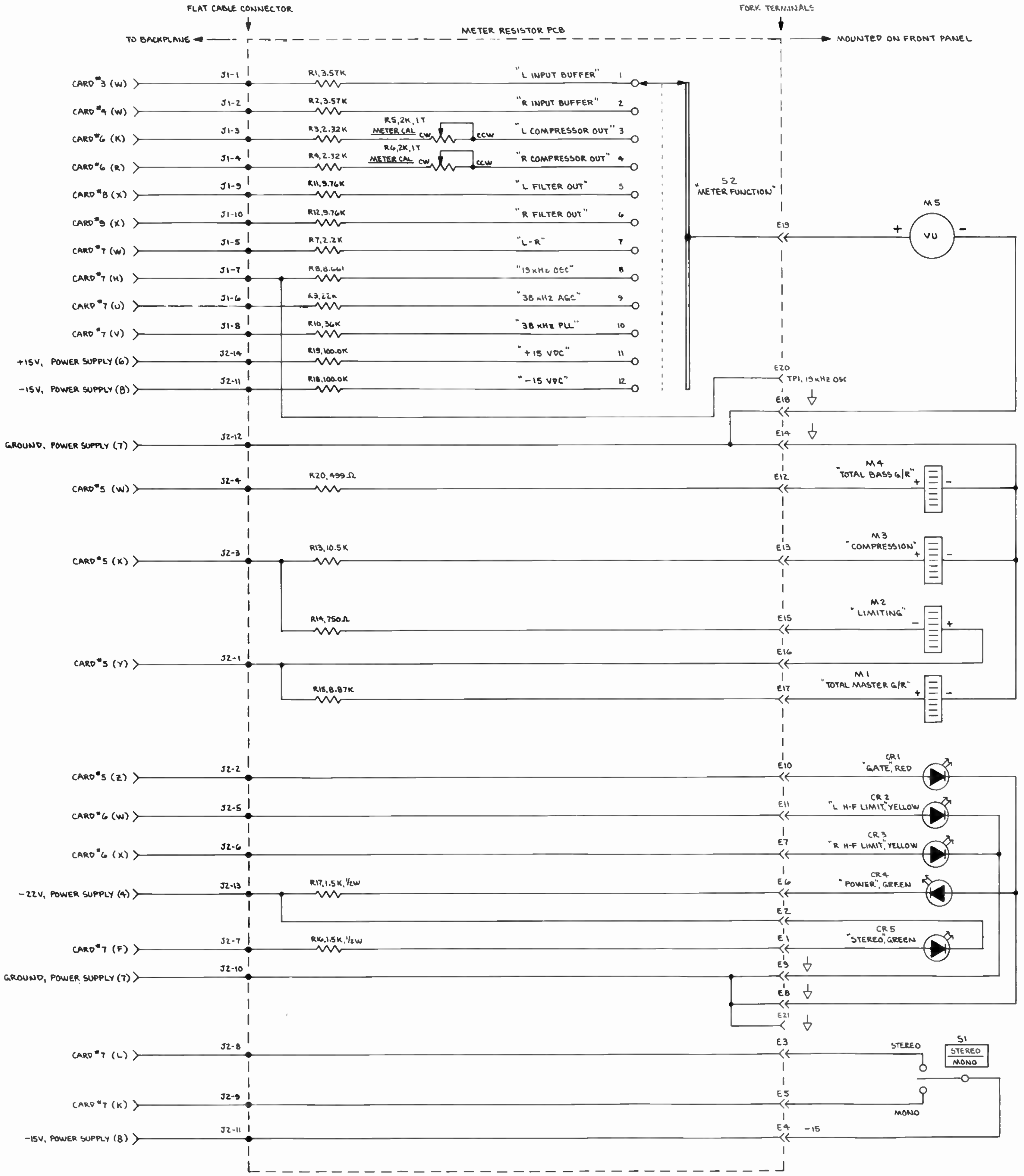
J



COMPONENT SIDE

Orban
Associates
Inc.

TITLE:
ASSEMBLY DRAWING
METER RESISTOR CARD
30440-000-07



VARIATIONS FOR 8180A

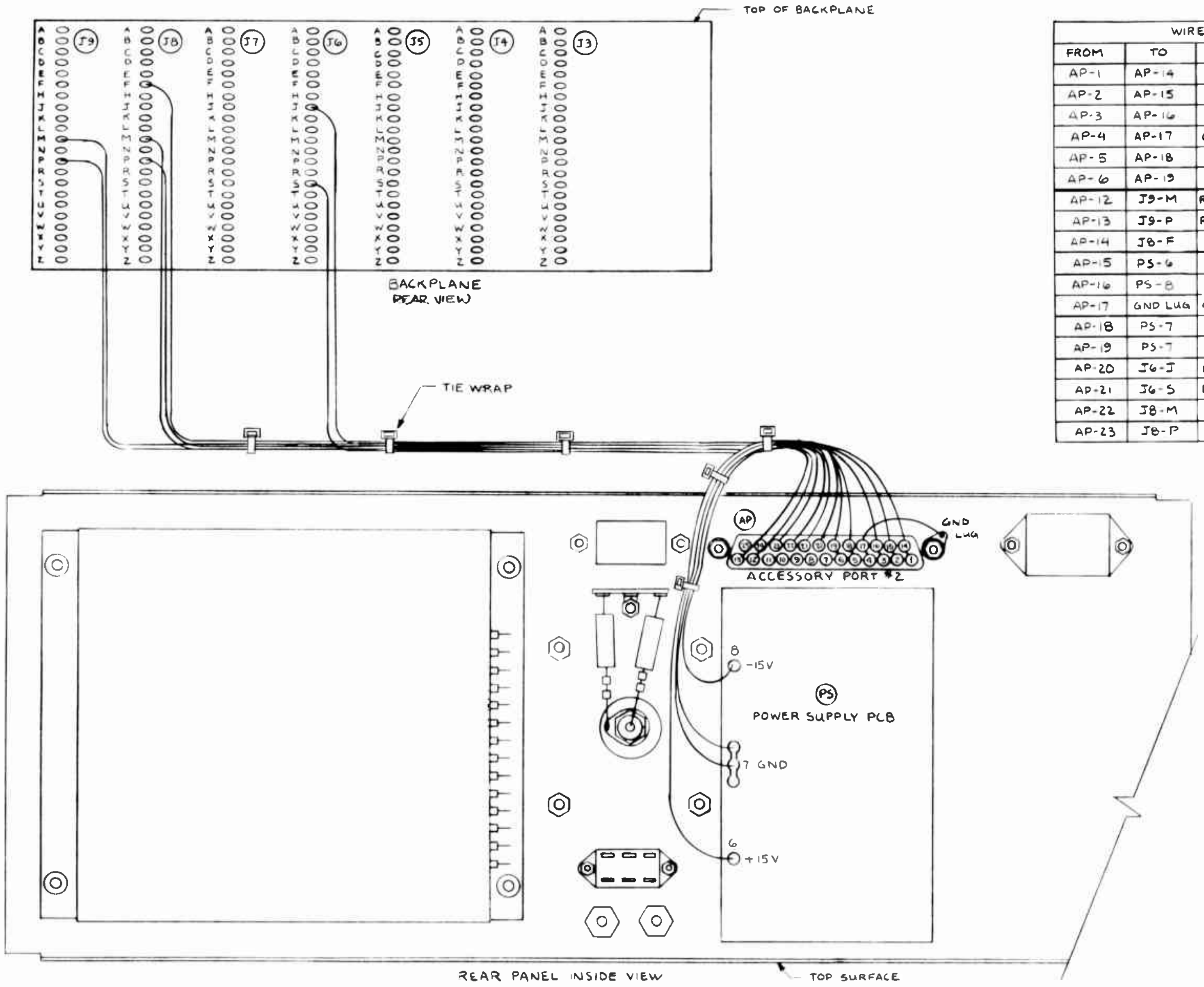
CR5	OMIT
R7	31.6K
R8	31.6K
S1	OMIT
S2-7	"L SYSTEM OUT"
S2-8	"R SYSTEM OUT"
S2-9	N/C
S2-10	N/C
TP1	OMIT
R9	OMIT
R10	OMIT
R16	OMIT

LAST REF DES. USED

CR	5
R	20
M	5
S	2

orban
Orban Associates Inc.

TITLE:
SCHEMATIC
METER RESISTOR CARD
60040-000-03



WIRE SCHEDULE					
FROM	TO	FUNCTION	COLOR	LENGTH	WIRE
AP-1	AP-14	15 V PROOF	–	.25 IN	BUSWIRE
AP-2	AP-15	+15V	–	.25 IN	"
AP-3	AP-16	−15V	–	.25 IN	"
AP-4	AP-17	CHASSIS GND	–	.25 IN	"
AP-5	AP-18	DIRTY GND	–	.25 IN	"
AP-6	AP-19	CLEAN GND	–	.25 IN	"
AP-12	J9-M	R 15 KHz FILTER	BLK	14.5 IN	INSULATED
AP-13	J9-P	R 2.2 KHz FILTER	WHT	14.5 IN	"
AP-14	J8-F	15 V PROOF	VIO	15.5 IN	"
AP-15	PS-6	+15 V	BLU	9 IN	"
AP-16	PS-8	− 15 V	GRY	7.5 IN	"
AP-17	GND LUG	CHASSIS GND	BLK	2.5 IN	"
AP-18	PS-7	DIRTY GND	BLK	7.5 IN	"
AP-19	PS-7	CLEAN GND	BRN	7.5 IN	"
AP-20	J6-J	L LIMITER OUT	WHT	12.5 IN	"
AP-21	J6-S	R LIMITER OUT	GRY	12 IN	"
AP-22	J8-M	L 15 KHz FILTER	RED	14 IN	"
AP-23	J8-P	L 2.2 KHz FILTER	ORN	14 IN	"

4\. AP 7-11, 24 & 25 ARE GND SHIELDS. (CONNECTED AT 8100A/XT UNIT)

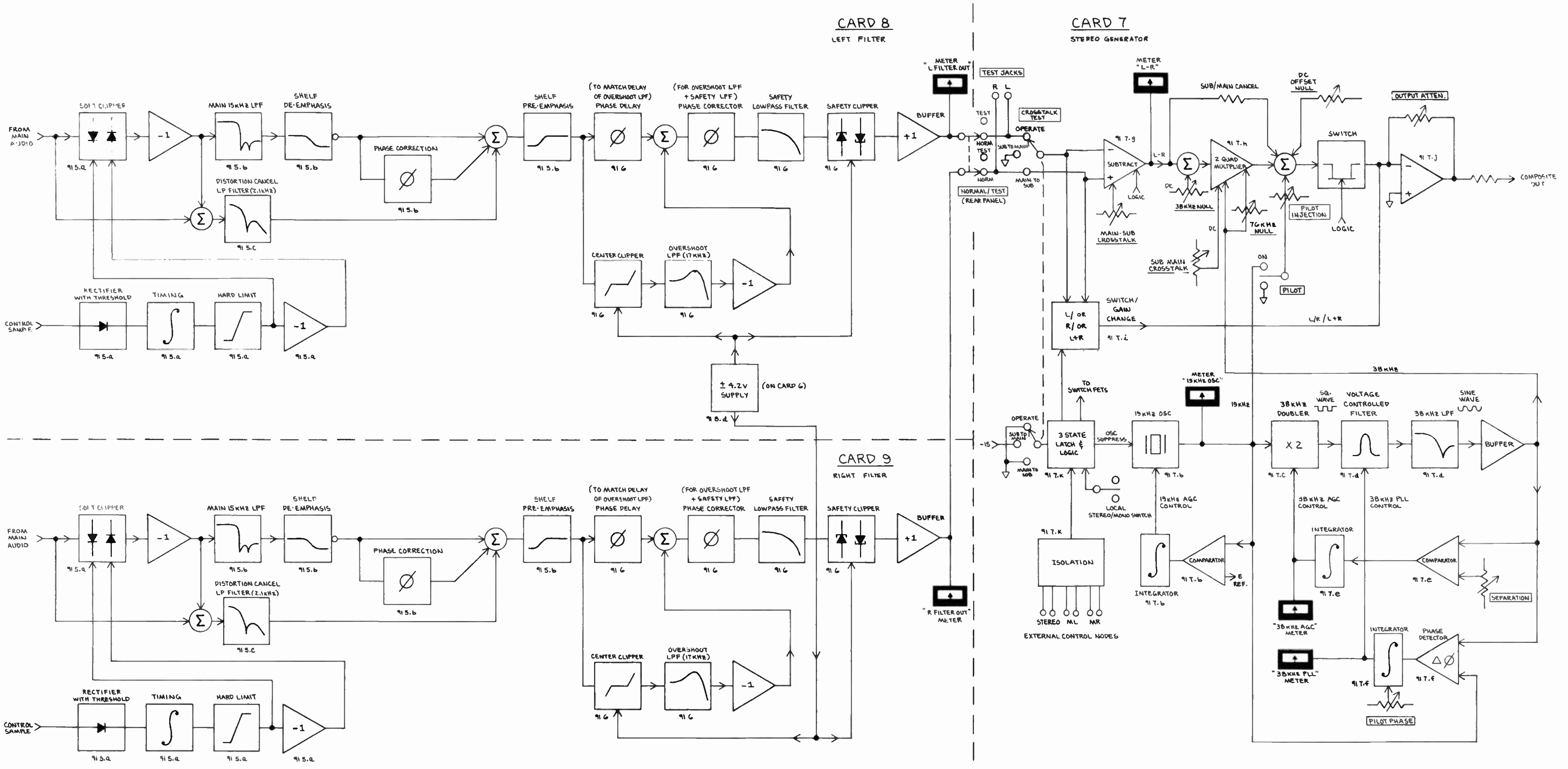
3\. REF DOCUMENTS : CONNECTOR SUB-ASSY P/N 47009-000-XX.
RET KIT #27 P/N 05033-000-XX.

NOTES: UNLESS OTHERWISE SPECIFIED

orban

Orban
Associates
Inc.

TITLE:
WIRING DIAGRAM
ACCESSORY PORT #2
60133-000-04



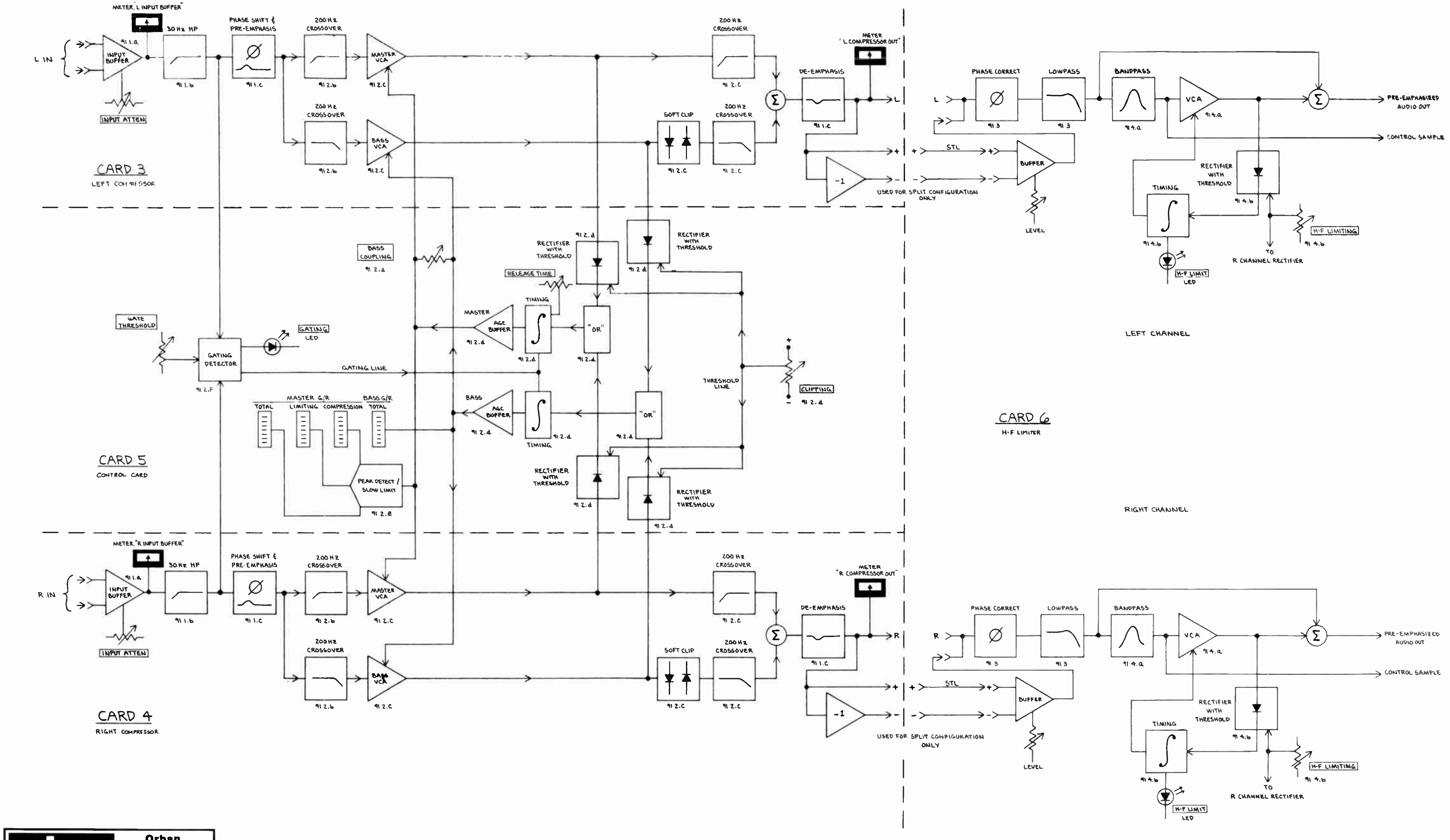
When used, the 8100A/XT Six-Band Limiter Accessory Chassis is inserted between the #6 card and the #8 and #9 cards (see Block Diagram in the 8100A/XT Operating Manual).

When used, the FM Filter Card (ACC-22, Card #0) is inserted between Card #7 and Cards #8 and #9 in this block diagram.

Orban Associates Inc.

TITLE:
BLOCK DIAGRAM
PAGE 2

J



orban **Orban Associates Inc.**

TITLE:
BLOCK DIAGRAM
PAGE 1

Parts List

Parts for this unit have been chosen from the catalogs of well-known manufacturers for ease in future maintenance. The U.S. headquarter addresses are listed at the end of the Parts List. Most manufacturers have extensive distribution facilities throughout the world and may often be contacted through local offices.

Parts are listed by assembly, by part class, in Reference Designator order except for certain widely used common parts such as:

Signal Diodes, Fixed Resistors, 3/8" Square Trimmer Resistors

which are described generally below and which must be examined to determine the exact value.

SIGNAL DIODES

ALL DIODES NOT LISTED BY REFERENCE DESIGNATOR ARE:

DESCRIPTION	ORBAN P/N	VEN	VENDOR P/N	ALTERNATE VENDORS
Diode, Signal	22101-000	FSC	1N4148	MANY

NOTE: This is a silicon small-signal diode, ultra fast recovery, high conductance. It may be replaced with 1N914 or, in Europe, with BAY-61.

(BV: 75V min. @ I_r = 5V I_r : 25nA max. @ V_r = 20V V_f : 1.0V max. @ I_f = 100mA t_{rr} : 4ns max.)

NOTE: For Zener Diodes (VR...) see Miscellaneous Section.

RESISTORS

ALL COMMON RESISTORS NOT SPECIFICALLY LISTED ARE GENERALLY SPECIFIED BELOW:
Replace resistors only with the same style and with the exact value as marked on the resistor body, lest performance or stability be compromised. If the resistor is damaged, consult the factory or refer to the Schematic to obtain the value.

Metal Film Resistors

Body: conformally-coated
I.D: five color band or printed value
Orban P/N: 20038-XXX - 20045-XXX
Power Rating: 1/8 Watt @ 70°C
Tolerance: 1%
Temperature Coefficient: 100 PPM/°C
U.S. Military Spec.: MIL-R-10509, Style RN55D
Manufacturers: R-Ohm (CRB-1/4FX), TRW/IRC, Beyschlag, Dale, Corning, Matsushita

Carbon Film Resistors

Body: conformally-coated
I.D.: four color bands
Orban P/N: 20001-XXX
Power Rating: 1/4 Watt @ 70°C
Tolerance: 5%
Manufacturers: R-Ohm (R-25), Piher, Beyschlag, Dale, Phillips, Spectrol, Matsushita

Carbon Composition Resistors

Body: molded phenolic
I.D.: four color bands
Orban P/N: 2001X-XXX
Power Rating: (70°C) 1/4 Watt (Body 0.090" x 0.250")
1/2 Watt (Body 0.140" x 0.375")
Tolerance: 5%
U.S. Military Spec.: MIL-R-11, Style RC-07 (1/4W) or RC-20 (1/2 W)
Manufacturers: Allen-Bradley, TRW/IRC, Stackpole, Matsushita

Cermet Trimmer Resistors

Body: 3/8" square (9mm)
I.D.: printed marking on side
Orban P/N: 20510-XXX, 20511-XXX
Power Rating: 1/2 Watt @ 70°C
Tolerance: 10%
Temperature Coefficient: 100 PPM/°C
Manufacturers: Beckman (72P, 68W- Series), Spectrol, Matsushita

OBTAINING SPARE PARTS

Because special or subtle characteristics of certain components are exploited in order to produce an elegant design at a reasonable cost, it is unwise to make substitutions for listed parts. It is also unwise to ignore notations in the Parts List indicating "Selected" or "Realignment Required" when replacing components. In such cases, the factory should be consulted if optimum performance is to be maintained.

Orban normally maintains an inventory of tested, exact replacement spare parts to supply any present or reasonable future demand quickly at nominal cost.

When ordering parts from the factory, we will need all of the following information:

- The Orban Part Number, if ascertainable
- The Reference Designator
- A brief description of the part

-From the Serial Label on the rear
  -The exact Model Number
  -The Serial Number
  -The "M" number, if any

Orban can supply standardized Spare Parts Kits for this product during its production life. Consult the factory for the contents of such kits and their prices.

REF DES	DESCRIPTION	ORBAN P/N	VEN (1)	VENDOR P/N	ALTERNATE VENDORS (1)	NOTES
CARD #3/4						
Capacitors						
C301,302	Not Used					
C303	Met. Polycarb., 100V, 1%; 0.1uF	21601-410	ECI	652A1B104F	IMB	
C304,305	Met. Polycarb., 100V, 2%; 0.1uF	21602-410	ECI	652A1B104G	IMB	
C306-308	Polystyrene, 50V, 2%; 0.01uF	21504-310	SPR	287P1032R5A3		
C309	Met. Polycarb., 100V, 2%; 0.12uF	21602-412	ECI	652A1B124G	IMB	
C310	Mica, 500V, 5%; 150pF	21020-115	CD	CD15-FD151J03	SAN	
C311	Mica, 500V, +1/2pF -1/2pF; 5pF	21017-005	CD	CD15-CD050D03	SAN	
C312	Mica, 500V, +1/2pF -1/2pF; 10pF	21017-010	CD	CD15-CD100D03	SAN	
C313	Tantalum, 35V, 10%; 4.7uF	21307-547	SPR	196D475X9035JA1	MANY	
C314	Met. Polycarb., 100V, 2%; 0.27uF	21602-427	ECI	652A1B274G	IMB	
C315	Mica, 500V, 5%; 150pF	21020-115	CD	CD15-FD151J03	SAN	
C316	Mica, 500V, +1/2pF -1/2pF; 5pF	21017-005	CD	CD15-CD050D03	SAN	
C317	Mica, 500V, 5%; 100pF	21020-110	CD	CD15-FD101J03	SAN	
C318-320	Polystyrene, 50V, 2%; 0.01uF	21504-310	SPR	287P1032R5A3		
C321	Met. Polycarb., 100V, 2%; 0.1uF	21602-410	ECI	652A1B104G	IMB	
C322	Met. Polyester, 100V, 10%; 1.0uF	21441-510	WES	60H105K100	WIM,SIE	
C323-326	Monolythic Ceramic, 50V, 20%; 0.1uF	21106-326	SPR	1C25Z5U104M050B		
C327,328	Alum., Radial, 25V, -20% +100%; 100uF	21206-710	PAN	ECE-A1EV101S		
C4xx	Subtract 100 and refer to C3xx series					
Integrated Circuits						
IC301	Linear, Dual Opamp	24207-202	SIG	NE5532N	TI,EXR	
IC302,303	Linear, Dual Opamp	24206-202	TI	TL072CP	MOT	
IC304	Linear, Single Opamp	24014-202	SIG	NE5534N	TI	
IC305	Linear, Dual Opamp	24208-303	ORB	24208-303		
IC306	Linear, Dual Opamp	24206-202	TI	TL072CP	MOT	
IC307	Linear, Dual Opamp	24207-202	SIG	NE5532N	TI,EXR	
IC308	Linear, Single Opamp	24014-202	SIG	NE5534N	TI	
IC309	Linear, Dual Opamp	24208-303	ORB	24208-303		
IC310	Linear, Dual Opamp	24206-202	TI	TL072CP	MOT	
IC4xx	Subtract 100 and refer to IC 3xx Series					
Resistors						
R309	Pot, Single, 25K, (5010R)	20742-000	CTS	270-Series	AB,BRN	10% CCW Log
R409	Pot, Single, 25K, (5010R)	20742-000	CTS	270-Series	AB,BRN	10% CCW Log

FOOTNOTES:
(1) See last page for 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 8100A/1 -- CARD #3/4
CAPACITORS thru RESISTORS

REF DES	DESCRIPTION	ORBAN P/N	VEN (1)	VENDOR P/N	ALTERNATE VENDORS (1)	NOTES
CARD #5						
Capacitors						
C501	Ceramic Disc, 1KV, 20%; 0.0022uF	21113-222	CRL	DD-222	SPR	
C502,503	Aluminum, Radial, -20% +100%; 1uF	21209-510	SPR	502D105G063BBIC	MANY	
C504	Ceramic Disc, 1KV, 20%; 0.0022uF	21113-222	CRL	DD-222	SPR	
C505,506	Aluminum, Radial, 63V, -20% +100%; 1uF	21209-510	SPR	502D105G063BBIC	MANY	
C507	Tantalum, 35V, 10%; 2.2uF	21307-522	SPR	196D225X9035JA1	MANY	
C508	Tantalum, 35V, 10%; 4.7uF	21307-547	SPR	196D475X9035JA1	MANY	
C509	Met. Polyester, 100V, 10%; 0.1uF	21441-410	WES	60C104K100	WIM,SIE	
C510	Ceramic Disc, 1KV, 10%; 0.001uF	21112-210	CRL	DD-102		
C511	Tantalum, 35V, 10%; 2.2uF	21307-522	SPR	196D225X9035JA1	MANY	
C512,513	Alum., Radial, 25V, -20% +100%; 100uF	21206-710	PAN	ECE-A1EV101S		
C514-517	Monolythic Ceramic, 50V, 20%; 0.1uF	21123-410	SPRC	IC25Z5U104M050B	MANY	
C518	Aluminum, Radial, 63V, -20% +100%; 1uF	21209-510	SPR	502D105G063BBIC	MANY	
Integrated Circuits						
IC501	Linear, Single Opamp	24002-202	TI	UA741CJG	RAY	
IC502	Multiple Discrete	24407-101	INS	IT131		
IC503	Linear, Dual Opamp	24206-202	TI	TL072CP	MOT	
IC504,505	Multiple Discrete	24406-302	RCA	CA3096AE		
IC506	Linear, Single Opamp	24002-202	TI	UA741CJG	RAY	
IC507	Multiple Discrete	24407-101	INS	IT131		
IC508	Linear, Dual Opamp	24206-202	TI	TL072CP	MOT	
IC509,510	Multiple Discrete	24406-302	RCA	CA3096AE		
IC511	Linear, Single Opamp	24003-202	RCA	CA301CN	NAT,TI	
IC512	Linear, Dual Opamp	24202-202	RAY	4558NB	MOT,FSC	
IC513	Linear, Dual Opamp	24203-201	MOT	MC1458CPI		
IC514	Linear, Dual Opamp	24202-202	RAY	4558NB	MOT,FSC	
Modules						
A1	Module Assy, Master Release Time	30455-001	ORB			
A2	Module Assy, Bass Release Time	30455-002	ORB			
Resistors						
R509	Pot, Single, 1 Meg, (5020)	20737-000	CTS	270-Series		20% CW Log
R521	Pot, Single, 5K, (5050)	20735-000	CTS	270-Series	BRN	Linear
R537	Pot, Single, 100K, (5020R)	20736-000	CTS	270-Series		20% CCW Log
R542	Pot, Single, 5K, (5050)	20735-000	CTS	270-Series	BRN	Linear
Switches						
S501	Switch, Toggle, Min.	26037-009	CK	7101SYA		
Transistors						
Q501,502	Transistor, JFET/P	23407-101	NAT	J174		

FOOTNOTES:
(1) See last page for 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 8100A/1 — CARD #5
CAPACITORS thru TRANSISTORS

REF DES	DESCRIPTION	ORBAN P/N	VEN (1)	VENDOR P/N	ALTERNATE VENDORS (1)	NOTES
CARD #6						
Capacitors						
C601,602	Mica, 500V, 1%; 1000pF	21022-210	CD	CD19-FD102F03	SAN	
C603	Polystyrene, 50V, 2%; 0.01uF	21504-310	SPR	287P1032R5A3		
C604	Mica, 500V, 1%; 150pF	21018-115	CD	CD15-FD151F03	SAN	
C605	Mica, 500V, 1%; 1000pF	21022-210	CD	CD19-FD102F03	SAN	
C606	Mica, 500V, 1%; 220pF	21018-122	CD	CD15-FD221F03	SAN	
C607,608	Mica, 500V, 1%; 1000pF	21022-210	CD	CD19-FD102F03	SAN	
C609	Mica, 500V, 5%; 1800pF	21024-218	CD	CD19-FD182J03	SAN	
C610	Met. Polycarb., 100V, 10%; 1.0uF	21604-510	ECI	652A1B105K	IMB	
C611	Polystyrene, 50V, 2%; 0.01uF	21504-310	SPR	287P1032R5A3		
C612	Polyester, 100V, 10%; 0.047uF	21401-347	SPR	225P47391WD3	PAN,PAK	
C613,614	Mica, 500V, 1%; 1000pF	21022-210	CD	CD19-FD102F03	SAN	
C615	Polystyrene, 50V, 2%; 0.01uF	21504-310	SPR	287P1032R5A3		
C616	Mica, 500V, 1%; 150pF	21018-115	CD	CD15-FD151F03	SAN	
C617	Mica, 500V, 1%; 1000pF	21022-210	CD	CD19-FD102F03	SAN	
C618	Mica, 500V, 1%; 220pF	21018-122	CD	CD15-FD221F03	SAN	
C619,620	Mica, 500V, 1%; 1000pF	21022-210	CD	CD19-FD102F03	SAN	
C621	Mica, 500V, 5%; 1800pF	21024-218	CD	CD19-FD182J03	SAN	
C622	Met. Polycarb., 100V, 10%; 1.0uF	21604-510	ECI	652A1B105K	IMB	
C623	Polystyrene, 50V, 2%; 0.01uF	21504-310	SPR	287P1032R5A3		
C624	Polyester, 100V, 10%; 0.047uF	21401-347	SPR	225P47391WD3	PAN,PAK	
C625	Tantalum, 35V, 10%; 1uF	21307-510	SPR	196D105X9035HA1	MANY	
C626,627	Alum., Radial, 25V, -20% +100%; 100uF	21206-710	PAN	ECE-A1EV101S		
C628-631	Monolythic Ceramic, 50V, 20%; 0.1uF	21123-410	SPR	1C25Z5U104M050B	MANY	
C632,633	Alum., Radial, 25V, -20% +100%; 100uF	21206-710	PAN	ECE-A1EV101S		
C634,635	Monolythic Ceramic, 50V, 20%; 0.1uF	21123-410	SPR	1C25Z5U104M050B	MANY	
C636	Tantalum, 35V, 10%; 1uF	21307-510	SPR	196D105X9035HA1	MANY	
Diodes						
CR604	Diode, Signal, Hot Carrier	22102-001	HP	5082-2800		

FOOTNOTES:
- (1) See last page for 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 8100A/1 -- CARD #6
CAPACITORS and DIODES

REF DES	DESCRIPTION	ORBAN P/N	VEN (1)	VENDOR P/N	ALTERNATE VENDORS (1)	NOTES
Integrated Circuits						
IC601,602	Linear, Dual Opamp	24206-202	TI	TL072CP	MOT	
IC603	Multiple Discrete	24405-303	NAT	AH5011CN		
IC604	Linear, Single Opamp	24014-202	SIG	NE5534N	TI	
IC605	Linear, Dual Opamp	24206-202	TI	TL072CP	MOT	
IC606	Linear, Dual Opamp	24203-201	MOT	MC1458CPI		
IC607	Multiple Discrete	24406-302	RCA	CA3096AE		
IC608,609	Linear, Dual Opamp	24206-202	TI	TL072CP	MOT	
IC610	Linear, Single Opamp	24014-202	SIG	NE5534N	TI	
IC611	Linear, Dual Opamp	24206-202	TI	TL072CP	MOT	
IC612	Multiple Discrete	24406-302	RCA	CA3096AE		
IC613	Linear, Dual Opamp	24202-202	RAY	4558NB	MOT,FSC	
Modules						
A1,2	Module Assy, H-F Limiter Release Time	30465-000	ORB			
Switches						
S602	Switch, Toggle, Min.	26037-009	CK	7101SYA		
Transistors						
Q601,602	Transistor, Signal, NPN	23202-101	MOT	2N4400	FSC	

FOOTNOTES:

(1) See last page for 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 8100A/1 — CARD #6, Cont'd
INTEGRATED CIRCUITS thru TRANSISTORS

REF DES	DESCRIPTION	ORBAN P/N	VEN (1)	VENDOR P/N	ALTERNATE VENDORS (1)	NOTES
CARD #7						
Capacitors						
C701	Mica, 500V, +1/2pF -1/2pF; 2pF	21017-002	CD	CD15-CD010D03	SAN	
C702	Mica, 500V, 5%; 22pF	21020-022	CD	CD15-ED220J03	SAN	
C703	Ceramic, Trimpot, 0.5pF-3pF	21811-000	ME	2502AOR503V		
C704	Met. Polyester, 100V, 10%; 0.1uF	21441-410	WIM	MKS-4100V5.0.1	WES,SIE	
C705	Ceramic Disc, 25V, 20%; 0.1uF	21106-410	CRL	UK25-104		
C706	Mica, 500V, +1/2pF -1/2pF; 3pF	21017-003	CD	CD15-CD030D03	SAN	
C707,708	Ceramic Disc, 25V, 20%; 0.1uF	21106-410	CRL	UK25-104		
C709	Tantalum, 35V, 10%; 1uF	21307-510	SPR	196D105X9035HA1	MANY	
C710	Monolythic Ceramic, 50V, 20%; 0.1uF	21123-410	SPR	1C25Z5U104M050B		
C711	Tantalum, 35V, 10%; 1uF	21307-510	SPR	196D105X9035HA1	MANY	
C712,713	Alum., Radial, 25V, -20% +100%; 100uF	21206-710	PAN	ECE-A1EV101S		
C714	Ceramic Disc, 50V, 20%; 0.015uF	21107-315	CRL	UK50-153		
C715,716	Tantalum, 35V, 10%; 1uF	21307-510	SPR	196D105X9035HA1	MANY	
C717	Monolythic Ceramic, 50V, 20%; 0.1uF	21123-410	SPR	1C25Z5U104M050B		
C718,719	Mica, 500V, 1%; 1000pF	21022-210	CD	CD19-FD102F03	SAN	
C720	Mica, 500V, 5%; 430pF	21024-143	CD	CD19-FD431J03	SAN	
C721	Mica, 500V, 2%; 39pF	21019-039	CD	CD15-ED390G03	SAN	
C722	Mica, 500V, 5%; 430pF	21024-143	CD	CD19-FD431J03	SAN	
C723	Ceramic Disc, 25V, 20%; 0.1uF	21106-410	CRL	UK25-104		
C724,725	Alum., Radial, 25V, -20% +100%; 100uF	21206-710	PAN	ECE-A1EV101S		
C726-729	Monolythic Ceramic, 50V, 20%; 0.1uF	21123-410	SPR	1C25Z5U104M050B		
Inductors						
L701	Inductor, RF Choke, 50mH	29504-350	MIL	70F502AF		
Integrated Circuits						
IC701	Linear, Single Opamp	24008-202	TI	LM318P	NAT	
IC702	Linear, Single Opamp	24014-202	SIG	NE5534N	TI	
IC703	Linear, Dual Opamp	24208-302	RCA	CA3280A		
IC704	Linear, Single Opamp	24007-202	AD	AD518N		
IC705	Linear, Single Opamp	24015-202	RAY	RC4131NB		
IC706	Linear, Single Opamp	24003-202	RCA	CA301AE	NAT,TI	
IC707,708	Special Function, Comparator	24701-302	NAT	LM711CN	RAY,TI	
IC709	Linear, Dual Opamp	24206-202	TI	TL072CP	NAT(LF353H)	
IC710	Linear, Single Opamp	24003-202	RCA	CA301AE	NAT,TI	
IC711	Linear, Single Opamp	24008-202	TI	LM318P	NAT	
IC712	Linear, Dual Opamp	24206-202	TI	TL072CP	NAT(LF353H)	
IC713	Digital, XOR Gate	24504-302	RCA	CD4030BE	SIG	
IC714	Digital, Nand Gate	24501-302	RCA	CD4011BE	MOT	
IC715-717	Optoisolator, NPN	25003-000	SIE	SFH-601-1		

FOOTNOTES:

(1) See last page for 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 8100A/1 — CARD #7
CAPACITORS thru INTEGRATED CIRCUITS

REF DES	DESCRIPTION	ORBAN P/N	VEN (1)	VENDOR P/N	ALTERNATE VENDORS(1)	NOTES
Resistors						
R701,702	Resistor Set, MF, 0.1% 4.99K	28520-001	ORB			
R708	Trimpot, Cermet, 20 Turn, 25K; 20%	20512-325	BEK	89PR25K	BRN	
R731	Trimpot, Cermet, 25K, "Pilot Inj"	20520-325	BEK	82PA25K		
R738	Trimpot, Cermet, 20 Turn, 25K; 20%	20512-325	BEK	89PR25K	BRN	
R769	Trimpot, Cermet, 20 Turn, 50K; 20%	20512-350	BEK	89PR50K	BRN	
R772	Trimpot, Cermet, 20 Turn, 1K; 20%	20512-210	BEK	89PR1K	BRN	
Switches						
S701	Switch, Rotary, Min., 2P3T	26201-000	STK	80-Series		Alt:Electroswitch
S702	Switch, Toggle, Min.	26037-009	CK	7101SYA		
Transistors						
Q701,702	Transistor, JFET/N	23403-101	NAT	J111	INS	
Q703,704	Transistor, Signal, NPN	23202-101	MOT	2N4400	FSC	
Q705,706	Transistor, JFET/N	23406-101	NAT	J113		
Q707,708	Transistor, JFET/N	23403-101	NAT	J111	INS	
Q709	Transistor, JFET/N	23406-101	NAT	J113		
Q710	Transistor, Signal, PNP	23001-101	MOT	2N4125	FSC	
Q711	Transistor, JFET/N	23403-101	NAT	J111	INS	
Q712	Transistor, Signal, PNP	23002-101	MOT	2N4402	FSC	

FOOTNOTES:
(1) See last page for 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 8100A/1 — CARD #7, Cont'd
RESISTORS thru TRANSISTORS

REF DES	DESCRIPTION	ORBAN P/N	VEN (1)	VENDOR P/N	ALTERNATE VENDORS(1)	NOTES
CARD #8/9						
Capacitors						
C801	Mica, 500V, 1%; 100pF	21018-110	CD	CD15-FD101F03	SAN	
C802	Mica, 500V, 1%; 1000pF	21022-210	CD	CD19-FD102F03	SAN	
C803	Mica, 500V, 1%; 160pF	21018-116	CD	CD15-FD161F03	SAN	
C804	Mica, 500V, 1%; 1000pF	21022-210	CD	CD19-FD102F03	SAN	
C805	Mica, 500V, 1%; 2400pF	21022-224	CD	CD19-FD242F03	SAN	
C806	Mica, 500V, 1%; 560pF	21022-156	CD	CD19-FD561F03	SAN	
C807	Mica, 500V, 1%; 2700pF	21022-227	CD	CD19-FD272F03	SAN	
C808-810	Mica, 500V, 1%; 1000pF	21022-210	CD	CD19-FD102F03	SAN	
C811	Mica, 500V, 1%; 220pF	21018-122	CD	CD15-FD221F03	SAN	
C812	Mica, 500V, 1%; 1000pF	21022-210	CD	CD19-FD102F03	SAN	
C813	Mica, 500V, 1%; 100pF	21018-110	CD	CD15-FD101F03	SAN	
C814	Mica, 500V, 1%; 220pF	21018-122	CD	CD15-FD221F03	SAN	
C815	Polystyrene, 50V, 2%; 0.022uF	21504-322	SPR	287P2232R5A3		
C816	Mica, 500V, 1%; 2700pF	21022-227	CD	CD19-FD272F03	SAN	
C817	Polystyrene, 50V, 2%; 0.0075uF	21504-275	SPR	287P7522R5A3		
C818	Mica, 500V, 1%; 2400pF	21022-224	CD	CD19-FD242F03	SAN	
C819,820	Mica, 500V, 1%; 1000pF	21022-210	CD	CD19-FD102F03	SAN	
C821	Met. Polycarb., 100V, 10%; 1.0uF	21604-510	ECI	652A1B105K	IMB	
C822,823	Mica, 500V, 1%; 1000pF	21022-210	CD	CD19-FD102F03	SAN	
C824	Mica, 500V, 1%; 41pF	21018-041	CD	CD15-ED410F03	SAN	
C825	Tantalum, 35V, 10%; 0.47uF	21307-447	SPR	196D474X9035HA1	MANY	
C826	Polyester, 100V, 10%; 0.0033uF	21401-233	SPR	225P33291WD3	PAN,PAK	
C827,828	Alum., Radial, 25V, -20% +100%; 100uF	21206-710	PAN	EVE-A1EV101S		
C829-832	Monolythic Ceramic, 50V, 20%; 0.1uF	21123-410	SPR	1C5Z5U104M050B	MANY	
C833	Tantalum, 35V, 10%; 1uF	21307-510	SPR	196D105X9035HA1	MANY	
C834,835	Aluminum Electrolytic, 50V, 100uF	21208-710	SPR	502D107F050DGIC	MANY	
C9xx	Subtract 100 and refer to C8xx series					
Diodes						
CR803-806	Diode, Signal, Hot Carrier	22102-001	HP	5082-2800		
CR9xx	Subtract 100 and refer to CR8xx series					
Inductors						
L801,802	Inductor, Variable	29702-004				
L803	Inductor, Variable	29702-003				
L804	Inductor, Variable	29702-002				
L805	Inductor, Variable	29701-002				
L9xx	Subtract 100 and refer to L8xx series					

FOOTNOTES:
(1) See last page for 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 8100A/1 — CARD #8/9
CAPACITORS thru INDUCTORS

REF DES	DESCRIPTION	ORBAN P/N	VEN (1)	VENDOR P/N	ALTERNATE VENDORS (1)	NOTES
Integrated Circuits						
IC801-806	Dual Opamp	24206-402	TI	TL072CJG	MOT	
IC807	Multiple Discrete	24406-302	RCA	CA3096AE		
IC808	Dual Opamp	24206-402	TI	TL072CJG	MOT	
IC9xx	Subtract 100 and refer to IC8xx series					
Modules						
A1	Module Assy, Phase Delay Network	30485-000	ORB			
A2	Module Assy, Distortion Cancel Module	30490-001	ORB			
CHASSIS						
Inductors						
L1,2	Inductor, RF Choke, 7uH	29501-004	OHM	Z-50	(2)	
Miscellaneous						
M1	Meter, Edge, 1mADC FS, 0-25dB	28009-104	EMI	132D5		900 Ohms
M2	Meter, Edge, 1mADC FS, 0-5dB	28009-105	EMI	132D5		900 Ohms
M3	Meter, Edge, 1mADC FS, 0-25dB	28009-104	EMI	132D5		900 Ohms
M4	Meter, Edge, 1mADC FS, 0-30dB	28009-102	EMI	132D5		900 Ohms
M5	Meter, VU, Brown/Tan	28002-007	DIX	330T	HOYT	
NONE	Connector, BNC	27101-000	AM	31-3376		
NONE	Connector, Card Edge, 22 Pos.	27035-004	SAE	SAC 22S/2-3	MANY	
CHASSIS (BACK PANEL)						
Capacitors						
C1-4	Ceramic Disc, 1KV, 10%; 0.0015uF	21112-215	CRL	DD-152		
Transistors						
Q101,102	Transistor, Power, NPN	23601-501	RCA	2N3055	FSC	
CHASSIS (FILTER BOX)						
Capacitors						
C1-14	Ceramic, Feed-thru, 1000pF	21118-210	ERE	2404-000-Series		Alt: Murata

FOOTNOTES:
(1) See last page for 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 8100A/1 -- CARD #8/9 Cont'd
CHASSIS, CHASSIS (BACK PANEL),
CHASSIS (FILTER BOX)

REF DES	DESCRIPTION	ORBAN P/N	VEN (1)	VENDOR P/N	ALTERNATE VENDORS (1)	NOTES
POWER SUPPLY AND REGULATOR BOARD						
Capacitors						
C101,102	Alum., Electrolytic, CG, 40V, 5000uF	21250-850	CD	FAH-5000-40-A2	MAL	
C103,104	Ceramic Disc, 50V, 20%; 0.05uF	21107-350	CRL	UK50-503		
C105-107	Ceramic, Feed-thru, 1000pF	21118-210	ERE	2404-000-Series		
C108	Tantalum, 10V, 10%; 33uF	21303-633	SPR	196D336X9010KE3	MANY	
C109	Mica, 500V, 5%; 470pF	21024-147	CD	CD19-FD471J03	SAN	
C110	Not Used					
C111,112	Alum., Radial, 50V, -20% +100%; 47uF	21208-647	SPR	502D476G050CD1C	PAN	
C113	Mica, 500V, 5%; 100pF	21020-110	CD	CD15-FD101J03	SAN	
C114	Polyester, 100V, 10%; 0.01uF	21401-310	SPR	225P10391WD3	PAN,PAK	
Diodes						
CR101-104	Diode, Rectifier, 400V, 3A	22203-400	MOT	MR504		
CR105,106	Diode, Rectifier, 400V, 1A	22201-400	MOT	1N4004	MANY	
Inductors						
L101,102	Inductor, RF Choke, 7uH	29501-004	OHM	Z-50	(2)	
Integrated Circuits						
IC101	D.C. Regulator	24301-302	NAT	LM723CN		
IC102	Linear, Single Opamp	24003-202	RCA	CA301AE	NAT,TI	
Miscellaneous						
F101	Fuse, 3AG, Slo-Blo, 1/2A	28004-150	LFE	313.500	BUS	(Use 1/4A Fuse for 230VAC Main)
F102,103	Fuse, Pico, 1A, Axial	28011-210	LFE		BUS	(On P.S. regulator board)
T101	Transformer, Power, 38VCT, 40VA	55002-000	ORB			
VR101,102	Diode, Zener, 16V, 5W, 5%	22005-160	MOT	1N5353B	MANY	
Resistors						
R103,104	Resistor, Wirewound, 2W, 0.62 OHM; 5%	20028-862	IRC	BWF Series		
R106	Trimpot, Cermet, 18 Turn, 500 OHM; 20%	20508-150	BEK	68XR500	BRN	
R108,109	Resistor Set, MF, .25% 20.0K	28521-001	ORB			
Switches						
S101	Switch, Toggle, SPST, AC Power	26002-001	CH	8280K21C		
S102	Switch, Slide, AC Line	26140-000	SW	EPSI-SLI		
Transistors						
Q103,104	Transistor, Signal, PNP	23002-101	MOT	2N4402	FSC	

FOOTNOTES:
(1) See last page for 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 8100A/1 -- POWER SUPPLY
CAPACITORS thru TRANSISTORS

REF DES	DESCRIPTION	ORBAN P/N	VEN (1)	VENDOR P/N	ALTERNATE VENDORS (1)	NOTES
FRONT PANEL						
Diodes						
CR1	LED, Red T-1 3/4	25103-000	GI	MV-5053	MANY	
CR2,3	LED, Yellow T-1 3/4	25105-000	GI	MV-5353	MANY	
CR4	LED, Green T-1 3/4	25104-000	GI	MV-5253	MANY	
Miscellaneous						
M5	Meter, VU, Brown/Tan	28002-007	DIX	330T	HOYT	
Switches						
S1	Switch, Toggle, Min., SPDT	26037-005	CK	7105P3		
INPUT FILTER BOARD						
Inductors						
L1-4	Inductor, RF Choke, 1.2mH	29503-000	MIL	73F123AF		
L5,6	Inductor, RF Choke, 7uH	29501-004	OHM	Z-50	(2)	
METER RESISTOR BOARD						
Resistors						
R5,6	Trimpot, Cermet, 1 Turn, 1K; 20%	20509-210	BEK	72XR1K	BRN	
Switches						
S2	Switch, Rotary, 1P12T, NS	26078-306	CTS	212 SERIES		
OTHER						
Miscellaneous						
NONE	Line Cord, IEC	28102-002	BEL	17500	MANY	
NONE	PCB Extender Board Assy	30705-000	ORB			

FOOTNOTES:
(1) See last page for 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 8100A/1 -- FRONT PANEL,
INPUT FILTER BD, METER RESISTOR BD

Vendor Codes

AB Allen-Bradley Co.
1201 South Second Street
Milwaukee, WI 53204

AD Analog Devices, Inc.
Route 1, Industrial Park
P.O. Box 280
Norwood, MA 02062

AM Amphenol North America
An Allied Company
2122 York Road
Oak Brook, IL 60521

BEK Beckman Instruments, Inc.
Helipot Division
2500 Harbor Blvd.
Fullerton, CA 92634

BEL Belden Corporation
Electronic Division
Richmond, IN 47374

BRN Bourns, Inc.
Trimpot Products Division
1200 Columbia Avenue
Riverside, CA 92507

BUS Bussmann Manufacturing Div.
McGraw-Edison Company
P.O. Box 14460
St. Louis, MO 63178

CD Cornell-Dubilier Elec.
150 Avenue "L"
Newark, NJ 07101

CH Cutler-Hammer
Landmark Office Center
2081 Landings Drive
Mountain View, CA 94043

CK C & K Components, Inc.
15 Riverdale Avenue
Newton, MA 02158

COR Corcom, Inc.
1600 Winchester Road
Libertyville, IL 60048

CRL Centralab, Inc.
A North American Company
5757 North Green Bay Ave.
Milwaukee, WI 53201

CTS CTS Corporation
905 North West Blvd.
Elkhart, IN 46514

DIX Dixson, Inc.
287 Twenty Seven Road
Grand Junction, CO 81501

ECI Electrocube
1710 South Del Mar Avenue
San Gabriel, CA 91776

EMI Emico
123 North Main Street
Dublin, PA 18917

ERE Erie Tech. Products, Inc.
644 West Twelfth Street
Erie, PA 16512

EXR Exar Integrated Systems, Inc.
P.O. Box 62229
Sunnyvale, CA 94088

FSC Fairchild Camera & Instr. Corp.
464 Ellis Street
Mountain View, CA 94042

GI General Instruments
Optoelectronics Div.
3400 Hillview Avenue
Palo Alto, CA 94304

HP Hewlet-Packard Corporation
1501 Page Mill Road
Palo Alto, CA 94304

INS Intersil, Inc.
10710 North Tantau Avenue
Cupertino, CA 95014

IRC TRW/IRC Resistors
401 North Broad Street
Philadelphia, PA 19108

LFE Littelfuse
A Subsidiary of Tracor
P.O. Box 2345
Des Plaines, IL 60016

MAL Mallory Timers Company
Emhart Electrical/Electronic Gr.
3029 East Washington Street
Indianapolis, IN 46206

ME Mepco/Electra, Inc.
Columbia Road
Morristown, NJ 07960

MIL J.W. Miller Division
Bell Industries
19070 Reyes Avenue
P.O. Box 5825
Compton, CA 90221

MOT Motorola, Inc.
P.O. Box 20912
Phoenix, AZ 85036

NAT National Semiconductor Corp.
2900 Semiconductor Drive
Santa Clara, CA 95051

NOB Noble
Teikoku Tsushin Kogyo Co. Ltd.
335, Kariyado, Nakahara-ku
Kawasaki 211, JAPAN

OHM Ohmite Manufacturing Company
A North American Philips Co.
3601 Howard Street
Skokie, IL 60076

ORB Orban Associates, Inc.
645 Bryant Street
San Francisco, CA 94107

PAK Paktron
Div. of Illinois Tool Works Inc.
900 Follin Lane, S.E.
Vienna, VA 22180

PAN Panasonic
Electronic Components Div.
P.O. Box 1503
Seacaucus, NJ 07094

RAY Raytheon Semiconductor Div.
350 Ellis Street
Mountain View, CA 94042

RCA RCA Solid State Division
Route 202
Somerville, NJ 08876

SAE Stanford Applied Eng.
340 Martin Avenue
Santa Clara, CA 95050

SAN Sangamo
Capacitor Division
P.O. Box 128
Pickens, SC 29671

SCH ITT Schadow, Inc.
8081 Wallace Road
Eden Prairie, MN 55343

SIE Siemens
Components Division
186 Wood Avenue, South
Iselin, NJ 08830

SIG Signetics Corporation
A Sub. of US Philips Corp.
P.O. Box 9052
Sunnyvale, CA 94086

SPR Sprague Electric Co.
125 Marshall Street
North Adams, MA 01247

STK Stackpole Components Co.
P.O. Box 14466
Raleigh, NC 27620

SYL Sylvania Conn. Prod. Op.
GTE Products Corp.
Box 29
Titusville, PA 16354

TI Texas Instruments
P.O. Box 225012
Dallas, TX 75265

TRW TRW Electronic Components
Connector Division
1501 Morse Avenue
Elk Grove Vlg., IL 60007-57

WES Westlake
5334 Sterling Ctr Drive
Westlake Village, CA 91361

WIM WIMA
P.O. Box 2345
Augusta-Anlage 56
D-6800 Mannheim 1
GERMANY

APPENDIX K:

Audio Quality Considerations in FM Plants

It is an unhappy fact that any audio processor degrades audio quality to achieve loudness, consistency, and absolute peak control. OPTIMOD-FM achieves an exceptionally advantageous tradeoff between loudness and quality degradation. One improvement that OPTIMOD-FM can make in audio quality is to automatically correct tonal balance when operated in "independent" mode. However, the HF limiting and peak control can only degrade quality, and it is only possible to justify this sacrifice when we consider that it buys us approximately ten extra dB of loudness (compared with unprocessed audio).

Computer people have long used the phrase: "Garbage in/garbage out." This is especially true of audio processing, wnere the rule is: "Garbage in/more garbage out!" The audio quality delivered by OPTIMOD-FM is entirely dependent upon the audio quality of the signal that OPTIMOD-FM receives at its input terminals. If the audio is immaculate, then the processed signal will sound excellent -- even with "heavy" processing. But a distorted input signal will be further degraded by the processing, and may well end up sounding offensive and unlistenable.

The purpose of this Appendix is to provide some hints on how to achieve that immaculate audio at the OPTIMOD-FM input terminals. Such a discussion could easily fill an entire book. We can only hope in a limited space to cover the most important points.

Achieving consistent state-of-the-art audio quality in FM broadcast is a very difficult task, requiring considerable skill, professionalism, and great dedication. But, as certain stations with stand-out audio have shown, it *is* possible!

DISK REPRODUCTION

Most radio programming still comes from phonograph records, either directly, or through tape dubs. We will address the problems of tape below; the current discussion centers on accurately retrieving as much information as possible from the grooves.

Disk is intrinsically a very high quality medium, and much effort has been expended by consumer manufacturers in developing audiophile cartridges, pickup arms, turntables, and phono preamps of highest quality. Unfortunately, much of this equipment is insufficiently mechanically rugged to withstand the pounding that it typically receives in day-to-day broadcast operations. At this writing, there are only two reasonably high quality cartridges made in the USA which are generally accepted to be sufficiently rugged to withstand professional use: the Stanton 681 series, and the Shure SG-39 series. Although rugged and reliable, neither produces the same cleanliness and transparency as the best audiophile cartridges. This phono cartridge dilemma is the prime argument for transferring all disk material to tape in the production studio, and playing only tape on the air. In this way, it is possible (with care) to use state-of-the-art cartridges, arms, and turntables in the dubbing process since requirements for mechanical ruggedness are relaxed. In addition, the problem of record wear is eliminated. However, maintaining tape equipment such that it causes no noticeable quality degradation is by no means easy, and the smaller station (particularly one without a full-time engineer) may well be able to achieve superior quality by playing disks directly on the air.

The following should be carefully considered when choosing and installing disk playback equipment.

1. The cartridge must be scrupulously aligned. When viewed from the front, the stylus must be absolutely perpendicular to the disc, or separation will suffer. The cartridge must be parallel to the headshell, or a fixed tracking error will be introduced. Overhang should be set as accurately as possible (± 1/16"), and vertical tracking angle should be set at 20 degrees (by adjusting arm height).

2. The tracking force must be correctly adjusted. Usually, better sound results from tracking close to the maximum force recommended by the cartridge manufacturer. If the cartridge has a built-in brush, don't forget to compensate for it by adding more tracking force according to manufacturer's recommendations.

3. Anti-skating force must be correctly adjusted. The accuracy of the anti-skating force calibration on many pickup arms is questionable. The best way to adjust anti-skating is to obtain a test record with an extremely high-level lateral cut (some IM test records are suitable). Connect the left channel output of the turntable preamp to the horizontal input of an oscilloscope, and the right channel preamp output to the vertical input. Operate the scope in the "X/Y" mode, such that a straight line is visible at a 45 degree angle. If the cartridge mistracks asymmetrically (indicating incorrect anti-skating compensation), then the scope trace will be "bent" at its ends. If this happens, adjust the anti-skating until the trace is a straight line, indicating symmetrical clipping.

   It is important to note that in live-disk operations, use of anti-skating may increase the incidence of the arm's sticking in damaged grooves instead of jumping over the bad spots. Increasing tracking force by approximately 15% has the same effect on distortion as applying antiskating, and in live-disk operations, the former expedient may be preferred.

4. A modern direct-drive turntable must be used. None of the older-design "professional broadcast" turntables have low enough rumble to be inaudible on the air. These old puck-, belt-, or gear-driven turntables might as well be thrown away! Don't even hand them down to your AM operation -- modern multiband processing will cause the rumble to be audible even on automobile AM radios.

5. Proper turntable mounting is crucial to avoid picking up footsteps or other building vibrations, and to avoid acoustic feedback from monitor speakers which will cause muddiness and severe loss of definition. The turntable is best mounted on a vibration isolator, which in turn is placed on a non-resonant pedestal mounted as solidly as possible to the building (preferably to a concrete slab).

6. Until recently, most "professional" phono preamps were seriously deficient compared to the best "audiophile" preamps. Fortunately, this situation has changed very recently, and a small number of high quality professional preamps are available (mostly from small manufacturers). A good preamp is characterized by extremely accurate RIAA equalization, high input overload point (better than 100mV @1kHz), low noise (optimized for the reactive source impedance of a real cartridge), low distortion (particularly CCIF difference-frequency IM), load

resistance and capacitance adjustable for a given cartridge and cable capacitance, and effective RFI suppression.

After the preamp has been chosen and installed, the entire disk playback system should be checked with a reliable test record for compliance with the RIAA equalization curve. IF YOU WISH TO EQUALIZE THE STATION'S AIR SOUND TO PRODUCE A CERTAIN "SOUND SIGNATURE" ON THE AIR, THE PHONO PREAMP IS NOT THE PLACE TO DO IT. Some of the better preamps have adjustable equalizers to compensate for frequency response aberrations in phono cartridges. Since deviations of 0.5dB can be detected by critical listeners, ultra-accurate equalization of the entire cartridge/preamp system is most worthwhile.

The load capacitance and resistance should be adjusted according to the cartridge manufacturer's recommendations, taking into account the capacitance of the cables. If a separate equalizer control is not available, load capacitance and resistance may be trimmed to obtain flattest frequency response. Failure to do this can result in frequency response errors as great as 10dB in the 10-15kHz region!

The final step in adjusting the preamp is to accurately set the channel balance on the basis of a test record, and finally to set gain such that output clipping is avoided on any record. If you need to operate the preamp close to its maximum output level due to the system gain structure, then put a scope on the output of the preamp, and play a loud passage from an "audiophile" or "direct-to-disk" record. Set the gain so that at least 6dB peak headroom is left between the loudest part of the record and peak clipping in the preamp.

7. It is our opinion that the single most significant cause of distorted on-air sound is worn phono styli. Styli deteriorate sonically before any degradation at all is visible under a microscope, because the cause of the degradation is usually deterioration of the mechanical damping and centering system in the stylus (or actual bending of the stylus shank) rather than diamond wear. This deterioration is primarily caused by back-cueing, although rough handling will always make a stylus die before its time.

   Styli used on-air in 24-hour service should be changed every two weeks as a matter of course -- damn the expense! D.J.'s and the engineering staff should listen constantly for audible deterioration of on-air quality, and should be particularly sensitive to distortion caused by defective styli. Such styli should immediately be replaced when problems are detected. One engineer we know immediately destroys old styli upon replacing them so that he is not tempted to keep a stock of old, deteriorated, but usable-looking styli!

   It is important to maintain a stock of new spare styli for such emergencies or for routine periodic replacement. There is no better example of false economy than waiting until styli fail before ordering new ones, or hanging onto worn-out styli until they literally collapse! Note also that smog- and smoke-laden air may seriously contaminate and damage shank mounting and damping material. Some care should be used to seal your stock of new styli to prevent such damage.

8. Several impulse noise reduction systems that effectively reduce the effects of "tics" and "pops" in disk reproduction without significantly veiling audio quality have been available. These are particularly useful in live-disk operations where disks tend to become worn and damaged. The Burwen TNE-7000 is very effective

in removing small "tics"; the SAE 5000 works well on larger scratches. Both devices can be connected in series at the output of the phono preamps to virtually eliminate the effects of disk damage. They must not be used elsewhere in the chain (such as in the program line) because the supersonic energy necessary to trigger their control circuitry will probably be rolled-off.

TAPE

Despite its undeniable convenience, the tape cartridge (even at the current state-of-the-art) is still inferior to reel-to-reel in almost every performance aspect. Unlike the sometimes mystical sonic differences attributed to preamps and amplifiers, performance differences between cart and reel are readily measured, and include differences in frequency response, noise, high frequency headroom, wow and flutter, and particularly azimuth and interchannel phasing stability.

Sum-and-Difference Recording: Because it is vital in stereo FM broadcast to maintain mono compatibility, "sum and difference" recording is preferred in either reel or cart operations. This means that the mono sum signal (L+R) is recorded on one track, and the stereo difference signal (L-R) is recorded on the other track. A matrix circuit restores L and R upon playback. In this system, interchannel phase errors cause frequency-dependent stereo-field localization errors rather than deterioration of the frequency response of the mono sum.

Audio Time Base Correction: Time base correction technology, which is in widespread use in video recording, has recently been developed for use in audio applications. These devices use reference delays or tones to detect and correct phase shift errors. Time base correction is preferable to sum-and-difference recording because it actually corrects phase errors, while sum-and-difference just limits the impact of the errors, producing stereo-field localization errors to avoid mono frequency response deterioration. In addition, recording in L/R form (as is done when audio time base correction is used) can improve signal-to-noise ratio by up to 6dB over that obtainable with sum-and-difference recording.

Cheap Tape: Cheap tape, whether reel or cart, is a temptation to be avoided. Cheap tape may suffer from any (or all) of the following problems:

1. Sloppy slitting, causing the tape to weave across the heads, or, if too wide, to slowly cut away your tape guides;
2. Poor signal-to-noise ratio;
3. Poor high frequency response and/or high frequency headroom;
4. Inconsistency in sensitivity, bias requirements, or record EQ requirements from reel to reel, or even within a reel;
5. Splices within a reel;
6. Oxide shedding, causing severe tape machine cleaning and maintenance problems;
7. Squealing due to inadequate lubrication.

High-line name-brand tape is a good investment. It provides high initial quality, and guarantees that recordings will be resistant to wear and deterioration as they are played. Whatever your choice of tape, you should standardize on a single brand and type to assure consistency and to minimize tape machine alignment problems. Some of the most highly regarded tapes in current use include Agfa PEM468, Ampex 406, Ampex 456, BASF SPR-50 LHL, EMI 861, Fuji type FB, Maxell UD-XL, TDK GX, Scotch (3M) 206, Scotch 250, Scotch 226, and Sony SLH11.

It goes almost without saying that cheap carts are to be avoided like the plague, considering that even the best carts provide barely adequate quality. Since carts will interact with different transport designs in different ways, one of the best ways to choose a cartridge brand is to make extensive test on the cart machines you have in-house, and to choose the brand exhibiting the best interchannel phase stability and lowest wow and flutter with your particular machines.

Tape Speed: If all aspects of the disk-to-tape transfer receive scrupulous care, then the quality difference between 15ips (38cm/sec) and 7.5ips (19cm/sec) recording is easily audible. 15ips has far superior high frequency headroom. The effects of dropouts and tape irregularity are also reduced, and the effects of interchannel phase shifts are halved. In addition, a playback machine can deteriorate (due to oxide buildup on the heads or incorrect azimuth) far more severely at 15ips than at 7.5ips before an audible change occurs in audio quality.

Nevertheless, because of playback time limitations at 15ips, most stations operate at 7.5ips. (Many carts will not operate reliably at 15ips, and are subject to jamming and other problems.) This speed seems to be the lowest that is practical for use in day-to-day broadcast practice. While 3.75ips can produce good results under carefully controlled conditions, there are few operations which can keep playback machines well enough maintained to obtain consistent high quality 3.75ips playback day-in, day-out. In addition, use of 3.75ips results in another jump in sensitivity to bad tape, high frequency saturation, and interchannel phase shifts.

Some are now promoting the use of cassettes as a serious broadcast program source. We feel that cassettes' low speed, tiny track width, sensitivity to dirt and tape defects, and severe high frequency headroom problems make such proposals totally impractical where consistent quality is demanded.

Use Of Noise Reduction: In order to reduce or avoid tape hiss, we recommend use of a compandor-type (encode/decode) noise reduction system in all tape operations. Dolby^R B-type noise reduction is probably most suitable. Its 9dB noise reduction is quite sufficient to move tape hiss below the level of record surface noise if high-quality low-noise tape is used. In exchange for its modest amount of noise reduction, Dolby B seems quite free of audible side-effects. (Dolby A-type noise reduction is even more effective, and is equally free of side-effects -- but it may strain your budget.)

Bear in mind that to achieve accurate Dolby tracking, record and playback levels must be matched better than 2dB. The Dolby tone should be faithfully recorded at the head of all reel-to-reel tapes, and level matching should be frequently checked. There should be no problem with level matching if tape machines are aligned weekly, as level standardization is part of this procedure. If a different type of tape is put in service, record machines must be immediately aligned to the new tape before any recordings are made.

In our opinion, all "single-ended" (i.e., dynamic noise filter) noise reduction systems cause totally unacceptable audible side-effects (principally program-dependent noise modulation) when used with music, and should never be used "on-line". They may have their place in the production studio, but even there they must be used extremely judiciously, with their operation constantly monitored by the station's "golden ears". Some possible applications include noise reduction of outside production work, and, when placed after the microphone preamp, reduction of ambient noise in the control room or production studio.

Tape Recorder Maintenance: Regular maintenance of magnetic tape recorders is of vital importance in achieving consistently high-quality sound. Maintenance of tape machines requires expertise and experience, and we can only lightly touch upon certain points.

1. Heads and guides should be cleaned every four hours of operation.

2. Tradition has it that machines should be demagnetized every eight hours. In our experience, magnetization is usually not a problem in playback-only machines in fixed locations. A magnetometer with a ±5 Gauss scale (R.B. Annis Co., Indianapolis, Indiana) should be used to periodically check for permanent magnetization of heads and guides. You will soon obtain experience on how long it takes for your machines in your environment to pick up enough permanent magnetization to be harmful. You may well find that this never happens with playback machines. (Record machines must be watched much more carefully.)

3. Deterioration of tape machine performance is usually gradual rather than catastrophic. It is therefore necessary to measure the performance of an on-air machine weekly with standard test tapes, and to take whatever corrective action is necessary if the machine is not meeting specifications. (Test tapes are manufactured by laboratories such as MRL, 229 Polaris Ave. #4, Mountain View, California 94043; and by STL, 26120 Eden Landing Rd. #5, Hayward, California 94545.)

4. Weekly maintenance should include measurement of flutter, using a flutter meter and high-quality test tape. Deterioration in flutter performance is often an early warning of impending mechanical failure. Spectrum analysis of the flutter can usually relate the flutter to a single rotating component. Deterioration in flutter performance can, at very least, indicate that adjustment of reel tension, capstan tension, reel alignment, or other mechanical parameter is required.

5. Weekly maintenance should also include measuring frequency response and interchannel phase shifts with a high-quality alignment tape. These measurements, which are expedited by the use of special swept-frequency or pink noise tapes available from some manufacturers (like MRL), provide an early indication of loss of correct head azimuth, or of head wear. (The swept tapes are used with an oscilloscope; the pink noise tapes with a third-octave real time analyzer.)

   If a head becomes worn, do not try to compensate by adjusting the playback equalizer. This will increase noise unacceptably, and will also introduce frequency response anomalies because the equalizer cannot accurately compensate for the shape of the rolloff caused by a worn head. Instead, the head must be replaced or lapped.

The reader should be particularly aware that alignment tapes wear out. With wear, the output at 15kHz may be reduced by several dB. If you have many tape machines to maintain, it is usually more economical to make your own "secondary standard" alignment tapes, and use these for weekly maintenance, while reserving your standard alignment tape for reference use.

Do-it-yourself alignment tapes are best made with the traditional series of discrete tones. First, align the playback section of the master recorder on which the home-made alignment tape is to be recorded, using a fresh standard alignment tape.

Coarse Azimuth: First, obtain a coarse adjustment by peaking the level of the 15kHz tone on the alignment tape, making sure that you have found the major peak. (There will be several minor peaks, many dB down. You will not encounter these unless the head is totally out of adjustment.)

Reproduce Equalization: Run the alignment tape and adjust the reproduce equalizers for flat high frequency response, and for low frequency response which corresponds to the "fringing table" supplied with your alignment tape. The "fringing" effect appears below 500Hz, and will ordinarily result in an apparent bass boost of 2-3dB at 100Hz due to the fact that the alignment tape was recorded full-track and is being reproduced on a half-track head. (Fine azimuth adjustment won't work correctly if the playback equalizers are not set for identical frequency response, since non-identical frequency response will also result in non-identical phase response!)

Fine Azimuth: This adjustment is ideally made with a full-track mono pink noise tape and a real-time analyzer. If this instrumentation is available, sum the two channels together, connect the sum to the real-time analyzer, and adjust the azimuth for maximum high frequency response.

Other possibilities include observing the mono sum of a swept-frequency tape and maximizing its high frequency response, or aligning by ear by listening to the mono sum of the announcer's voice on the standard alignment tape, and adjusting for crispest sound. (The azimuth on the announcer's voice will be just as accurate as the rest of the tape.)

If the traditional Lissajous pattern is used, use several frequencies, and adjust for minimum differential phase at all frequencies. Using just one frequency (say 15kHz) can give totally erroneous results.

Calibration: After the azimuth has been carefully adjusted and the playback equalizer adjusted for maximally flat response from the standard alignment tape, write down the actual VU meter reading produced at each frequency on the spot-frequency standard alignment tape. Use the fringing table by subtracting the compensation from the readings you have just made. You will use the compensated readings below when you are recording your tape, since you are recording in half-track stereo instead of full-track mono.

Record Azimuth: Record your alignment tape using an audio oscillator. First, adjust the azimuth of the record head, by observing the mono sum from the playback head and exciting the record amplifier with pink noise, swept tones, or wide-range music (for "by-ear" adjustment). If you use a Lissajous pattern, be sure to use several frequencies as mentioned above.

Recording The Tape: You are then ready to record the spot frequencies on tape. Set the VU meter to "playback" and observe the reading as each tone is recorded. Adjust the tape recorder RECORD GAIN immediately after each frequency is switched until the VU meter reads the same as it did when you were playing the standard alignment tape. Your home-made tape should have an error of only 0.5dB or so if you follow these steps carefully.

REMEMBER THAT YOUR HOME-MADE TAPE WILL DETERIORATE WITH USE -- check it frequently against your "standard" reference tape.

The home-made tape is not suitable for critical azimuth adjustments. These should be made using the methods described above, employing a test tape recorded with a full-track head. Even if you happen to have an old full-track mono machine around, getting the azimuth exactly right is not practical, and a standard commercial alignment tape should be used for azimuth adjustments. Because ordinary wear does not affect the azimuth properties of the tape, it should have a very long life if properly stored.

ALL TEST TAPES SHOULD BE STORED "TAILS-OUT" (UNDER CONTROLLED TENSION) IN A TEMPERATURE AND HUMIDITY-CONTROLLED ENVIRONMENT. NEITHER EDGE OF THE TAPE PACK SHOULD TOUCH THE REEL FLANGES. This cannot be achieved unless the tape is wound onto the storage reel in normal PLAY mode, not in fast forward or rewind!

6. After the reproduce section of the tape machine is aligned, record alignment should be checked and adjusted as necessary. This involves setting record head azimuth, bias, equalization, and meter calibrations according to manufacturer's recommendations. We recommend that tape machines be adjusted so that +4dBm in and out corresponds to "0 VU" on the tape recorder's meter, and also to Dolby level and "Standard Operating Level". This is ordinarily 185 nW/m for "standard" tape and 250 nW/m for "high output" tape.

   Current practice calls for adjustment of bias by the "high-frequency overbias" method rather than by the "peak bias with 15 mil wavelength" method, as was formerly standard practice. Briefly, bias is adjusted by recording a 1 mil wavelength on tape (5Khz @7.5ips), and increasing bias until maximum output is produced from tape. Bias is then further increased until the output has decreased by a fixed amount -- usually 1.5 to 3dB. The correct amount of decrease is a function of both tape formulation and the width of the gap in the record head. The tape manufacturer's datasheet should be consulted.

7. In addition to the steps listed above, the manufacturer's recommendations should always be followed regarding maintenance. Most tape machines require periodic lubrication, and checking of reel holdback tensions and brake adjustments. With time, critical bearings will wear out in the motors and elsewhere. Failures here are usually indicated by incorrect speed, increased flutter, and/or audible increases in the mechanical noise made by the tape machine in operation. Use only lubricants and parts specified by the manufacturer.

8. Last, but by no means least -- KEEP IT CLEAN. Dust is a great destroyer of precision mechanical parts (and cigarette smoke is none too good, either). In addition to keeping dust away from the heads and guides, periodically clean the rest of the machine with a vacuum cleaner (suction mode, please!), or soft clean paint brush.

Cartridge Machine Maintenance: The general comments above apply to cart machines as well. However, these devices have their own set of idiosyncrasies, largely because much of the tape guidance system is located in the cartridge, and is thus subject to the vagaries of the construction of the individual carts.

1. Because the lubricated tape deposits lubricant on pressure rollers and guides, frequent cleaning is advisable to assure lowest wow and flutter and to prevent possible cart jams. Cleaning should be performed as often as experience proves necessary. (Interestingly, because of the nature of the tape lubricant, it does not tend to deposit on head gaps and head cleaning is rarely required.).

2. Even with the best maintenance, interchannel phase shifts in conventionally-designed cart machines will usually prove troublesome. Check head alignment frequently. In addition, different brands of cart will show significant differences in phase stability in a given brand of machine. Run tests on various brands of cart, and standardize on the one offering best phase stability.

3. Because of the vast differences in design between manufacturers, it is difficult to provide much more specific advice. Precisely follow manufacturer's instructions regarding periodic maintenance, mechanical alignment, tensioning, lubrication, etc.

4. Many early (and some not-so-early) cart machine designs were saddled with completely inadequate electronics. Considerable improvement can be achieved in some of these machines by electronics modifications. Check electronics for record-amplifier headroom (be sure the amplifier can completely saturate the tape before it clips); record amplifier noise and equalization (some record amplifiers can actually contribute enough noise to dominate the overall noise performance of the machine); playback preamp noise and compliance with NAB equalization; power supply regulation, noise, and ripple; and line amplifier headroom. Check the alignment of the record level meter. (In order to improve apparent signal-to-noise ratio at the expense of distortion, some meters are calibrated so that "0" corresponds to significantly more than 1% third-harmonic distortion!)

   Probably the most universal problem is inadequate record amplifier headroom. In many cases, it is possible to improve the situation by increasing the operating current in the final record-head driver transistor close to its power dissipation limits. This is usually done by decreasing the value of emitter (and sometimes collector) resistors while observing the collector voltage to make sure that it stays at roughly half the power supply voltage under quiescent conditions, and adjusting the bias network as necessary if it doesn't.

COMPACT DISC

More and more stations are broadcasting program material recorded on compact discs. The compact disc (CD) approaches the "ideal" program source recording medium. Compact discs do not wear out, they cue easily, and the recordings are generally of much higher quality than that obtainable with the turntables and tape machines found in broadcast studios. Currently, the main disadvantages of using CD's for program material are the limited (but rapidly growing) number of recordings available and the inability to make recordings in-house of material originated by the station.

SYSTEM CONSIDERATIONS

Headroom: Other than bad styli, the single most common cause of distorted air sound is probably clipping. The gain and overload point of every electronic component in the station must be critically reviewed to make sure that components are not being operated so that they introduce clipping distortion or excessive noise.

VU meters are worthless for checking peak levels. Even Peak Program Meters are insufficiently fast to indicate clipping of momentary peaks, as their integration time is approximately 10ms. While the design of PPM's makes them excellent for monitoring operating levels in media with limited dynamic range (like magnetic tape) where small amounts of peak clipping are acceptable to achieve optimum signal-to-noise ratio, there is no excuse for any clipping at all in the purely electronic part of the signal path, since low noise and wide dynamic range are readily achieved with good design. For this reason, the peak levels should be monitored with a true peak-reading meter or with an oscilloscope, and gains adjusted so that peak clipping never occurs under any reasonable operating conditions (including sloppy D.J. gain riding!).

In the case of older equipment with very "soft" clipping characteristics, it may be impossible to see a well-defined clipping point on a scope. Or worse, audible distortion may occur many dB below the apparent clip point. In this case, the best thing to do is to determine the peak level producing 1% THD, and to arbitrarily call this level the clipping level. The scope can be calibrated to this 1% THD point, and headroom measurements can then be made.

The canny engineer will also be aware that certain system components (like microphone or phono preamps) have absolute input overload points. Difficulties often arise when gain controls are placed after early active stages, because the input stages can be overloaded without clipping the output. Many broadcast mike preamps are notorious for low input overload points, and can be easily clipped by high-output mikes and/or screaming D.J.'s. Similar problems can occur inside consoles if gain structures and operating points have been poorly chosen by the console designer, or if "Master" gain controls are operated with unusually large amounts of attenuation.

When operating with nominal line levels of +4 or +8dBm, the absolute clipping point of the line amplifier becomes critical. The headroom between nominal line level and the amplifier clipping point should be greater than 16dB. This implies that a line amplifier for a +4dBm line should clip at +20dBm or above, and that an amplifier for a +8dBm line should clip at +24dBm or above. In particular, it means that IC equipment (which almost always clips at +20dBm or so unless transformer-coupled) is not suited for use with +8dBm lines. +4dBm lines have become standard in the recording industry, and are preferred for all new studio construction (recording or broadcast) because of their compatibility with IC opamp operating levels.

The following components of a typical FM audio plant should be checked for operating level and headroom:

1. Phono preamps
2. Tape and cart preamps
3. Record amplifiers in tape machines
4. Microphone preamps
5. Console summing amplifiers
6. Line amplifiers in consoles, tape recorders, etc.
7. Distribution amplifiers (if used)
8. Signal processing devices such as equalizers
9. Specialized communications devices, including remote broadcast links and telephone interface devices
10. STL's, whether land-line or microwave

Voice/Music Balance: The VU meter is very deceptive in indicating voice/music balance. The most artistically pleasing balance between voice and music usually results from peaking voice 4-6dB lower than music on the console VU meter. If heavy processing is being used, this factor may have to be increased further.

Following this practice will also help reduce the possibility of clipping voice (which is much more sensitive to clipping distortion than most music) in the electronics.

It is sometimes difficult to train operators to follow this practice. If the console has (or can be modified to have) separate summing amplifiers for live voice and music, then the correction factor is easily automated by building a separate summing amplifier (using a single IC opamp) to drive the VU meter, and summing the output of the voice summing amplifier into the VU amplifier with greater gain than the output of the music summing amplifier.

Electronic Quality: FM has certain limitations which prevent it from ever becoming a transmission medium totally satisfying to the "golden-eared" audiophiles. These limitations must be considered when discussing the quality requirements for FM electronics. The problems in disk and tape reproduction discussed above are much grosser by comparison, and the subtle masking of basic FM transmission limitations is irrelevant to those discussions.

There are three fundamental limitations. The first is multipath distortion. In most locations, a certain amount of multipath is unavoidable, and this is exacerbated by the inability of many apartment-dwellers to use rotor-mounted directional antennas. The second is the bandwidth limitation of the FM stereo multiplex system, which is theoretically 19kHz, and practically limited by the characteristics of "real-world" filters to between 15 and 17kHz. The third is the IF bandwidth of receivers necessary to eliminate adjacent and alternate channel interference. This effect can be clearly heard by using a tuner with switch-selectable IF bandwidth. Most stations cannot be received in "wide" mode because of interference. But if the station is reasonably clean (well within the practical limitations of current broadcast practice) and free from multipath, then a clearly audible reduction in high-frequency "grit" is heard when switching from "normal" to "wide" mode.

These limitations have considerable significance in gauging cost-effectiveness in current broadcast design practice. Most of the older-design broadcast electronic equipment (whether tube or transistor) is measurably and audibly inferior to properly-designed modern equipment. This is primarily due to a design philosophy which stressed ruggedness and RFI immunity over distortion and noise, and to the excessive use of inferior transformers. Frequency response was purposely rolled-off at the extremes of the audio range to make the equipment more RFI-immune. Cascading equipment of such design tends to increase both distortion and audible frequency response rolloffs to unacceptable levels.

Modern design practice emphasizes the use of high-slewrate, low-noise IC operational amplifiers such as the Signetics NE5534 family and the Texas Instruments TL070 family (both of which are used internally in the 8100A/1). While some designers insist that only discrete designs can provide ultimate quality, the performance of the best of the current IC's is so good that discrete designs are just not cost-effective for broadcast applications when the basic FM quality limitations are considered.

It has recently been discovered that capacitors have a subtle, but discernible effect upon sonic quality. Polar capacitors such as tantalums and aluminum electrolytics behave very differently from ideal capacitors. In particular, their very high dissipation factor and dielectric absorption can cause significant deterioration of complex musical waveforms. Ceramic capacitors have problems of similar severity. Polyester film capacitors can cause a similar, although less severe, effect when audio is passed through them.

For this reason, DC-coupling between stages (which is easy with opamps operated from dual positive and negative power supplies) is best. Coupling capacitors should be used only as absolutely necessary (to keep DC offsets out of faders, thus preventing "scratchiness", for example). If capacitors must be used, film types such as polystyrene, polypropylene, or polycarbonate are preferred.

If it is impractical to eliminate capacitors or to change capacitor types, don't be too concerned. It is probable that other quality-limiting factors will largely mask the capacitor-induced degradations.

It goes almost without saying that the number of transformers in the audio path should be kept to an absolute minimum. Transformers are sometimes the only practical way to break ground loops and/or eliminate RFI. If a transformer is necessary, use a high-quality device like those designed by Deane Jensen and manufactured by Reichenbach Engineering, North Hollywood, California.

In summary, the path to highest quality is that which is closest to a straight wire. More is <u>not</u> better: every device removed from the audio path will yield an improvement in clarity, transparency, and fidelity. Use only the <u>minimum</u> number of amplifiers, capacitors, and transformers. Never leave, say, a line amplifier or compressor in "test" mode on line because it seems too much trouble to take it out. Small stations often sound dramatically superior on the air to their "big-time" rivals because the small station has a <u>simple</u> audio path, while the big-budget "big-timer" has thrown everything but the kitchen sink on line. The more equipment the station has (or can afford), the more restraint and self-discipline is required to keep the audio path simple and clean. Every amplifier, resistor, capacitor, transformer, switch contact, patch-bay contact, etc., is a potential source of audio degradation. Corrosion of patch-bay contacts and switches can be <u>particularly</u> troublesome, and the distortion caused by these problems is by no means subtle!

Any P.D. who boasts of his station's $20,000 worth of "enhancement" equipment should first be taken to a physician who can clean the wax from his ears, then forced to swear that he is not under the influence of any suspicious substances, and finally placed gently but firmly in front of a high-quality monitor system and shown exactly the sort of degradation that $20,000 worth of "enhancement" causes!

There is no situation where an old '70's cliche is more valid: Less is more.

PRODUCTION PRACTICES

General: The role of the production studio varies widely from station to station. If the production studio is used only for creation of spots, promos, ID's, etc., then quality requirements are considerably relaxed compared to a production studio in which programming is transferred from disk to tape or cart. Our discussion centers on the latter case.

Choosing The Monitor Loudspeakers: The production studio monitor system is the quality reference for all production work, and thus the air sound of the station. Considerable care in choice of equipment and its adjustment is necessary to assure a monitor sound that can be relied upon.

The loudspeakers are the single most important influence on quality. They should be chosen to complement room acoustics. In general, a production studio is fairly small, and "bookshelf"-sized speakers must be used because of space limitations.

It is desirable to assess the effect of equalization or other "sweetening" on small speakers to make sure that excessive bass or high-frequency boost has not been introduced. Such equalization errors can sound spectacular on big, wide-range speakers, while sounding terrible on small speakers with limited frequency response and power-handling capacity.

The recording industry has standardized on the Auratone Model 5C "Super Sound Cube" as a "small-speaker reference". We recommend that every production studio be equipped with a pair of these speakers, and that they be regularly used to assure the production operator that his work will sound good on small table and car radios.

The main loudspeakers should be chosen for high power-handling capacity, low distortion, high reliability and long-term stability, controlled dispersion (omnidirectional speakers are not recommended), good tone burst response at all frequencies, lack of cabinet diffraction, relatively flat axial and omnidirectional frequency response from 40-15,000Hz, and physical alignment of drivers such that if all drivers are excited simultaneously by an impulse, then the resulting waveforms arrive at the listener's ears simultaneously (sometimes called "time alignment").

Many high-quality speakers are available from American and Japanese manufacturers. We recommend reading loudspeaker reviews in back issues of Studio Sound magazine and consulting with your local professional audio dealer. It is important that you corroborate the opinions of reviewers and dealers with your own listening tests. Your local discount hi-fi emporium is not the place to do this; the "high-line" audio salons tend to have more knowledgeable and helpful salespeople as well as better listening conditions.

Loudspeaker Location: The bass response of the speakers is strongly affected by their location in the room. Bass is weakest if the speaker is mounted in free air, away from any walls, and strongest when it is mounted in a corner. The corner location is to be avoided because it tends to excite standing waves, and the best location is probably against a wall at least 18" from any junction between walls. If the bass response is weak at this location because the speaker was designed for wall-junction mounting, it can be corrected by equalization (see below).

It is important that the loudspeakers be mounted so as to avoid acoustic feedback into the turntable, as this can produce a severe loss of definition and "muddy" sound.

Loudspeaker Equalization: The performance of any loudspeaker is strongly influenced by its mounting location and room acoustics. Provided that room acoustics are good, the third-octave real-time analyzer provides an extremely useful means of measuring any frequency response problems intrinsic to the loudspeaker, and of partially indicating problems due to loudspeaker placement and room acoustics.

By their nature, the third-octave measurements combine the effects of direct and reflected sound and may be misleading if room acoustics are unfavorable. Problems can include severe standing waves, reverberation time which is not well-behaved as a function of frequency, insufficient number of "normal modes" (Eigenmodes), lack of physical symmetry, and a number of other problems which, if discussed in adequate detail, would require a whole book!

There is a technique of measuring the loudspeaker/room interface called Time-Delay Spectrometry™ which provides much more information about acoustic problems than does the third-octave real-time analyzer. A certain number of sound contractors are now licensed to practice this technique. The technique is primarily used in "tuning" recording studio control rooms, and the cost may be prohibitive for a small or medium-sized station, particularly if measurements reveal that acoustics can only be improved by major modifications to the room.

Thus, the third-octave real-time analyzer is probably the best compromise for the typical radio station. If the station does not have a third-octave real-time analyzer and pink noise source, these can usually be rented from a local sound contractor or instrument rental house. To obtain meaningful results from the analyzer, the calibrated microphone which comes with the analyzer should be placed where the production engineer's ears would ordinarily be located. Each loudspeaker should be excited in turn with pink noise, and the acoustic response observed on the analyzer.

We recommend the Orban 674A dual-channel quasi-parametric equalizer as a monitor equalizer to "tune" the monitor system. The equalizer should be adjusted per manufacturer's instructions to obtain a real-time analyzer readout which is flat to 5kHz, and which rolls off at 3dB/octave thereafter. (A truly "flat" response is is not employed in typical loudspeakers, and will make most records sound unnaturally bright and noisy.)

If the two channels of the equalizer must be adjusted differently to obtain the desired response from the left and right channels, this indicates room acoustic problems or poorly-matched loudspeakers. The match is easy to check; just physically substitute one loudspeaker for the other, and see if the analyzer reads the same.

It is very revealing to move the microphone over a space of two feet or so while watching the analyzer to see how much the response changes. If the change is significant, then room acoustic problems or very poorly-controlled loudspeaker dispersion is indicated. In this case, you should measure the response at several positions and average. (There are devices called "microphone multiplexers" available to do this automatically. They require the use of several microphones, and they average the microphone outputs in a phase-insensitive way.)

Although it is permissible to adjust left and right equalizers differently below 200Hz, they should be set close to identically above 200Hz (to preserve stereo imaging), even if this results in less than ideal curves indicated on the third-octave analyzer. In this case, the limitations of this analysis technique (as described above) are coming into play.

Other Production Equipment: The general comments above on disk reproduction, tape, and electronic quality apply equally to the production studio. It is preferable to install "audiophile-quality" phono cartridges, arms, and turntables here, and to make sure that one person has responsibility for production quality and for making sure that the record playing equipment is not abused. The use of a single production director will also help achieve a consistent air sound, which is an important contribution to the "big-time" sound desired by many stations.

Although some people still swear by certain "classic" vacuum-tube power amplifiers (notably those manufactured by Marantz and McIntosh), the best choice for a monitor amplifier is probably a medium-power (100 watts/channel or so) solid-state amplifier with a good record of reliability in professional applications. Popular brands include Crown and BGW. (It may be tempting to dust off an old Gates or RCA power amplifier and place it in service as an economy measure. Don't. And don't use the monitor amplifiers built into your console, either!)

Production Practices: The following represents our opinions on production practices. We are aware that certain stations operate under substantially different philosophies. But we feel that the recommendations below are rational, and offer the best hope of achieving consistently high quality.

1. Audio processing should not be applied in the production studio. The 8100A/1 provides all the processing necessary, with a remarkable lack of audible side-effects. Any further compression is not only undesirable, but is likely to be very audible. If the production compressor has a slow attack time (thus producing overshoots which can activate gain reduction in the 8100A/1), it will probably "fight" with the 8100A/1, thus yielding air sound which is substantially worse than one might expect given the individual "sounds" of two units.

2. Substandard recordings may be "sweetened" with equalization to achieve tonal balance more typical of the best current product. However, excessive treble boost (to achieve a certain "sound signature" for the station) must be avoided if 7.5 ips tape is used because the tape is subject to high frequency saturation due to the high frequency boost applied by the recorder's equalization network. Substantially more freedom can be obtained by using an Orban 418A Compressor/Limiter/HF Limiter between the output of the production console and the input of the tape recorder or cart recorder. By adjusting the gain on the 418A such that broadband gain reduction never occurs when the console VU meters are peaking normally,

only high frequency control will ordinarily occur, thus preventing high frequency tape saturation without adding unwanted broadband compression. However, the subtle broadband compressor will come into play to prevent tape overload if the console output level is peaked too high.

3. A compandor-type noise reduction system should be used on all taped material (see section on **TAPE**, above).

4. Even greater care than that employed in maintaining on-air equipment should be used in production studio maintenance, since quality loss here will appear on the air again and again. The production director should be acutely sensitized to audible quality degradation, and should immediately inform the engineering staff of any problems detected by ear.

5. Ideally, tape machines with noisy motors should be installed in alcoves under soffits, and there surrounded by acoustic treatment to prevent motor noise from leaking into the production microphone. In the real world of budget limitations, this is often not possible. Even in an untreated room, it is possible to use a directional microphone such as a figure-of-eight, and to place the noisy machine on the "dead" axis of the microphone. Choosing the frequency response of the microphone to avoid exaggerating low frequencies will help. In particularly difficult cases, a noise gate or expander can be used after the microphone preamp to shut off the microphone except during actual speech.

6. Audio processing can be profitably applied to the microphone channel to give the sound more punch. Suitable equalization may include gentle low- and high-frequency boosts to "crispen" sound, aid intelligibility, and add a "big-time" quality to the announcer. (But beware using too much bass boost -- it can degrade intelligibility.) Effects like telephone and transistor radio can be achieved with equalization, too. Orban manufactures a line of parametric equalizers (the 622A, 622B, 672A, and 674A) which are ideal for this work. The instruction manuals supplied with these units provide many hints on how to use them to achieve the effects often desired in broadcast production.

   The punch of production material can often be enhanced by tasteful application of compression to the microphone chain. But avoid using an excessive amount of gain reduction and excessively fast release time, lest room noise and announcer breath sounds be exaggerated to grotesque levels. (This problem can be minimized if the compressor has a built-in expander or noise gate function.)

   The close-miking customary in the production studio can exaggerate voice sibilance. In addition, many women's voices are sibilant enough to cause unpleasant effects. If high frequency equalization and/or compression are applied, sibilance will be further exaggerated.

   These problems can be totally controlled by means of a dedicated de-esser like the Orban 536A Dynamic Sibilance Controller located after all other processing in the microphone chain. (The 536A can also be used with the on-air microphone. Its built-in microphone preamp makes installation straightforward.)

   Orban's "Studio Optimod" Model 424A, which combines several audio processing functions, is an excellent all-in-one vocal processor. The controls for 424A's compressor, limiter, and de-esser interact to enhance ease of operation without limiting versatility.

SUMMARY

These comments only touch the surface of the techniques necessary to achieve audio quality in FM broadcast comparable to a typical "high-end" home stereo system. Because of the built-in limitations of the FM medium, audio quality equal to that delivered by "state-of-the-art" audiophile equipment from top-quality disks or master tapes cannot be achieved, even if the signal entering the input terminals of the 8100A/1 lives up to that quality level. This fact provides a useful guide to evaluating the cost-effectiveness of any equipment and/or techniques which are proposed to improve quality. In particular, it leads to the conclusion that today's high-quality IC opamps are ideally suited as amplification elements in broadcast. Compromises in disk playback and tape are far more likely to be audible on the air, and extreme care must be used.

Maintaining a high level of on-air quality is a very difficult task, requiring constant dedication and cooperation between air talent and engineering. With the constantly increasing quality of home receivers and stereo gear, the results of such dedication and cooperation are more and more easily perceived by the radio audience. One suspects that in the future, FM will have to deliver a state-of-the-art signal in order to compete successfully with the many other program sources vying for audience attention, including CD's, videodiscs, digital audio, subscription television, direct satellite broadcast, and who knows how many others!

The future belongs to the quality-conscious.

APPENDIX L:

Specifications

Frequency Response (System in PROOF mode)

Follows standard 75us preemphasis curve ±0.75dB, 50-15,000 Hz. 50us preemphasis available on special order. All preemphasis networks include a fourth-order lowpass filter and fourth-order phase corrector prior to the high-frequency limiter and clipper to prevent these elements from processing out-of-band program material and to minimize overshoot, thus minimizing the amount of high-frequency limiting and clipping.

Input Conditioning

Highpass Filter: Third-order Chebychev with 30Hz cutoff and 0.5dB passband ripple. Down 0.5dB at 30Hz; 10.5dB at 20Hz; 31.5dB at 10Hz. Protects against infrasonic destabilization of certain exciters' AFC's, as well as infrasonic gain modulation in the compressor.
Phase Scrambler: Allpass network makes peaks more symmetrical to best utilize the symmetrical peak overload characteristics of the FM medium.

Noise

-75dB below 100% modulation, 50-15,000 Hz maximum; -81dB typical.

Total System Distortion (PROOF Mode; 100% Modulation)

Less than 0.05% THD, 50-15,000Hz (0.02% typical); less than 0.05% SMPTE Intermodulation Distortion (60/7000Hz; 4:1).

"Master" Band Compressor Characteristics

Attack Time: approximately 1ms.
Release Time: program-controlled -- varies according to program dynamics and amount of gain reduction (see text). Process can be scaled fast or slow by means of continuously variable RELEASE TIME control. Employs delayed release for distortion reduction.
Total Harmonic Distortion (measured at VCA output, OPERATE mode, RELEASE TIME control centered): Less than 0.1%, 50-15,000Hz, 0-25dB gain reduction
Available Gain Reduction: 25dB
Metering: Three dB-linear edgewise-reading gain reduction meters --
  MASTER is true peak-reading with electronic acceleration and peak-hold (0-25dB).
  COMPRESSOR indicates slow compression component of gain reduction (0-25dB).
  LIMITER indicates fast peak limiting component of gain reduction (0-5dB).
Gain Control Element: True VCA. Proprietary Class-A design eliminates crossover notch distortion, modulation noise, and slewrate limiting found in competitive Class-AB designs.

"Bass" Band Compressor Characteristics

Attack Time: program-controlled; not adjustable.
Release Time: program-controlled; not adjustable. Incorporates delayed-release distortion reduction.
Total Harmonic Distortion (at VCA output, OPERATE mode): Less than 0.1% THD, 50-200Hz, 0-30dB gain reduction.
Available Gain Reduction: 30dB.
Metering: single dB-linear edgewise-reading gain reduction meter (0-30dB).
Gain Reduction Element: Proprietary Class-A true VCA.
Crosscoupling (U.S. patent #4,249,042): Enables gain of "Bass" band to track gain of "Master" band to any degree, from identical tracking to fully independent operation. Adjustable with BASS COUPLING control.

Crossover Characteristics

Control: 6dB/octave @200Hz.
Program: 12dB/octave @200Hz in unique "distributed crossover" configuration (U.S. patent #4,249,042).

High Frequency Limiter Characteristics

Attack Time: approximately 5ms.
Release Time: approximately 20ms. Delayed release included for distortion reduction.
Mode: Left and right channels operate independently to avoid high frequencies in one channel causing audible timbre modulation of opposite channel.
Control Element: Junction FET.
Metering: Two LED's indicate hf limiting in L and R channels.
Threshold of HF Limiting: User-adjustable over 3dB range to meet format requirements.

FM "Smart Clipper" Output Processor Characteristics

Nominal Bandwidth: 15.4kHz.
Distortion Cancellation: Clipping distortion (below overshoot compensator threshold) cancelled better than 30dB (40dB typical), 0-2200 Hz (U.S. patent #4,208,548).
Delay Correction: Fourth-order allpass.
Amount of Clipping: User-adjustable over 6dB range to match format requirements.

Frequency-Contoured Sidechain (FCS) Overshoot Compensator Characteristics (U.S. patent #4,460,871)

System Overshoot: The FCS circuit is best thought of as a "bandlimited safety clipper". It operates like a hard clipper, but does not produce out-of-band frequency components as a simple hard clipper would. Because the audio processing will sometimes limit steady-state material with high average energy (like sinewaves) or with very little high-frequency energy to levels below the threshold of clipping, it is difficult to state a clear and meaningful specification for the system overshoot performance of the FCS circuit.

The FCS circuit is followed by a safety clipper. The overshoot specification could be slightly improved if this safety clipper were set up to clip more frequently. However, the system is aligned at the factory such that the safety clipper is almost never active, thus fully preserving the bandlimiting provided by the FCS circuit. With this safety clipper alignment, the peak modulation will be controlled ±3.5% on arbitrary waveforms clipped to any degree by the FCS circuit (acting as a bandlimited safety clipper); peak modulation will not exceed this level on other material. With typical program material, peak modulation uncertainty is less than 2%.

Sinewave Modulation Ability: 93% modulation (i.e., 0.6dB below maximum overshoot level) at all sinewave frequencies, assuming sinewaves are applied to FCS input.

Dynamic Separation: better than 45dB.

Difference-Frequency Intermodulation: FCS circuit causes no more audible IM (such as sibilance splatter) than would a simple hard clipper clipping to the same depth. The entire 8100A/1 processing system is specifically configured to prevent the FCS circuit from audibly degrading the difference-frequency distortion-cancellation properties of the earlier FM "Smart Clipper".

System Separation

Greater than 45dB, 50-15,000Hz; 60dB typical.

Stereo Generator Characteristics

Crosstalk (Main Channel-to-Subchannel, or Subchannel-to-Main Channel): better than -40dB, 50-15,000Hz as measured at input terminals to stereo generator, or using internal crosstalk test mode which applies left-channel audio to either main or sub stereo generator inputs. Crosstalk representing distortion components (non-linear crosstalk) typically better than -80dB as measured on a baseband spectrum analyzer.

38kHz Subcarrier Suppression: Greater than 40dB below 100% modulation; 60dB typical.

Suppression of 76kHz and its Sidebands: Greater than 70dB below 100% modulation.

Pilot Frequency: 19.000kHz ±2Hz.

Pilot Injection Adjustment Range: Less than 8% to greater than 10% modulation.

Input

Impedance: greater than 10K ohms, electronically balanced by means of true instrumentation amplifier. Requires balanced source.
Common Mode Rejection: Greater than 60dB @60Hz.
Sensitivity: -10dBm produces 10dB "Master" Band gain reduction @1kHz. Removal of internal 20dB pad permits -30dBm to produce same effect.
Connector: Cinch-Jones 140-style barrier strip (#5 screw).

Composite (Baseband) Output

Source Impedance: 470 ohms, independent of OUTPUT ATTEN setting, unbalanced.
Level: variable 0 to greater than 4V p-p by means of 15-turn OUTPUT ATTEN control.
Connector: Type BNC held floating over chassis ground to permit interface to various exciters without need for wideband transformer for ground loop suppression. RF suppressed.
Recommended Maximum Cable Length: 6ft (1.8m) RG-58A/U.

Auxiliary Input/Output (for Test use only)

Provides L and R lowpass filter output or L and R stereo generator input depending upon setting of rear-apron NORMAL/TEST switch. Connectors are RCA phono-type, unbalanced. Stereo generator requires approx. 3V RMS for 100% modulation, unbalanced, with source impedance of test generator less than 50 ohms.

Operating Controls

VU Meter Selector switches ASA-standard VU meter to read:

- L or R Input Level
- L or R Compressor Output
- L or R Filter Out
- L-R Level
- 19kHz Oscillator Level
- 38kHz PLL Control Voltage
- 38kHz AGC Control Voltage
- ±15 V Power Supply Voltages

Stereo/Mono Mode Switch: Momentary front panel switch may be conveniently strapped for either left or right mono by means of a plug-in internal jumper. Mode may be remote-controlled by application of 6-24 V AC or DC pulses to appropriate rear terminals. Terminals are optically isolated, and may be floated ±50 V above ground. Three pairs of remote terminals will select either left or right audio inputs in mono mode, or stereo. Another internal jumper selects which of the three modes will be entered on powerup.

Setup Controls (front-panel, behind lockable swing-down door -- see Fig. 4-5)

Compressor:
- Left and Right Input Attenuators
- "Master" Band Release Time
- Gate Threshold
- Bass Coupling
- Clipping
- High-Frequency Limiter Threshold

Stereo Generator:
- Pilot Injection
- Pilot Phase
- L-R Gain (Separation)
- Pilot ON/OFF Switch
- NORMAL/MAIN-TO-SUB/SUB-TO-MAIN Crosstalk Test Switch (see text)

General:
- Output Attenuator
- PROOF/OPERATE Switches
  (to defeat gain reduction, hf limiting, clipping, and gating)
- Power ON/OFF
- 115V/230V Selector Switch

Power Requirements

115/230VAC, ±15%, 50-60Hz, approx. 19VA.
IEC mains connector with detachable 3-wire "U-Ground" power cord supplied.
Leakage to chassis less than 0.5uA. AC is RF-suppressed.

Dimensions

19" (48.3cm) W x 7" (17.8cm) H x 12.5" (31.2cm) D -- 4 rack units.

Environmental

Operating Temperature Range: 0-50° C (32-122° F).
Humidity: 0-95% R.H., non-condensing.

Warranty

One year, parts and labor. Subject to limitations set forth in our Standard Warranty.

All specifications subject to change without notice.

Appendix M:
Functions of Jumpers on PC Cards

Several cards used in OPTIMOD-FM Model 8100A/1 are also used in other Orban products. These cards have jumpers which determine their mode of operation. This appendix provides a card-by-card quick reference to jumper functions and normal 8100A/1 jumper positions. See assembly drawings in **Appendix J** for jumper locations and diagrams.

Card #3/4: The jumpers on these cards determine the gains of the 20dB pads ahead of the input differential amplifiers. They should be set according to the nominal levels of the lines driving the OPTIMOD-FM. (Shipped with pads IN.)

Card #5: Jumper A converts the Master Release Time module from its normal timing mode to a slow averaging mode for use with the Model 8100A/XT Six-Band Limiter Accessory Chassis. Unless the 8100A/XT is installed, jumper A should be set to NORMAL mode (as shipped).

Jumper B determines the threshold of compression of the Master band control circuitry. Unless the 8100A/XT is installed, both links should be set to NORMAL mode (as shipped).

Jumper C determines the attack time of the Master band. Unless the 8100A/XT is installed, both links should be set to NORMAL mode (as shipped).

Card #6: Jumper A should always be in the 8100A,8180A position.

Jumper positions B and C are used to route the outputs of Card #6. When the two links are mounted in the NORMAL B position, the outputs are sent to the 8100A/XT through Accessory Port #2. When the links are mounted in the NORMAL C position (as shipped), the outputs are sent to 8100A/1 Cards #8 and #9. Note that NORMAL B orientation is not the same as NORMAL C orientation. The REVERSE B and C positions reverse the left and right channel outputs of Card #6, which may be useful for fault diagnosis.

Card #7: The Powerup Mode jumper determines whether the stereo generator comes up in STEREO (as shipped), MONO LEFT, or MONO RIGHT mode when AC power is applied.

The Mono Mode jumper selects whether MONO LEFT (as shipped) or MONO RIGHT mode is entered when the front-panel STEREO/MONO switch is set to MONO.

Card #8/9: Jumper A should always be in the 8100A,8180A position.

Jumper B should be in the NORMAL position (as shipped) unless the 8100A/XT is installed, and then it should be in the ACCESSORY CHASSIS position. The ACCESSORY CHASSIS position allows TP2 to be used as an input to the distortion-cancel sidechain filter. (TP2 is available at Accessory Port #2.)

USER FEEDBACK FORM

USE THIS FORM TO PROVIDE US YOUR COMMENTS ON:

- * Potential Improvements To The Product
- * Corrections Or Improvements To The Manual (Include Relevant Page #'s)

Second Fold

Cut along this line

Cut or copy the form. No postage necessary if mailed in the U.S.A.

First Fold

Orban may use or distribute any of the information you supply in any way it believes appropriate without any obligation whatever. We will not use your name for advertising or promotion.

Thank you for your input.

Model # 8100A/1

Serial # ____

OPTIONAL INFORMATION:

Name ____

Organization ____

Address ____

City, State, Country ____

Mail code ____

Telephone/Telex ____

To request a reply, check here [ ]

Date ____

TAPE, DO NOT STAPLE TAPE, DO NOT STAPLE

NO POSTAGE
NECESSARY
IF MAILED
IN THE
UNITED STATES

BUSINESS REPLY MAIL
First Class Permit No. 19935 San Francisco Calif. 94107

Postage will be paid by:

Orban Associates Inc.
645 Bryant Street
San Francisco, CA 94107
U.S.A.

TAPE, DO NOT STAPLE TAPE, DO NOT STAPLE