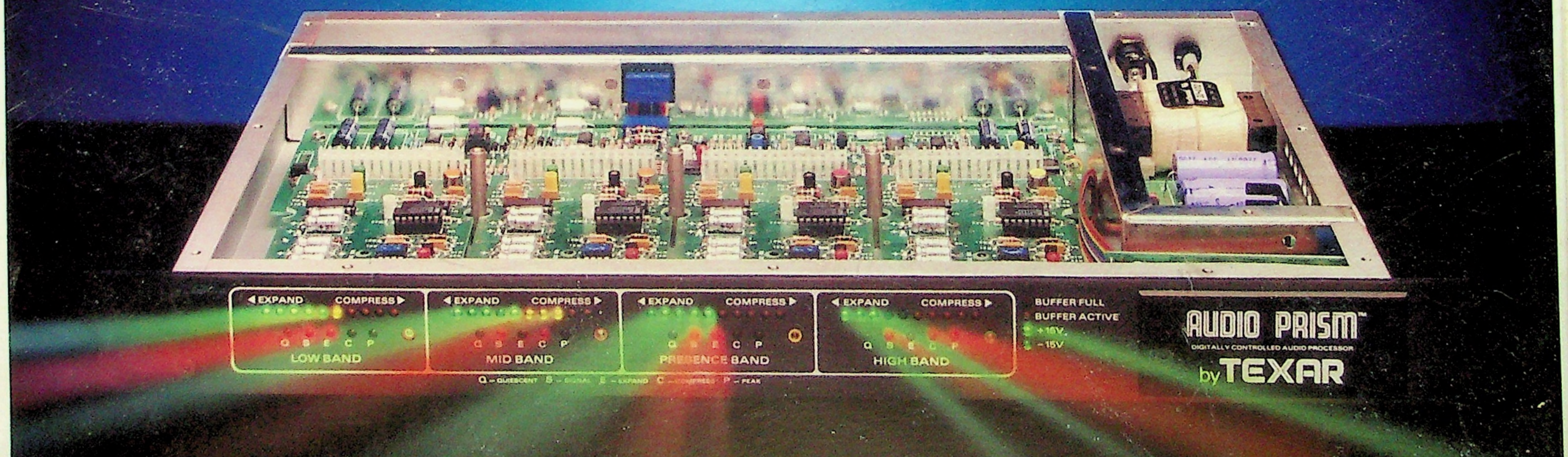
TEXAR

AUDIO PRISM™

DIGITALLY CONTROLLED AUDIO PROCESSOR



USERS' MANUAL

CONGRATULATIONS!

You have just purchased the most advanced broadcast audio processor which money can buy! The TEXAR AUDIO PRISM™ has been the audio processor of choice for the #1, #2 and #3 rated stations in America's largest market, New York City, ... for four Arbitrons in a row.*

DIGITAL CONTROL allows the AUDIO PRISM to deliver high modulation power for maximum signal range while producing the clarity to keep listeners tuned-in for hour after hour.

The enclosed unit(s) include several recent improvements in the design of the AUDIO PRISM. The sound of the unit has not changed, but several convenience features have been added. As a result, the operation of some of the controls has changed. These changes are detailed on the following eight colored pages. If you have prior experience with the AUDIO PRISM (Serial Number 504 or lower), these pages will bring you up to date without having to re-read the whole manual. If you have no prior experience with the AUDIO PRISM, the new features are explained completely in the manual which follows, and you may discard the colored pages now.

We want you to be happy with the enclosed unit(s). Our technical support people stand ready to serve you. Please read the manual completely before calling with a question, as many potential questions are answered in the manual. However, if after reading the manual, you still have questions, please call immediately. We'll do everything we can to help.

In addition, it is feedback from users in the field like yourself which allows us to continue improving our equipment and the manuals which accompany it. If you have suggestions for how a design change would make the equipment more useful, or how additional information would make the manual more understandable, we would like to hear from you.

Best regards and thank you!

Glen Clark
President
TEXAR Inc.

* Summer '85, Fall '85, Winter '86 and Spring '86 Arbitron Ratings, New York MSA, Total Persons 12+ Share, Mon-Sun, 6A-12M. (Used with permission.)

CHANGES NOTICE

Many improvements are incorporated into AUDIO PRISM's with Serial Numbers greater than 505. Most of these changes were to improve operator convenience or serviceability. These changes will not affect the sound of the unit. However, they will affect some aspects of the adjustment procedure. Most noteably, to produce the same type and degree of processing as on prior units, the rotational position of most of the front panel controls on these later units will be different. Further, the operation of what was formerly the "PROOF" function has been changed, and is now called the "BYPASS" function.

The MB-1 Motherboard has been replaced by the MB-2 Motherboard. The most apparent change is the addition of keyed socket Y302, power and signal connections for the optional PR-1 Phase Rotator. The PR-1 option is recommended for all applications, AM and FM. If other equipment in the audio chain, such as the Orban Optimod 8100*, contains an additional phase rotator, the later unit should be bypassed or removed. The proper placement for the phase rotator in any audio processing system is prior to the first variable-gain stage. The procedure for bypassing the phase rotators in the Optimod 8100 is contained in the AUDIO PRISM USERS' MANUAL on Page 25.

If the PR-1 option is not used in the AUDIO PRISM, pins 4 and 5 of Y302 are jumpered to complete the signal path.

It is possible to retrofit the PR-1 Phase Rotator into earlier AUDIO PRISM's which used the MB-1 motherboard. It will require razor-blading a single trace on the MB-1. Consult the factory for details.

Additional improvements of the MB-2 include improved trace layouts in critical areas, resulting in improved signal-to-noise performance.

Lastly, R308 and R309 on the motherboard have been reduced from 100K to 33K. This reduces the as-shipped-from-the-factory input gain to accept nominal input levels between 0 dBm and +12 dBm. Prior units typically had an input range of from -15 dBm to 0 dBm.

Users requiring additional input gain may replace R308 and R309 with a pair of 100K resistors to obtain the previous, higher, input gain.

(CONTINUED ON NEXT PAGE)

* OPTIMOD 8100 is a registered trademark of Orban Associates, Inc., San Francisco, California.

The CX-1 Control Board has been replaced by the CX-2 Control Board. Four significant changes have been implemented.

1) The electrical range of adjustment of all eight controls has been limited. As an example, the DENSITY control on prior units had the capability to vary the voltage on the density buss from 0.0 volts to 5.0 volts. Voltages below approximately 2.5 volts do not produce a useful mode of operation of the AUDIO PRISM and are not desirable. At full counter-clockwise rotation, this control now causes about 2.5 volts to be placed on the density buss. In addition to making undesirable settings inaccessible, this makes the action of the control within the desired range more gradual, permitting more accurate settings.

The range of the GATE control has been similarly limited to restrict the voltage on the gate buss to no more than 2 volts.

Similar changes in the operation of the INPUT GAIN, OUTPUT GAIN, and the four MIX LEVEL controls prevent them from being turned fully off.

The net result is that, while units with Serial Numbers greater than 505 can be adjusted to sound identical to earlier units, the physical position of the controls will be different. The MIX LEVEL controls should be adjusted to produce the desired voltages on the front panel test points. The GATE and DENSITY controls should be adjusted to produce the desired voltages on TP306 and TP307 respectively.

** NOTE **

The voltages on the front panel test points are <u>AC</u> and the reference levels given in APPENDIX A are in <u>dBm</u> (not volts). The voltages on TP306 and TP307 are <u>DC</u>.

** NOTE **

The reference levels in APPENDIX A were originally intended for the Simpson Model 260 Volt-Ohmmeter. As they were empirical data, identical results could not be insured when using other types of meters. In fact, erroneous results had been obtained when using other types of meters. Tests have shown that results identical to those using the Simpson 260 are obtained when using the POTOMAC INSTRUMENTS AA-51, Audio Analyzer. The AA-51 is now approved for adjusting AUDIO PRISM's in accordance with APPENDIX A.

Remember that the AA-51 is an AC voltmeter only. It has no facility to measure DC. If it is desired to set the MIX LEVELS and GATE and DENSITY controls during the same maintenance session, it will be necessary to have on hand a separate meter for the DC readings. While digital read-out meters cannot be used to make the AC measurements, due to the rhythmic nature of the pink noise, either analog or digital instruments may be used for the DC readings.

2) The PINK NOISE switch now automatically forces the GATE buss to 0 volts and the DENSITY buss to 5 volts. To insure uniform results in the set-up procedure, it is necessary for the density buss voltage to equal the compression reference buss voltage (permissible window equals zero) and for all four bands to be gated-on. This was accomplished on previous units by turning the DENSITY control fully clockwise and the GATE control fully counter-clockwise during the set-up procedure, even though these positions were not the same positions used during operation.

That procedure had two drawbacks. Users would sometimes forget this step, resulting in improper adjustment of the AUDIO PRISM. Secondly, it required disturbing an adjustment which may have been arrived at after considerable effort. Unless one documented the voltages at TP306 and TP307, one could later reset these voltages only to the accuracy permitted by turning the controls back to their original physical positions.

Changing the position of the GATE and DENSITY controls during the set-up procedure is no longer necessary, as the PINK NOISE switch removes these controls from the circuit and forces the buss voltages to the desired levels. When the PINK NOISE switch is turned OFF, the buss voltages are again controlled by the front panel controls and return to their previous levels.

3) The former PROOF mode has been changed in function and is now called the BYPASS mode.

The original AUDIO PRISM (units using the CX-1 PC board) PROOF mode was unlike that of most other discriminate processors. The PROOF mode froze the gain of all processors at their nominal levels, but kept all normal signal paths in the circuit. This was in contrast to the more common practice of driving the output amplifier directly from the input amplifier, bypassing all frequency-dependent circuits and VCA's. The underlying logic of the PROOF mode assumed that many circuit malfunctions could go undetected if they were not in the signal path when equipment performance measurements were performed.

Several years of experience with the AUDIO PRISM design have shown no known cases of internal circuits whose deteriorated performance would have been detected by this facility. Over the same period, many instances have occurred of users misconnecting the patch panel while bypassing the AUDIO PRISM's for set-up purposes. Some users have inadvertently interrupted the program. Others have even gotten unintended sources, including pink noise, on the air.

Further, many stations, including one major New York City FM, do not even have patch panels in their program lines, making the PINK NOISE set-up procedure almost impossible for them.

With this as background, the PROOF function has been changed to the more conventional BYPASS function, effective with the change to the CX-2 circuit board. The bandpass filters and the M-101 processor cards are now removed from the circuit when S102 is in the ON position. This permits the AUDIO PRISM to perform two functions simultaneously: 1) adjust the spectral MIX LEVEL controls under pink noise conditions, and 2) serve as a line amp, maintaining the normal program circuit from the studio to the limiting device which follows the AUDIO PRISM.

The gain of the BYPASS function is variable, allowing the user to adjust the BYPASS output level to be equal in amplitude (but not in density) to the normal output level. This gain is controlled by R115. This control is mounted behind a small unlabeled hole, located directly above the left threaded mounting for the escutcheon cover plate. Due to physical constraints, this control is recessed slightly more than an inch, making it not on an even plane with the rest of the front panel controls. Use a thin flat-blade screwdriver to access this control, such as the Xcelite "Greenie" which was included with the AUDIO PRISM. While the control is not easily visible, a few seconds of experimentation will usually allow the user to seat the screwdriver in the adjustment slot. This control is a multi-turn potentiometer.

When in the BYPASS mode, only the INPUT GAIN control and R115 will have any effect on the output signal. The OUTPUT GAIN control is out of the circuit.

If your AUDIO PRISM is equipped with the EAGLE Intelligent Clipper, the EAGLE is also removed from the signal path in the BYPASS mode.

4) A partial facility is implemented to allow "post-mix voice injection". Some stations prefer to process their live announcer mike separately from their music and recorded sources. Typically, the end goal is to permit the use of a <u>significant</u> amount of equalization on the mike channel. If subsequently fed through a multi-band processor, the processor will tend to redistribute the spectral energy, negating the function of the equalizer.

The desired end goal is accomplished by feeding the mike channel into the program circuit at a point after the multi-band processing. Several stations have implemented this configuration, using the summing-point on the CX-1 board as the point to introduce the mike channel.

This was accomplished by connecting a fifth resistor (in addition to the existing four representing the four bands) to the inverting input of the summing amp. This practice has been formalized in the CX-2. A signal introduced at Pin 1 of connector Y301 on the CX-2 will be superimposed on the AUDIO PRISM's output.

The gain at this mixing port is determined by the value of R126. Lower values of R126 will produce higher gain. Solder pads are provided for R126 and its position is marked, although it is normally not inserted at the factory, to minimize the possibility of stray noise pickup.

A value of 33K is recommended for R126 if this facility is implemented, although the exact value is not critical. Values less than 10K are discouraged. If more gain appears necessary to establish the desired voice injection level, the level provided to the AUDIO PRISM should be increased.

As described earlier, this is a partial facility for post-mix injection, a convenience for those who would add this feature themselves to their AUDIO PRISM's. No provision has been made to get the post-mix signal into the AUDIO PRISM chassis. This task, as it did before, rests with the user making the modification.

The RF and transient protection integrity of the AUDIO PRISM should be preserved. In the past, the method of getting the signal into the chassis has been to "borrow" two (or one) of the STEREO STRAP lines on the rear panel. As few stations implement the stereo strap function, these conductors, with their attendant RF filters and surge protection, are usually available. Consult the schematics for the FP-1 and the MB-2. These conductors can be intercepted either at the cable connecting the FP-1 and MB-2, or on the bottom of the MB-2.

When done with reasonable care (!), implementing these post-mix injection modifications will not invalidate the AUDIO PRISM warranty.

In low-RF environments, acceptable performance may be had by bringing a single, unbalanced conductor into the AUDIO PRISM. The proper connection point is at the motherboard end of the cable which interconnects the CX-2 and the MB-2. A signal introduced on pin #1 is connected through to R126 on the CX-2. There are no traces on the MB-2 connecting to pin #1. This conductor "dead-ends" there.

In high-RF environments, a balanced input is recommended. A small balanced-to-unbalanced converter will have to be constructed for inside the AUDIO PRISM. This can be made from a TL071 and four resistors.

Be certain that the combined voltage of the post-mix injection signal and the outputs of the four bands do not result in clipping on the white front panel test point (TP205 labeled BROADBAND SAMPLE). This is best determined by observing TP205 with an oscilliscope.

When implementing post-mix injection of the mike channel, in addition to the equalization, it is usually necessary to add some type of active processing to the mike channel. This processing would be located in the circuit after the mike preamp, but before it was fed to R126 in the AUDIO PRISM. Without separate processing of its own, the mike channel would sound underpowered when compared to the music channel. Multi-band processing is generally not required in the mike channel. A single-channel, fast limiter, and possibly some noise control circuitry, is generally all that is required. The Gain Brain and the Kepex by Valley People (Nashville, TN) are popular devices for these applications.

When implementing post-mix injection, the air staff will have little or no control of voice-over levels. As both the music and voice are processed prior to the summation point, which source dominates and by how much, is fixed by the output level of the voice channel limiter. For the station attempting to insure uniform voice-overs and air sound from a pre-occupied air staff, this can be a desired feature. For the station needing maximum flexibility for control by the air staff, this can be a hinderance.

One alternative is to place the announcer's mike fader in the line-level circuit between the output of the voice channel limiter and the post-mix input of the AUDIO PRISM.

There is one other consideration, not readily obvious, which must be kept in mind when contemplating separate voice channel processing. Similar provisions must be made on the back-up audio chain. More than one station has installed separate voice and music processing systems only to switch to the back-up system many months later, and wonder why the announcer's voice was not coming over the air.

Additionally, the voice post-mix input is not operational when the AUDIO PRISM is in BYPASS MODE. When doing pink-noise set-up on the AUDIO PRISM's where separate voice and music processing is employed, it will be necessary to switch to the back-up processing system, or confine adjustment to periods when no live announcing is necessary.

Separate channel processing is an involved procedure which affects daily operations in an ongoing fashion. These can be as significant as the hardware considerations. Generally, it is warranted only in larger markets when the operational inconveniences are outweighed by the larger potential rewards of the market.

OTHER IMPROVEMENTS

Several other less significant changes were previously made to the AUDIO PRISM, and are enumerated here for completeness.

The M-100 processor board is replaced by the M-101 processor. The former "dot-graph" of the M-100 is replaced by a "bar-graph". If driven beyond the end of the scale to the right, the entire bar display will blank out, calling attention to the overdriven condition. While this indicates that the 6 dB SAFETY BUFFER will be activated, it in no way indicates any damage to the equipment. This is an intended function.

Lastly, a sixth, black, front panel test point has been added to the left of the "Q" light in the LOW BAND. This is a circuit ground intended for connecting the black probe from the Simpson 260 meter during setup. Many users had requested this feature, citing a difficult time in obtaining a reliable ground.

Instructions for Installing the
TEXAR PR-1
Phase Rotators

1). Ascertain that you have AUDIO PRISM's with serial number(s) greater than 505. If below 505 please contact the factory at (412) 856-4276.

2) Remove the AUDIO PRISM top cover by removing the retaining screws.

3) Search the motherboard for ID symbol Y305. This is located physically at the back of the unit near the center on the actual motherboard.

4) Remove the four (4) M-101 processor cards. Disconnect the power supply connections at the right hand side of the motherboard at ID Y304.

   NOTE: Make sure you mark the three (3) wires you remove so you can reconnect them in their proper positions.

5) Disconnect the connector at ID Y303 at the left side of the motherboard.

6) Remove the motherboard retaining hex nuts (8).

7) Solder the 5 pin SOCKET (attached for shipment on the end of the multi-color flat cable) to Y305. The keyed end of the socket must be at pin 5 (to the right).

8) Route the PR-1 Phase Rotator cable UNDER the motherboard to the left wall of the chassis and attach the black module to the left hand wall of the chassis. (If you received PR-1's WITHOUT mounting holes, attach to the wall with Velcro or double-sided tape.)

9) Reinstall the motherboard by reversing steps 4 thru 6 above, making sure the multi-color flat cable is not damaged or pinched as the motherboard is installed above it.

10) Plug the multi-color flat cable into the newly installed socket at ID Y305. Again, make certain that the KEYED end is at pin 5.

11) Reinstall the four (4) M-101 processor cards.

12) Reinstall the AUDIO PRISM into service.

There are no further modifications needed to the AUDIO PRISM to allow the PR-1 Phase Rotators to operate.

Referring to the AUDIO PRISM USERS' MANUAL defeat the Optimod 8100 phase rotator circuits on cards 3 and 4.

If you experience any problems with the installation or have any questions, please call TEXAR customer support at (412) 856-4276.

USERS' MANUAL

TEXAR AUDIO PRISM™

DIGITALLY CONTROLLED AUDIO PROCESSOR

JULY 1982

REVISED - AUGUST 1984

REVISED - DECEMBER 1985

REVISED - APRIL 1986

REVISED - JULY 1986

TEXAR
TEXAR INCORPORATED
616 BEATTY ROAD
MONROEVILLE, PA 15146-1502

(412) 856-4276

(412) 85-MICRO

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INTRODUCTION

The writer of any technical manual is caught between the conflicting goals of keeping it direct and to the point, and including design philosophy so the user can better understand why certain design choices were made. We have tried to accomplish both goals in the following pages by use of indented paragraphs. Where the text includes design philosophy or tangential information felt to be worth inclusion, but not essential for installation or to the majority of readers, that text is presented in paragraphs that are indented ten spaces from the left margin.

Looking to "get it on the air"? Read only those paragraphs which begin at the left margin. Looking for a fuller understanding of the philosophy of the unit and possible unusual situations which you may encounter? Read all paragraphs.

Part numbers in the AUDIO PRISM are assigned according to which circuit board they are on. Ninety-nine of each type part number are allocated to each circuit board according to the following system:

M-101 Processor	1- 99
CX-2 Control Board	101-199
DB-2 Display Board	201-299
MB-2 Mother Board	301-399
PS-1 Power Supply	401-499
FP-1 Filter & Protection	501-599
AMC-1 EAGLE™ Clipper	601-699

By this system, U5 is on the M-101 Processor, R301 is on the MB-2 Mother Board, etc.

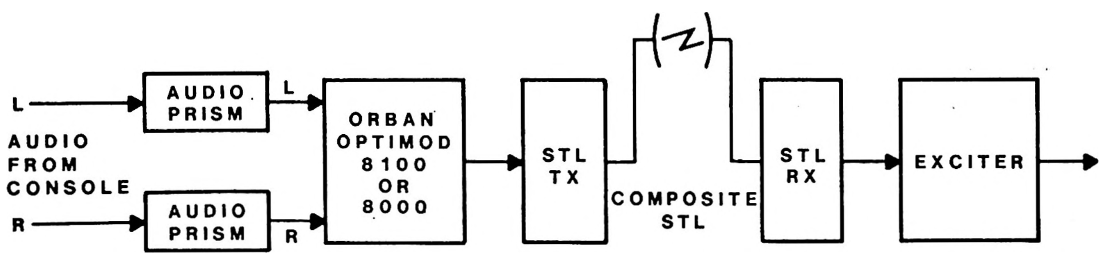
The last three pages in this manual are forms masters for photocopying. They are included for your convenience and are exempt from the Copyright Notice of this manual when used by our customers for the purpose intended.

The first, a suggestion form, is self explanatory. Use it whenever you think you have an idea which could make the AUDIO PRISM or it's manual better. The second form, an Equipment Malfunction Report, should be filled out as completely as possible and included in the shipping carton anytime equipment is returned to the factory for repairs. The last form will assist you in adjusting the controls of the AUDIO PRISM. It will be described in detail later in the manual, in the section ADJUSTMENT, SETTING THE MIX LEVELS.

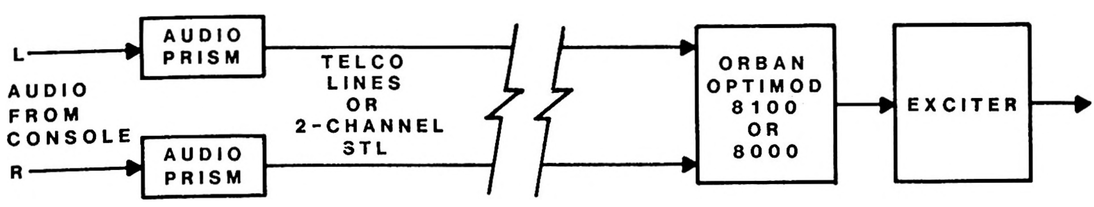
DESCRIPTION

The TEXAR AUDIO PRISM™ is a very high-performance multi-band audio processor designed for major market broadcast use and other applications where sophisticated program handling is required. It utilizes four intelligent, digitally-controlled processor cards to achieve high apparent loudness while producing few processing artifacts. This absence of undesirable subliminal listener aggravation reduces listener tune-out, producing higher quarter-hour ratings.

For AM broadcast, the AUDIO PRISM is designed to be used with the TEXAR EAGLE™ family of AM Modulation Controllers. For FM, the AUDIO PRISM is designed to be used in conjunction with the Orban 8100, 8000*, or other high-quality limiter/stereo generator combination. If transmitter and studio are at separate locations, the preferred placement for the AUDIO PRISM is at the studio as shown in FIGURE 1A, below, and Figure 1B, on the following page. This prevents accidental overdriving of the telephone lines or STL and provides the maximum signal-to-noise ratio over the program circuit.



FOR USE WITH COMPOSITE STL (SUCH AS MOSLEY PC-606)

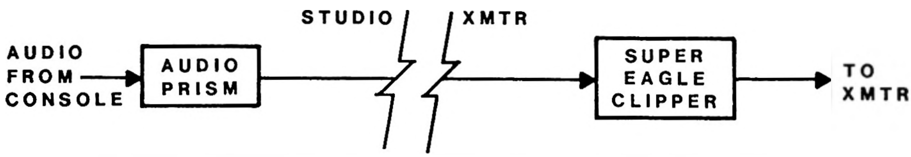


FOR USE WITH TELCO LINES OR 2-CHANNEL (SPLIT) STL

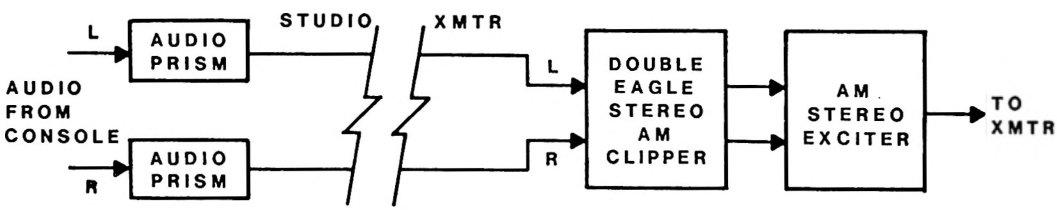
- FIGURE 1A -
RECOMMENDED FM CONFIGURATIONS

* OPTIMOD 8100 and OPTIMOD 8000 are registered trademarks of ORBAN ASSOCIATES INC., San Francisco, California.

EAGLE™ CLIPPER (INTERNAL) FOR SINGLE-SITE MONAURAL OPERATION



SUPER EAGLE™ CLIPPER FOR SPLIT-SITE MONAURAL OPERATION



DOUBLE EAGLE™ STEREO CLIPPER FOR STEREO OPERATION (ONE or TWO SITE)

- FIGURE 1B -
RECOMMENDED AM CONFIGURATIONS

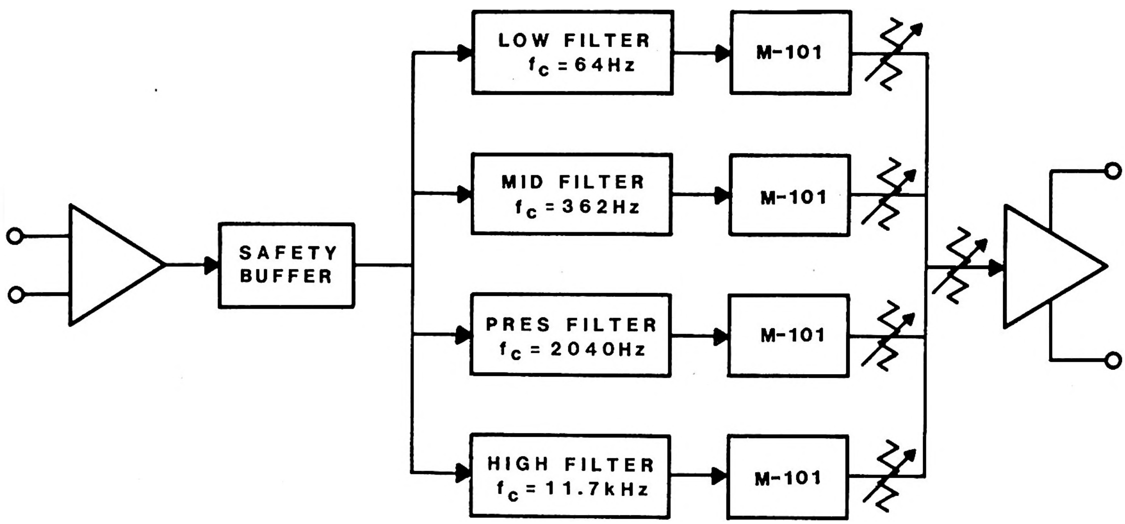
The unit is completely self-contained in a single 1 3/4" rack-height enclosure and will operate normally in severely hostile electrical environments. Extensive RFI filtering and a three-part lightning protection circuit are standard on all conductors leaving the chassis.

The input impedance is factory-wired for 600 ohms resistive, but can be changed to 10 K (nominal) bridging with the removal of a single resistor. See INSTALLATION section. The output will drive a 600 ohm balanced load to +12 dBm nominal program level.

All controls are front panel mounted. There are no controls on the rear panel or inside the unit. Complete set-up takes less than two minutes and requires no test equipment other than a simple VOM. The internal PINK NOISE alignment generator is flat within plus or minus 0.8 dB.

The AUDIO PRISM is arranged in typical motherboard/daughterboard fashion. There are a total of nine circuit boards of six different types. There are four M-101 digitally-controlled processors, and one each of the following: motherboard, display board, control board, RFI filter and lightning protection board, and power supply board. Each board, including the motherboard, can be easily removed .

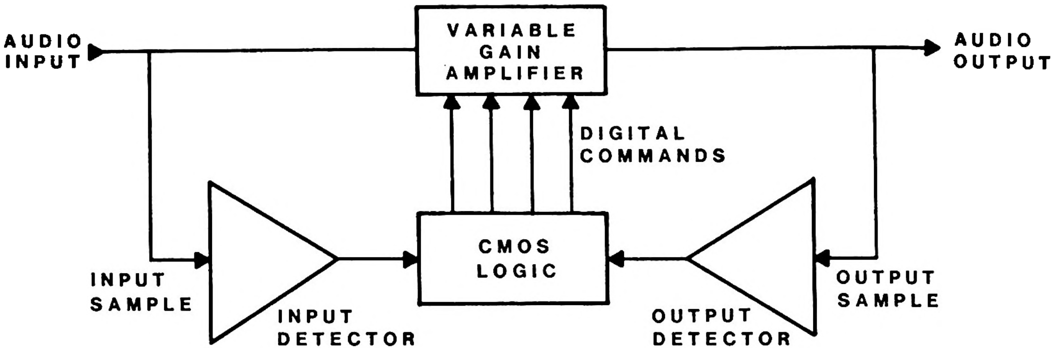
FIGURE 2 is a simplified block diagram of the unit.



- FIGURE 2 -

THE M-101™ PROCESSOR

The heart of the AUDIO PRISM is four digitally controlled, model M-101 processor boards. The input and output signals are level-detected as shown in FIGURE 3. This information, plus reference voltages from the motherboard, are considered by the CMOS digital logic. Taking into account a knowledge of past events, the logic chooses what change (including no change) to make in the gain of the variable gain element.



BLOCK DIAGRAM OF M-100 PROCESSOR CARD

- FIGURE 3 -

The M-101 implements a hardware digital algorithm. The instruction set is determined by non-volatile CMOS devices and the arrangement of their interconnections on the PC board. (There is no central microprocessor). This eliminates the problems of keeping the memory alive in the event of a power outage and of using sensitive RAM memory in a harsh electrical environment.

It is important to point out that the audio does not pass through any active device on the M-101's. The audio merely loops onto the M-101, through the variable attenuator, and back onto the motherboard. The remainder of the board is sidechain. (This should relieve the engineer who spied the TL084CN on the M-101 and questioned the use of this chip in a critical audio application.)

The capacitors which determine the M-101 attack and release speeds are located on the motherboard, permitting the use of identical M-101's in different bands. When an M-101 board is inserted, it is automatically connected to whatever time constant components are appropriate for that frequency band. Only one spare board is required as a result.

If necessary, M-101 boards can be replaced while the unit is energized and in circuit. No damage to either the AUDIO PRISM or the M-101 will result.

FRONT PANEL INDICATORS

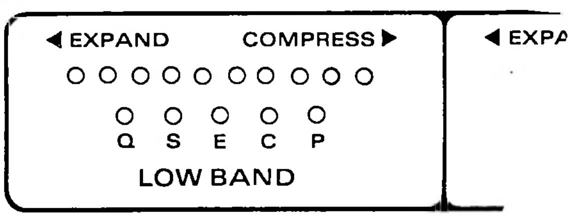
The action of each of the four digitally controlled processors is completely described by the front panel indicator LED's. Both the immediate action which it is implementing, and where it is in its range of action, are called out.

Ten LED's are arranged in typical bargraph fashion along the top of each band's display area. The four leftmost LED's in each group are colored green and indicate EXPANSION. The next five LED's in each bargraph are colored yellow and represent COMPRESSION. The LED furthest to the right in each bargraph is colored red and also indicates COMPRESSION. Its red color signifies the maximum amount of compression possible in that band without activating the SAFETY BUFFER (Described on the following page).

With no signal, each processor will recover to the QUIESCENT position, the center of the bargraph. As the number of LED's is even, there is not an LED at the exact midpoint; the display will come to rest on the first yellow LED.

Each processor has a range of 20 dB from maximum expand to maximum compress. Each LED then represents approximately 2 dB of gain change.

Each band has five additional red LED's as status indicators. From left to right, they are labeled Q, S, E, C and P; corresponding in order to QUIESCENT, SIGNAL, EXPAND, COMPRESS, and PEAK.



- FIGURE 4 -

The SIGNAL LED is just that; it lights when there is signal present in that band, and goes out immediately when there is none. This function is frequently called the "gate". The threshold between what amplitude is considered signal and what lesser level is not, is controlled by the GATE control. The interaction between this LED and control is described more fully in the ADJUSTMENT section. The processor will expand or compress only when there is signal present and the SIGNAL LED is lit.

If the signal in a particular band falls below threshold and the SIGNAL LED goes out, the gain of that band will freeze. The processor will then wait for approximately 1.7 seconds without changing gain. If the signal has not returned by the end of that period, the QUIESCENT LED will light and the gain will slowly recover to the QUIESCENT position.

At the instant that the signal does return, the SIGNAL LED will relight, and the QUIESCENT LED will go out.

The EXPAND and COMPRESS LED's signify respectively that the processor is increasing gain and decreasing gain. In normal operation, these two will flash back and forth, much the same as a railroad crossing warning flasher. The degree of activity of these two LED's is controlled by the DENSITY control. Increasing the setting of the DENSITY control increases the activity of these LED's, and vice versa. Again, see the ADJUSTMENT section for a more complete description of this interaction.

The PEAK LED signifies that a sudden transient has occured which threatens to flattop the amplifier following the individual processor (U302B, U303B, U304B or U305B). The magnitude of the peak is such that the COMPRESS circuit may not respond quickly enough to prevent the flattopping and attendant distortion. At this time the PEAK circuit acts to prevent the possible distortion. Under normal operation, this LED will flicker frequently, but will not remain lit for any sustained period.

There is a wealth of information available in the LED displays. A few minutes of watching them in reduced light while listening to familiar program input will greatly help in understanding their action.

Lastly, there are four status LED's which reflect the operation of the overall AUDIO PRISM. They are marked BUFFER FULL, BUFFER ACTIVE, +15V and -15V.

The safety buffer is a broadband, variable-gain element preceding the point where the signal is split into four bands. Decreasing the gain of this stage will decrease the level fed to all four M-101's. Normally, this stage will operate at maximum (unity) gain. If the input level increases to the point where one of the four M-101's reaches maximum compression (the bargraph reaches the red COMPRESSION LED on the right), the broadband stage will reduce its gain, and the input level to the M-101's. The BUFFER ACTIVE LED will light to indicate that the safety buffer has acted to protect the following stages.

If driven beyond its control range, the bargraph for an M-101 card will blank out. This is an intended function designed to attract attention to the condition. It in no way reflects an equipment malfunction or damage to the unit.

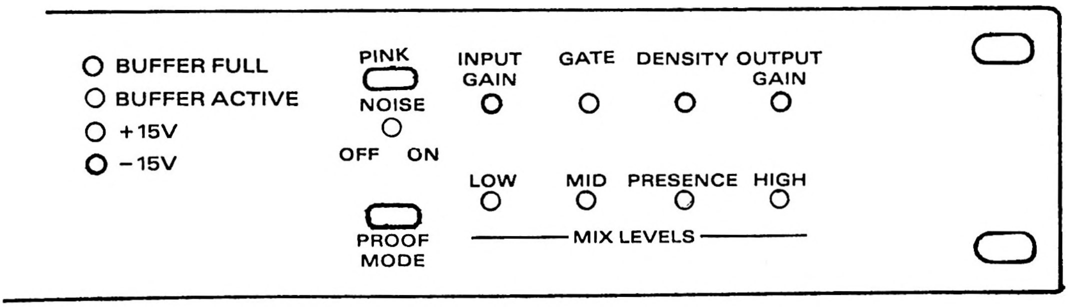
The safety buffer has a range of 6 dB. In the event that the input level is so large as to require this full amount of attenuation, the BUFFER FULL LED will light to indicate that the AUDIO PRISM is no longer able to protect itself. Any further input level increase will result in gross distortion.

The BUFFER ACTIVE LED will light only infrequently, but is no cause for alarm when it does. While it indicates that the input level to the AUDIO PRISM is higher than normal, it also indicates that the unit has taken corrective action to compensate for this condition. The BUFFER FULL LED, on the other hand, should never light. If it does, it indicates gross overdriving, and the input level should be reduced.

CONTROLS

All controls for the AUDIO PRISM are mounted on the front panel, easily accessible behind the removable name plate. There are two toggle switches and nine potentiometers. See Figure 5.

The toggles are PINK NOISE, which turns on the internal calibration generator, and BYPASS MODE (formerly the PROOF MODE), which bypasses the processors to permit equipment set-up. As activation of either of these functions is undesirable while the unit is on line, the switches are recessed to prevent accidental use. The toggles can easily be manipulated by the end of a small screwdriver when desired.



- FIGURE 5 -

The potentiometers are broken into three functional groups. The first group serves simply to adapt the AUDIO PRISM to its surroundings. It contains the INPUT GAIN, BYPASS GAIN and OUTPUT GAIN controls.

The second group, which could be called the "coloration" group, contains the MIX LEVEL controls, LOW, MID, PRESENCE and HIGH.

The last group could be called the "character" group and contains the DENSITY and GATE controls. The DENSITY control is ganged to all four M-101 processors and controls the degree of action which they will take. At low settings, the AUDIO PRISM serves as an intelligent but gentle program control device very desirable for the aural channel of television. Further clockwise, the DENSITY control will produce a very dense (but not fatiguing) output to maximize the coverage of AM or FM broadcast.

This single control smoothly and harmoniously performs the same transformation which on other processors might require several controls to change attack times, release times, and compression ratios. It does, in fact, change the whole "character" of the AUDIO PRISM.

The DENSITY control is described in more detail in the THEORY OF OPERATION section, later in this manual.

The GATE control sets the arbitrary threshold which determines when the SIGNAL LED will light. Signal inputs to each band above this level will be considered desired signal. Inputs below this level will be considered undesired noise. Like the DENSITY control, this control adjusts all four M-101's simultaneously. Note that each of the four bands is individually gated, that is to say that one may gate off while the other three do not.

One station may, due to excellent soundproofing of live studios and excellent signal-to-noise ratio of recorded material, have low ambient noise. It will probably use a relatively low setting of the GATE control. Another station, due possibly to lower quality cartridge tape or air conditioner noise, may have a higher ambient noise. It will likely use a higher setting.

The BYPASS GAIN control sets the output level when the unit BYPASS MODE switch is ON. See the BYPASS MODE OPERATION Section later in this manual.

REAR PANEL CONNECTIONS

Interconnection between the AUDIO PRISM and the outside world is by way of a 12-terminal barrier strip located on the rear panel. In addition to input and output, the AGC voltage of each of the bands is available on the strip for stereo strapping. The far right terminal (# 12) is not connected and may be used for auxilary connections.

Each of the conductors at this barrier strip is thoroughly RFI-shielded and transient-suppressed inside the unit.

When used in a stereo application, it is possible to stereo strap two AUDIO PRISM's so that the gain of each of the bands tracks with the gain of the same band in the other channel. This facility is included for the purist who desires it; however, it is not recommended in most applications.

In addition to its normal functions of level control and dynamic noise reduction, the AUDIO PRISM enhances the spatial dimension of program material. This is in fact NOT a minor point. The effect, which is very pleasing, is immediately noticeable to even the most untrained ear. The perception is that there is more depth and dimension to the music. Strapping the gains of the channels together reduces the AUDIO PRISM's ability to produce this effect, and is therefore not recommended.

Unstrapped or independent action of the units will not produce "platform motion" or other negative processing artifacts. Extensive on-air use shows independent action, in fact, produces no negative effects whatever.

The power line fuse is located on the left side of the rear panel. A 1/4 amp slow-blow fuse is supplied with the unit for use on 120 VAC. This should be replaced with a 1/8 amp slow-blow fuse if the unit is rewired for use with 240VAC.

Both the input and output IC's are mounted in sockets to permit easy replacement. The input IC, U301, is a TL072CP. It is located between the orange and blue test points on the motherboard. The output IC, U306, is an NE5532N. It is located next to the violet test point.

INSTALLATION - ELECTRICAL CONSIDERATIONS

As mentioned earlier, for stations with separate studio and transmitter sites, the preferred placement of the AUDIO PRISM is at the studio. This protects the telephone lines or STL from uncontrolled levels, while negating the need for another, less intelligent, processor to protect same. The AUDIO PRISM is intended to receive completely unprocessed audio direct from the audio console or automation machine. The placement of any other processing or preparation device before the AUDIO PRISM will greatly hinder its ability to respond properly to the program.

The AUDIO PRISM integrates about 6 dB of "Burwen-style" noise reduction into the operation of the HIGH BAND processor. Many audio processors accentuate record noise and tape hiss. Noise "suck-up" or "swish-up" artifacts are most noticeable during record fades.

By contrast, the AUDIO PRISM actually improves the signal to noise ratio of marginal material. Sensing the fade, it reduces the HIGH BAND gain, masking the poor quality of the source material.

Proper operation of this function is dependent upon the AUDIO PRISM having immediate knowledge of the fade. Placing another variable-gain device between the console and the AUDIO PRISM will render this function inoperative, and is thus, discouraged.

We have recently noticed a minor resurgence in the use of echo and reverb devices in the air-chain. The proper placement for these devices is <u>before</u> the AUDIO PRISM. Standard reverb and echo devices do not include variable gain devices, and will not impair the proper operation of the noise reduction feature.

The AUDIO PRISM is supplied from the factory wired for 110 VAC. To adapt the unit to 240 VAC operation, do the following:

1) Remove the lid by removing the 20 flat-head machine screws.
2) Loosen all four screws in terminal block Y401 on the power supply.
3) Remove the short jumper between terminals 1 and 2.
4) Making sure the other wire which was not removed is secure in terminal 1, retighten the screw.
5) Remove the short jumper between terminals 3 and 4.
6) Making sure the other wire which was not removed is secure in terminal 4, retighten the screw.
7) Insert one of the short jumpers previously removed between terminals 2 and 3, and tighten the screws.
8) Replace the line fuse with a 1/8 amp fast-blow device.
9) Replace and secure the lid.

All units are shipped from the factory with the input wired for 600 ohms resistive, terminating. Where the signal source is already terminated, the input can be converted to 10 K (nominal) bridging by removing R303 on the motherboard. To find this resistor, do the following: 1) locate the 14-pin DIP header which passes from the motherboard through the rear partition; 2) locate the group of three resistors directly in front of it. R303 is the center resistor in this group of three and has a value of 180 ohms (brown-grey-brown).

The gain of the input stage is such that proper operation will result with input levels between 0 dBm and +12 dBm. For lower input levels, the input gain can be increased by increasing the value of R308 and R309. These resistors are both 33K on units with Serial Numbers over 505.

Higher input gain may be useful to permit the use of passive telephone line equalizers where the AUDIO PRISM is mounted at the transmitter. For input levels of -10 to 0 dBm nominal, both should be replaced with resistors of approximately 100K. The precise value is not critical, so long as the values of the two new resistors match each other within 2%. Differences in value of over 2% will sacrifice the common-mode rejection performance of the unit.

The input stage operates in full differential mode. That is, it responds only to the difference between the two input terminal voltages, not the absolute value of either. As a result, either input terminal may be grounded with no ill effects.

The balanced output is actually two separate single-ended outputs, operating 180 degrees out of phase. If an unbalanced output is desired, use either output terminal with reference to the output ground. Do not ground either of the output terminals.

The output stage is a Signetics NE5532 and will produce a nominal program output of +12 dBm, while providing ample headroom to pass transients without clipping. The absolute output voltage at hard clipping is about 9 volts either side of ground. This is more than required to drive most broadcast equipment.

For those rare instances where additional output is required, the absolute output voltage can be raised from 9 volts to approximately 12 volts by connecting jumpers across R330 and R331. These are 100 ohm, 1/2 watt, resistors, and are located about an inch to the right of the violet test point on the motherboard.

These resistors are part of a very effective surge supression circuit which protects the NE5532 output stage from external transients, as may be encountered when the AUDIO PRISM is used to drive a telephone line.

Because of the danger of external transients, these resistors should not be jumpered out if the unit is used to drive telephone lines. However, because of the signal level limitations of telephone lines, logic dictates this is not one of the applications where additional output level is required.

If you are installing an AUDIO PRISM to drive telephone lines and the AUDIO PRISM was acquired second-hand, or is being moved from some other location in the program chain, check to make certain these resistors are not jumpered out of the circuit.

Connect the signal source to the rear panel barrier strip using a well shielded cable intended for broadcast use, such as Belden 8451. Connect the output using similar cable.

We have noticed on some AUDIO PRISM's returned to the factory for service that the original power plug had been removed and replaced with one with an internal surge supressor. While this degree of thoroughness is to be commended, the AUDIO PRISM is nearly impervious to surges and "brown-outs". Additional protecion is unnecessary.

INSTALLATION - MECHANICAL CONSIDERATIONS

The AUDIO PRISM generates very little heat and is not temperature sensitive. It also requires no regular maintenance or adjustment. As a result, it can be mounted in practically any rack convenient to the program signal path. This could be in the studio itself, a central "master control" room, or if the transmitter and studio are co-located, in a transmitter equipment rack.

It is never necessary, except for repairs, to remove the top cover or remove the entire unit from the rack. As a result, it may be mounted in practically any location where the front panel is easily accessible. Two test points, TP306 and TP307 are accessible through holes in the aluminum cover. Access to these holes may be required during initial adjustment.

Sufficient slack in the input and output audio cables should be provided so that the AUDIO PRISM can be slid partially out of the rack for access to these holes. Several additional access holes are present in the lid if the unit is equipped with the integral TEXAR EAGLE™ AM Modulation Controller. In extremely hostile environments, it may be beneficial to cover all lid holes with tape when not in use to prevent foreign matter from falling through the holes to the circuits below.

There are no ventilation holes on either the top or the bottom of the AUDIO PRISM. Except where its presence would interfere with the ventilation of the OTHER equipment, it may be mounted directly above or below any other piece of equipment.

The only other caveat is that the AUDIO PRISM should not be mounted in a very strong induction field, such as that found in the rack space directly below a Continental 510R-1 (Collins 310Z-2) exciter.

Mount the unit in the desired rack using the screws and nylon "beauty washers" provided. If the installation is temporary, such as for evaluation, and it is desired to only use two screws, choose the bottom two. (Do NOT employ a diagonal two-screw mounting technique.)

USE OF THE TEXAR PR-1 PHASE ROTATOR

The TEXAR PR-1, Phase Rotator is available as a plug-in option to the AUDIO PRISM. It's purpose is to increase the modulation level by insuring that positive and negative peaks are of equal amplitude. For stereo, one is required for each channel. The PR-1 is recommended for all aplications, including AM where asymmetrical modulation is desired. This apparent contradition is explained in the individual manuals for TEXAR AM limiting products: the TEXAR EAGLE™, the TEXAR SUPER EAGLE™ and the TEXAR DOUBLE EAGLE™.

The Orban Optimod 8100 also contains internal phase rotators (one for each channel). When AUDIO PRISM's are used with the Optimod 8100, it is recommended that the phase rotators in the 8100 be bypassed and PR-1's be installed in the AUDIO PRISM's. The justification and simple procedure for bypassing the Optimod phase rotators are contained in the subsection "REMOVING THE PHASE ROTATORS IN THE 8100" later in this manual.

Further description of installation and operation of the PR-1 is contained in a separate section of this manual. See the the yellow plastic tab divider entitled "PR-1 Phase Rotator".

ADJUSTMENT OF THE AUDIO PRISM

One of the most notable features of the AUDIO PRISM is the small amount of time required to adjust it for use. Many processors, particularly those of the "high performance" variety, seem to have an endless number of adjustments. Set-up is a long co-operative project between the engineering and programming departments where one changes first this control, and then that control, to see what effect each has.

The number of controls on the AUDIO PRISM has deliberately been kept to nine. Of those, two are the INPUT and OUTPUT gains, and four are the MIX LEVEL controls. Great care has been taken to insure that the controls do not interact. Each control adjusts its own assigned function and no other. This makes set-up a process of directly setting the desired parameters, rather than experimenting to find the "magic combination".

The external test equipment necessary for set-up has been kept to a minimum. A simple VOM, such as a Simpson 260, is all that is required.

Remove the AUDIO PRISM name plate using a 2mm hex wrench. A small flatblade screwdriver such as a jeweler's screwdriver or Xcelite "greenie" is necessary for the following adjustments. Set the controls to the following approximate positions: INPUT GAIN - fully counter-clockwise, GATE - 10 o'clock, DENSITY - fully clockwise, OUTPUT GAIN - fully counter-clockwise, MIX LEVELS (all four) - 12 o'clock. (These apply to units #505 and later.)

Apply program input at a normal level. Adjust the INPUT GAIN so that the average position of the LED bargraph in the MID band display is the fourth LED from the right end. This is the yellow LED under the letter "O" in COMPRESS. The BUFFER ACTIVE and BUFFER FULL LED's should not be illuminated at this time.

Adjust the OUTPUT GAIN control for the desired output.

Do not be alarmed if the individual processors indicate values of compression and expansion which differ radically from each other. This is normal. With typical program material, the MID band processor will show the most compression, with the dot-graph display approximately centered in the COMPRESSION (yellow) range. The HIGH band will show the least, with the display midway between the QUIESCENT position and maximum EXPANSION. The LOW and PRESENCE bands will show intermediate compression levels, averaging near midscale. These are, of course, generalizations. The exact levels will depend on the instant program input.

This system of staggered compression levels allows the AUDIO PRISM to implement a very effective dynamic noise reduction function. This reduces tape hiss and other undesired high-frequency noise.

As noted previously, the HIGH band processor normally operates with some degree of expansion; it will seldom compress. On music passages lacking in high frequency content, and sometimes during newscasts, it will gate off and recover to the QUIESCENT position. Movement from the expansion range (the left half of the bargraph) to the QUIESCENT position (center-scale on the bar graph) is from left to right. This corresponds to a DECREASE in gain for that band, reducing undesired noise by as much as 6 dB.

Most conventional audio processors will sound differently when they are driven hard than when they are driven with a lower level. This is because the decay constant (release time) of the timing capacitor changes as the voltage on the capacitor increases. Driving the conventional processor harder increases the capacitor voltage, which speeds the discharge rate. This makes the program output "denser". For this reason, many people drive their analog-based processors far into compression on normal program level inputs.

Some announcers will also run their console output levels high, claiming that it makes the transmitted signal louder. To the extent that the higher level quickens the conventional processor's action, they are correct.

By contrast, the acoustic sound of the AUDIO PRISM does not change with changes in absolute input level. The charge and discharge rates of the timing capacitors of the AUDIO PRISM's four individual processors are a function of program character, not program level. Although the absolute control voltage will be different, the voltage recovery rate will be exactly the same when the processor is driven moderately or very heavily. Consequently, there is absolutely no advantage in driving the unit beyond the level recommended in this section.

SETTING THE GATE CONTROL

Observe the operation of the SIGNAL LED's during normal program. They should remain steadily lit during most fully orchestrated popular music, but should completely extinguish between sentences in human speech. They may also extinguish during breath pauses. The exception is the HIGH band, which may gate off much more frequently, as most program sources have less high frequency content.

It should be noted that the above guidance is a generalization. Unlike a computing application where the digital world is one of black and white, music and voice are moving targets. They are best described statistically, so it is not surprising that the AUDIO PRISM's digital control circuit response to them is also statistical. It is unlikely that one will find an unbreakable rule where the SIGNAL LED will always light for one type of event, and always extinguish for another. Rather, one will find that it usually responds in a given way.

If the SIGNAL LED's extinguish too frequently, lower the gate threshold by turning the GATE control to the left. If they remain lit too frequently, raise the threshold by turning the control to the right. This adjustment, like the others, is very predictable and easy to control. For typical radio applications, the optimum setting of this control (Serial Number greater than #505) will usually be about 11 o'clock. For television or radio talk show applications, a setting near 2 o'clock may be desirable to prevent "suck-up" of the higher ambient noise.

The setting of the gate control can be quantized exactly, as the gate reference voltage appears on TP306, the BLACK test point on the motherboard. This test point is accessible without removing the AUDIO PRISM chassis lid through a hole in the left side of the lid. Two holes are seen. TP306 is the rightmost one of the two. (The other is TP307, the DENSITY reference buss).

The useful range of the GATE control for radio applications is between approximately 0.25 and 1.0 volts. A good starting point is 0.7 volts. These voltages are DC.

This control can be fine tuned later without disturbing subsequent adjustments. If, after a few days listening, one finds ambient air conditioner noise in the studio and background tape hiss to be more noticeable than it should be, increase the GATE control setting. If noise is not noticeable, but subtle high end passages, known to be on the record, are missing on the air, decrease the GATE control position.

SETTING THE MIX LEVELS

The next item in order is to set the MIX LEVELS.

The MIX LEVEL controls are simply faders on the outputs of the four individual processors (the inputs to the processors are not individually adjustable). They control the proportions of the signal from each band which are summed together. This is simply called the "spectral mix".

Setting the spectral mix on a multi-band processor has typically been a time-consuming project. Probably the largest reason is the fact that there are few reference points to go by.

It is unlikely a competent broadcast engineer would begin adjusting the phasor controls of a directional antenna system without having a phase monitor, and perhaps also a field intensity meter, to measure the effect of his adjustments. To properly adjust any system, one must have some type of quantizable feedback from that system. The subjective nature of the human ear makes it anything but a laboratory instrument. From what calibrated meter does one read while setting the mix levels on a multi-band audio processor?

One repeatable and easily interpreted method of setting multi-band spectral mix levels is to use a Real Time Analyzer (RTA). On AUDIO PRISM's after Serial Number 320, the recombined output of the four bands is available on the front panel on the white test point labeled BROADBAND SAMPLE, TP205. For those stations with an RTA, this is the proper sample point.

This provides a convenient method of alignment for the station with a robust engineering budget. But it finds little favor to tell a small market station he must buy a three-thousand-dollar piece of test equipment in order to properly adjust a two-thousand dollar audio processor.

A calibration and set-up circuit is included in the AUDIO PRISM which produces accurate results for the station not having access to an RTA. This circuit and its procedure use nothing more than an AC VOM. The outputs of each of the MIX LEVEL controls are available at convenient test points on the front panel. Each is buffered and short-circuit protected so that backfeeding or loading these points will not affect the output on the air. The level is such as to give a useful reading on a 2.5 VAC VOM scale with each of the MIX controls in its normal operating range.

Using the AUDIO PRISM's internal precision PINK NOISE generator and these test points, it is possible to accurately quantize the spectral mix. This is instrumental both in arriving at a chosen mix, and in being able to return to it, should it be tampered with, and in being able to repeat a particular sound at another station.

No inference regarding relative spectral balance should be drawn from the mechanical positions of the mix level controls. The gains of the follower amplifiers (U302A, U303A, U304A and U305A) are not equal, nor are the gains of the bandpass filters equal (U302B, U303B, U304B and U305). As a result, matching physical position of the mix level control shafts does not indicate "flat" response. Judgements regarding the spectral mix should be made only by measuring the voltages appearing at the front panel test points.

It was formerly necessary to turn the DENSITY control to the fully clockwise position to make spectral adjustments, even if the DENSITY control would not be in that position during normal operation. That step is no longer necessary and the control may be left in its normal position. The PINK NOISE switch automatically removes the DENSITY control from the circuit and forces the DENSITY buss to 5 volts for these tests. It also removes the GATE control from the circuit and drops the GATE buss to 0 volts. When the PINK NOISE switch is turned off, it returns both controls to the circuit. This function is accomplished by FET switch Q101 on the CX-2 PC board.

APPENDIX A of this manual is a tabulation of field-proven spectral mixes. They are grouped according to the transmitting or limiting equipment used in conjunction with the AUDIO PRISM.

When used for FM broadcast, the AUDIO PRISM is designed to be used primarily in conjunction with the Orban 8100 or 8000 limiter/stereo generators. In addition to providing suggested settings for the AUDIO PRISM when used with the Optimods, suggested set-up parameters are also provided for both Optimods when using them in conjunction with the AUDIO PRISM.

APPENDIX A does not pretend to describe how the display should appear on an RTA or other precision instrument, as few VOM's have a flat AC voltage response across the audio frequency range. In fact, it is apparent that using the Triplett brand meter equivalent to the Simpson 260, different results will be obtained. This is not to say that the frequency response of the Simpson meter is superior to that of the Triplett, or vice versa, or that either one of them in fact has flat response. The purpose of the table is simply to present a repeatable starting point for the largest possible number of engineers. That meter, to which it was felt the largest number of people have access, was chosen for that purpose.

Tests have shown that identical results are obtained when making these adjustments using the Potomac Instruments AA-51 Audio Analyzer. The AA-51 is now an approved device for this purpose.

Remember that the AA-51 is an AC voltmeter only. If, during this same maintenance session, it is desired to also set the GATE and DENSITY controls according to meter readings, it will be necessary to have on hand a separate meter for these DC readings.

Because of the pulsating nature of pink noise, particularly in the two lower frequency bands, an analog meter (one with a rotating mechanical pointer) must be used for these measurements. A digital readout meter cannot be used. A digital <u>or</u> analog meter may be used for the DC readings on TP306 and TP307.

Using a small flat blade screwdriver, switch the PINK NOISE generator "ON". Ground the black probe lead of an AC VOM or VTVM to the AUDIO PRISM. There is a black test point provided for this purpose, located to the left of the "Q" LED in the LOW BAND section of the front panel. Grounding to the equipment rack may produce incorrect results. Insert the other probe into the green front panel test point in the HIGH band sector. Adjust the HIGH band MIX LEVEL control until the voltmeter reads as suggested in APPENDIX A for the equipment you are using. Move the probe to the PRESENCE band test point and adjust the PRESENCE band MIX LEVEL. Repeat for the two remaining bands.

The output of the PINK NOISE generator is pseudo-random and has some sizeable low frequency thumps in it. As a result, on the two lower bands, the voltmeter will not come to rest on a specific number as it did on the higher bands. Rather it will continuously swing over a small arc on the meter. Note in APPENDIX A that the set-up numbers for each of the two lower bands represent the average reading of the meter. This can be arrived at by reading the maximum and minimum values and averaging them.

The performance and sound of one AUDIO PRISM, set to a given set of parameters, is identical to the sound of another, adjusted to the same parameters. As a result, the parameters listed in APPENDIX A should be close to the optimum for that equipment.

There are three common exceptions:

1) Research indicates female listeners are far more sensitive to high frequencies than are men. Stations whose formats are deliberately skewed for high female listenership may choose to reduce the two higher frequency MIX LEVEL controls slightly.

2) FM stations using music sources with particularly clean high end, such as from a Compact Disk (CD) player, may find it desirable to reduce the HIGH BAND mix level.

3) AM stations operating into particularly narrowband directional antenna systems may find additional coloration is desirable to compensate for the bandwidth of the antenna.

> AM stations which believe they are placed at a competitive disadvantage either by limited antenna bandwidth or directional antenna service area restrictions should know that great strides have been made in the last few years in these areas.

Modern computer synthesis methods permit the antenna design engineer to produce directional antenna patterns which are far more efficient at maximizing service area than were patterns produced by earlier methods. Even for existing stations with towers already in place, significant improvements are usually possible by filing for a new FCC Standard Pattern, specifying new tower currents and phases.

These "optimized" patterns frequently also exhibit superior AM stereo performance.

In addition, many FCC Regulations pertaining to AM allocations were relaxed in 1984 and 1985. Prior to 1984, the antenna design engineer had to satisfy much more restrictive conditions than he would today. Antenna systems designed prior to these dates can benefit from re-evaluation in light of the new regulations.

For the broadcaster content with the <u>extent</u> of his coverage, but desiring improvements in the signal's <u>fidelity</u>, modern computer methods also permit design of matching networks which are higher fidelity. They present the transmitter with a more uniform load impedance and have a more uniform transfer function across the audio band. This procedure of redesigning matching networks, which has become quite popular in recent years, is called "broadbanding". Broadbanding will also improve a station's AM stereo performance.

A technical paper describing these new techniques in greater detail was delivered at the 1984 NAB Convention by Glen Clark and Edward A. Schober, P.E. Reprints of the paper, entitled <u>COMPUTER-OPTIMIZED DIRECTIONAL ANTENNA PATTERNS IMPROVE AM COVERAGE</u>, are available from TEXAR at no charge.

*** * * CAUTION * * ***

Care should be exercised when adjusting the spectral mix not to operate all four controls near the tops of their range. The gain of the summing amplifier (U101 on the CX-2 board) is chosen to permit full utilization of the capabilities of the output stage with normal adjustment of the mix level controls. Clipping at the summing point may result under some combinations of settings. No damage to the unit will result, but noticeable distortion can be produced.

This distortion cannot be removed by reducing the output gain control as it is already present before the output stage. While the output stage can produce voltage swings to approximately 9 volts either side of ground (12 volts with the surge supression circuit defeated), if this problem is present, distinct flattop clipping will be apparent, but the amplitude of the clip may be at considerably less than 9 volts. It will be a scaled model of a clipped wave (scaled by the output gain control, R104).

This problem usually occurs only after a series of empirical adjustments where first one, and then the other, mix level is increased. All of the recommended mix levels in APPENDIX A have been tested to insure they do not clip at the summing point.

An approximate rule of thumb is to add the mix level voltages at the front panel test points and see if they total 20 or greater. If they do, clipping is likely and the settings should be reduced. (The recommended settings for use with the Orban 8100 are 4,4,4 and 3 dBm, which total 15, which is less than 20.)

If you have arrived at a desired spectral mix by trial-and-error, would like to preserve that mix, but find it produces clipping, the solution is simple. With the PINK NOISE generator on, measure and record the voltages found at each of the front panel test points. Then readjust the mix controls to subtract an equal number of dB from each. Increase the OUTPUT GAIN to make up the difference in output level.

> EXAMPLE: Front panel voltages of 6, 7, 8 and 5 dBm are encountered at the test points. The sum of the values is 26 so clipping is likely. Reduce the settings of each control by 3 dB so the voltages are 3, 4, 5 and 2 dBm. The sum is now 14, which is less than 20. Increase the OUTPUT GAIN control by 3 dB to return the output level to its previous value.

As described earlier, later production AUDIO PRISM's have a front panel test point marked BROADBAND SAMPLE. This is the output of U101. An oscilloscope with accurate vertical calibration can be used at this point to be sure that no flattopping is present. The voltage swing here should not exceed + or - 15V.

There is a tremendous temptation, on a multiband audio processor, to attempt to measure its performance at the output while sweeping the input with sine wave tones. Steady-state tones can frequently provide erroneous information for a number of reasons. Probably the most obvious source of such abberations relates to the gate circuit. For tones of certain amplitudes, when the tone is in the center of one particular band, the corresponding M-101 processor card will expand to capture the signal. The other three bands will gate off. Only the active band will provide a significant output to the equipment which follows.

When a tone of the same amplitude is injected at the crossover frequency between two bands, the M-101 processor cards for both bands will expand to capture the signal. Unlike the previous case where only one band provided significant output, now both M101's will provide significant output, making the overall output up to 3 dB higher than when the tone was centered in one of the bandpass filters. This phenomenon, known as "crossover buildup", is not specific to the AUDIO PRISM, but is a function of all multi-band processors.

It is best to make adjustments and draw information only when the AUDIO PRISM is fed either pink noise or program material. Attempting to draw conclusions from sine wave signal sources will seldom produce meaningful information and should be avoided.

If it appears that settings other than those appearing in APPENDIX A are most suited for the station, a simple familiarization with the frequencies in each band will often make adjustment much easier. Each of us usually has a general idea of what the sound of a certain frequency is, when expressed to us in Hertz. However, under program conditions, we may be less able to accurately judge which band a certain group of frequencies falls into, even if we know the frequency limits of the band.

One way to make a direct correlation in our minds between a particular aural sound and the band it falls into is to listen to the outputs of the four individual bands. This is easily accomplished at the four front panel test points (TP201, TP202, TP203 and TP204). Connect an amplifier to drive a speaker or pair of headphones. The gain should be such that sufficient sound level can be produced from a 1 volt AC source. Connect a test probe to the input of the amplifier and insert it into one of the front panel test points. The sound will be very unnatural because it contains only one band of frequencies. Make a mental note of what frequencies are in that band. Move the probe to the next band and again make a mental note. Repeat the process for the remaining two bands.

Then monitor the overall on-air sound with typical program material. When attempting to boost or attenuate certain frequency bands to achieve the desired overall air sound, it will be obvious which mix level control affects the desired band of frequencies.

ESTABLISHING THE L-R CHANNEL NULL

A monaural source fed to any stereo system (AM or FM) should produce no modulation of the difference (L-R) subcarrier. This, of course, requires matching both the phase and amplitude of the signals in the left and right channels. In a multi-band processor, this means matching the spectral mix levels of the left and right channel units to each other to achieve the null.

One method to null the L-R subcarrier is to simply set the front panel test point sample voltages to equal levels as observed on the Simpson 260 or Potomac Instruments AA-51. This will produce a channel-to-channel balance closer than can be detected by the human ear (i.e. listeners). However, if one measures stereo performance of the station on an oscilliscope lissajous figure (X-Y display format), a better null is desirable.

> Frequently the oscilliscope is used to monitor phase alignment of stereo sources, particularly cartridge and reel-to-reel tapes. A spreading of the diagonal scope trace on a source which should be monaural indicates a poorly phased source, which will cause loss of high-frequency response, when received on monaural receivers.
>
> The lissajous pattern will respond to mismatches in both phase and amplitude. Small phase mismatches will degrade the station's fidelity significantly; small amplitude mismatches will affect it only slightly. However, it is nearly impossible to tell one from the other on the lissajous pattern. While a small amplitude mismatch will not significantly affect the station's quality, it will hinder detection of phase mismatches which will.
>
> As a result, a better null than can be produced by the above method is desirable for the station using a lissajous monitor. Unit-to-unit tracking of the AUDIO PRISM's bandpass filters and dynamic characteristics is excellent, allowing a DYNAMIC, broadband null of the difference channel of greater than 30 dB.

Certain techniques simplify achieving the L-R null. The preferred method requires an external pink noise generator.

Connect the pink noise generator to the paralleled inputs of both AUDIO PRISM's. (One cannot simply turn on both INTERNAL pink noise generators. For this procedure, it is necessary for the units to both operate on the same source material. While the two internal pink noise generators will have the same spectral output, they are not synchronized in time.) Adjust the input gain controls on both units so that the mid band bar graphs read about 3/4 scale. Connect a 10K resistor to the "+" output terminal of one AUDIO PRISM. Connect a second 10K resistor to the "-" output terminal of the other AUDIO PRISM. Twist or solder the two loose ends together. At this junction is a high-impedance source of L-R signal. One can look at it with any of three measurement devices: real time analyzer, oscilliscope or voltmeter.

Regardless of the selection of measurement devices, the object is the same: to minimize the signal present at this junction while the two AUDIO PRISM's are fed the same pink noise signal. One should specify that one unit is the reference unit and that the other unit should be matched to it. If one tunes first one AUDIO PRISM and then the other, it is possible to creep away from the original desired settings.

If using the real time analyzer, it is a simple matter to tell which frequency band(s) the residual signal is in, and therefore which mix level controls should be adjusted to improve the null. The oscilliscope also displays frequency information in somewhat less direct form. Using the oscilliscope, one can judge the relative frequency of the residual from its wavelength on the display tube. Long, loose lines which bounce every few seconds indicate a residual in the low band. Tight waveforms that appear like grass on the scope tube indicate a residual signal in the high band.

To judge what waveshape will be displayed on the scope tube by each band can best be judged by momentarily unbalancing the null to see what happens. Rock the control for a given band and channel a significant distance from the null. Then turn the control back through the null and well past it in the other direction. Return the control to the null and then unbalance the next band. Repeat this procedure for the other two bands until your eye can recognize the characteristic wavelengths of the frequencies contained in each band.

Then null each of the four bands. Repeat the procedure until the null is at least 30 dB. It may be necessary to go through each control several times to reach this level.

If neither a real time analyzer or an oscilliscope is conveniently available, a simple audio voltmeter can be used. The voltmeter is the least preferred device because while it displays amplitude information, unlike the other two, it does not display frequency information.

Turn the low band mix level control for the best null on the audio meter. Then repeat for the other three bands. Return to the low band control and repeat the sequence until a null of 30 dB is achieved.

At any one time, an L-R imbalance is usually caused primarily by one particular band. That band which is furthest out of balance will mask the other bands which have less imbalance. As a result, adjusting one of the bands which is less out of balance will have little of no effect on the meter reading. Only the band which is the primary contributor will have much effect. Once this control has been adjusted so that its contribution to the total imbalance is less than that of another band, it will have little effect, and it will then be necessary to determine which control is the new primary contributor.

Without frequency information, this can be ascertained only by trial and error. For this reason, adjusting the L-R null using only an audio voltmeter is a recursive procedure. While the null produced using only an audio voltmeter can be equal in quality to that produced by the other instruments, to achieve it will require more patience.

If an external pink noise generator is not available, acceptable results can generally be obtained using the real time analyzer or oscilliscope methods and monaural program material as source. (It will not work with the audio voltmeter method because of the wide swings in level.) As mentioned previously, tones are not an acceptable signal source when adjusting the L-R null. They will usually cause more confusion than progress. Only full-spectrum signal sources are useful for this procedure.

Some consoles have a "mono" switch which straps the left and right channel outputs together. Otherwise, patch one channel of the console to a patch mult and feed both AUDIO PRISM's from the mult output. Adjust the AUDIO PRISM mix level controls in the recursive manner described above for the best null. This will require full spectrum music input to work properly. Too much level variation exists in voice to produce meaningful results. If this procedure is performed while on the air and a spot cluster or news block comes up, it may be better to wait for a return to music before continuing.

After the null is achieved at the output of the PRISM's, the two 10K resistors should be removed from the barrier strips. The RF protection filters make the AUDIO PRISM dynamic output source impedance some small finite value greater than zero. Accordingly, some minor but measurable crosstalk between channels will result if the resistors are left in. Do not yet reconfigure the input source for stereo.

With the spectral mixes of the two AUDIO PRISM's matched, producing the proper on-air L-R null requires only balancing the left and right channel overall gains. This can be done using one of the AUDIO PRISM output controls, or one of the input controls of the limiter device which follows the PRISM's. For AM it is likely this limiter device would be the TEXAR DOUBLE EAGLE™. For FM it is likely the limiter would be the Orban Optimod 8000 or Optimod 8100.

Ideally, the L-R subchannel should be read off-air on the station's modulation monitor. This will test the system closed-loop, including any anomalies which may also be present in the exciter, transmitter and antenna system. If this is not convenient, a reasonable null can be achieved by turning the limiter metering switch to the L-R position. Slowly vary the input gain of one channel of the limiter. Some limiters switch 20 dB of additional gain into the meter circuit in the L-R position to allow locating the null more precisely.

Remember to reset the console to stereo or reconfigure the patch bay to stereo operation when the procedure is complete.

During the early set-up procedure, when one intends to make further adjustments after listening tests, it is often desirable to have a reasonable balance of channels, but not efficient to spend the time to adjust for a perfect channel balance. When using the Optimod 8100, a reasonable balance can be obtained by adjusting one of the input gain controls for equal flashing of the two front panel "HF LIMIT" LED's. Even with stereo program material, this is a reasonably good indication of channel balance. Minor variations can usually be averaged out with the eye to produce an acceptable null.

SETTING THE DENSITY CONTROL FOR RADIO APPLICATIONS

As was noted earlier, the DENSITY control establishes the character or aggressiveness of the processing. While the intelligence of each processor is unchanged, varying the DENSITY control changes the number of actions which it will choose to take in a given time period. At mid-range, it provides dynamic noise reduction, loudness control, and a gentle average level control. This setting is very adaptable to TV and CATV audio channels, where high average modulation levels are neither necessary nor desirable. Rotating this control further clockwise provides all of the above functions, plus a very dense average modulation level.

The DENSITY reference buss, like the GATE reference buss, is available on a test point and can be quantized exactly. The normal setting of the DENSITY control for television and CATV applications will produce approximately 2.5 DC volts on TP307, the GREEN test point on the motherboard. As also described earlier, this test point is accessible through an access hole in the chassis lid.

The normal setting for radio applications (fully clockwise) will produce approximately 4.0 volts at this test point. The exception is satellite up-link program control for radio. Where it is expected that the program will be processed again before it reaches its final destination, a lesser setting may be desirable.

While proper adjustment of this control is essential for proper program handling, its effects are subtle. It is not the primary control by which the broadcast user will determine his loudness/quality tradeoff. Rather that function will be determined almost entirely by the amount of VCA action in the limiter which follows the AUDIO PRISM.

When used with the integral TEXAR EAGLE™ AM intelligent clipper, that function is labeled VCA DRIVE. When used with the separate TEXAR SUPER EAGLE™ or TEXAR DOUBLE EAGLE™ AM intelligent clippers, that function is labeled INPUT GAIN. When used with the Orban Optimods, that function is labeled INPUT ATTENUATOR.

The proper adjustment of those controls on the TEXAR AM intelligent clippers is discussed in the manuals for those units. The proper adjustment for the INPUT ATTENUATOR controls on the Optimod's is discussed later in this manual, in the sections SUGGESTED ADJUSTMENT OF THE ORBAN 8100 and OPERATION WITH THE ORBAN 8000.

A circuit-level description of the operation of the DENSITY control is contained in the THEORY OF OPERATION section, later in this manual.

SETTING THE DENSITY CONTROL FOR TELEVISION APPLICATIONS

Television audio, because of its diverse nature, is perhaps, the most difficult audio to handle. It contains widely varying levels, long pauses,. and no background orchestration to forewarn the processor of what it will do next. For that reason, a set of processor adjustments desirable for one program might be intolerable for another. Many television stations process the audio very little as a result.

Radio usually processes for maximum loudness. Television processes for uniform loudness. Much as it may sound like a game of words, the two are not at all alike. Each radio broadcaster is granted a transmission channel of a given size. Most process the audio to fill the channel as well as possible. This insures that the signal will penetrate every geographic area which the channel size will allow. Increasing the physical coverage area increases the number of potential listeners served, the potential for ratings, and hopefully, eventually, ... advertising revenues.

Television audio, on the other hand, is not the limiting factor which determines the coverage area in that medium. The visual signal is. Commission rules permit an aural carrier power of no less than 10 but no more than 20 percent of the peak visual power. This essentially guarantees that the audio signal will go everywhere that the visual signal does. Frequently, the aural signal will go substantially further. Louder or more dense audio will not then increase the coverage area. Overprocessed television audio will simply sound unnatural and irritate listeners.

The goals for television audio processing are twofold: 1) maintain the peak modulation within the FCC's 100% limit, and 2) produce a consistent product.

The FCC and the NAB have both, from time to time, convened a number of panels looking into the varying loudness levels typically found in television broadcasts. Commercials which have a higher apparent loudness have been a recurring problem. Part of the problem is the number and differing types of television program sources. A commercial on video cassette has a magnetic sound track. The movie which follows it may have an optical sound track. The following live local newscast may originate several floors away. In that newscast, there may be a live, on-location, ENG feed. Each source is usually very different from the last.

Even among similar sources, there is considerable individualism. Of two video cassettes, one may have been recorded locally on the same machine it will play on. The other may have been recorded a continent away on a different machine.

In an effort to produce a consistent product from these varied sources, television broadcasters have sought to control loudness through processing, being conscious that overprocessing is equally irritating in a different way.

The AUDIO PRISM is very adept at creating a consistent product without producing processing artifacts. For this reason it is extremely well suited to this task. A DENSITY control setting of 10 o'clock is a good starting position.

If, after a period of listening, the program appears to lack uniformity, turn the DENSITY control to the right. If it seems too uniform, turn the control to the left. This adjustment, like the GATE, is very predictable and easy to control.

The key to final adjustment of the AUDIO PRISM in television applications is to listen to a number of different source programs. One particularly difficult test is the popular "ROAD RUNNER" cartoon show, which is characterized by extended pauses followed by explosive percussion.

SPECTRAL MIX ADJUSTMENT WITH THE ORBAN 8100

Final adjustment of the MIX LEVEL controls should be done in two parts when the AUDIO PRISM is used in conjunction with the ORBAN 8100. The 8100 is a complex and sophisticated device. It has the ability to change tonal balance in an effort to find the optimum compromise between loudness and fidelity. When doing on-air listening adjustments, it is possible to make a MIX LEVEL adjustment on the AUDIO PRISM which is counteracted by a dynamic response in the 8100.

You have just dialed in more highs. The 8100, in an effort to maintain a uniform sound, has just removed them. The net effect on the air is no change.

To prevent this problem, establish a high-quality monitoring ability at the input of the 8100. Set the MIX LEVEL controls for the desired tonal balance monitored at this point. Then compare the on-air sound of the 8100 to its input. If they are not the same, adjust the 8100 to suit.

Additionally, the use of a composite clipper will have some effect on spectral coloration and balance. The most desirable setting of the AUDIO PRISM MIX LEVEL controls when using the 8100 in conjunction with a composite clipper may be slightly different from what they would be without a composite clipper. Composite clippers are discussed in greater detail, later in this manual.

THE TEXAR REPLACEMENT CARD FIVE (RCF-1)

TEXAR manufactures a plug-in replacement for Card #5 of the Optimod 8100 which more perfectly matches the characteristics of the Optimod and the AUDIO PRISM's. This card installs in three minutes using only a screwdriver and an allen wrench. No soldering or other modifications are required. Installing the RCF-1 does not preclude also installing the Orban "Card Zero"

Adding a pair of AUDIO PRISM's in the program line before the Optimod 8100 will produce significant benefits. As such, the RCF-1 is not a "necessary" addition for the PRISM investment to function properly. However, the RCF-1 does add to the improvement. The RCF-1 will produce up to 2 dB more loudness and has a variable bass boost control, which can significantly increase bass definition.

The time constants of the RCF-1 have been tuned to assume the Optimod input has been pre-processed by two AUDIO PRISM's, thus eliminating radical variations in input level. The RCF-1 should <u>not</u> be installed in a "barefoot" Optimod. The AUDIO PRISM's may be used without the RCF-1, but the RCF-1 cannot be used without the PRISM's.

A complete description of the RCF-1 is contained in a separate section of this Manual. See the green plastic tab divider entitled "RCF-1 Replacement Card Five".

SUGGESTED ADJUSTMENT OF THE ORBAN 8100

The following settings of the Orban 8100 are recommended for use in conjunction with the AUDIO PRISM and when operating with the original Orban Card #5. For suggested adjustment using the TEXAR RCF-1, Replacement Card Five, see the manual section referenced in the preceding paragraph.

RECOMMENDED SETTING FOR STOCK OPTIMOD 8100:

CLIPPING .... 0 (Slightly past 12 o'clock - see text)
HF LIMITING 5

RELEASE TIME .... 0 (Fully CCW)
BASS COUPLING ...... 0 to 4

GATE .... 0 (Fully CCW)

INPUT ATTENUATORS .. (See text)

Some very early 8100's had a different legend silk-screened on the CLIPPING control than the one referred to above. The present legend is in dB of clipping and has both positive and negative values. On it, the "0" point is slightly past the 12 o'clock position. The earlier legend was labeled simply from "0" to "10" in exactly the same manner as the HF LIMITING control is labeled now. For those users with the older legend, the recommended setting of the CLIPPING control is 6.

Be aware that the above recommended settings for the Optimod 8100 are applicable only when the unit is used in conjunction with two AUDIO PRISM's. These settings, if used on a standalone 8100, will produce a very undesirable sound.

No blanket statement can be made as to what the physical position of the INPUT ATTENUATORS should be, because the audio input level presented to the Optimod will be different at different stations. Rather, these controls should be adjusted to produce the proper deflection of the "TOTAL" meter on the Optimod. This is the leftmost edgewise meter on the Optimod face.

As discussed previously, the setting of these two controls, more than any others, will determine the loudness/quality tradeoff of the entire system. Greater amounts of gain reduction will produce increased perceived loudness and reduced quality.

Naturally, this meter will fluctuate with program material. The references below give the range through which the meter should swing. It is important that the RELEASE TIME control be set at 0, as recommended above, for these readings to be meaningful.

The following are also only guidelines. Some radio markets are more hostile toward taste than others. Extremely intense local market conditions may require moving to a higher category than might otherwise be appropriate for your situation.

Light processing can be characterized as the TOTAL meter hanging at 5 dB of gain reduction and peaking to 9 dB. This is appropriate for the broadcaster who has a format unique to his market or who feels the additional quality far outweighs a small reduction in perceived loudness.

Moderate processing can be characterized as the TOTAL meter hanging at 8 dB of gain reduction and peaking to 12 dB. This is appropriate for the large majority of broadcasters who must be competitive but who are concerned with quarter-hour maintenance.

Robust processing can be characterized as the TOTAL meter hanging at 11 dB of gain reduction and peaking to 15 dB. These settings are appropriate for a station which is concerned primarily with listener cumes.

Use of the TEXAR AUDIO PRISM in conjunction with the 8100 simultaneous with the Orban XT chassis is not recommended.

DISCUSSION OF THE RECOMMENDED SETTINGS OF THE GATE THRESHOLD, BASS COUPLING AND RELEASE TIME CONTROLS

Every on-air processing system should include a gate function to prevent "swish-up" of noise during periods of reduced program level. However, there is no additional benefit in having two gates in the same chain. The two gate circuits may, under certain circumstances, interact, causing undesirable effects. The GATE THRESHOLD control on the 8100 should be set fully counter-clockwise.

Reducing the setting of the BASS COUPLING control increases the low "bone-jarring" bass and general excitement of the air product. Most program material is handled satisfactorily with this control fully counter-clockwise (0 setting). However, about one record in four will be handled in an unnatural manner. The phenomenon is not rooted in distortion or sound coloration; rather, the music will seem awkward and syncopated. 4 appears to be the lowest practical setting of this control when used with AUDIO PRISM's.

Relative to the recommended settings for the 8100, the largest number of questions received on earlier revisions of this manual pertained to the setting of the RELEASE TIME. Typically, the question was asked "Do you think it advisable to run the Optimod that fast?"

Much of the circuitry of the 8100 is proprietary and epoxy encapsulated and therefore it is not known exactly how this control functions. However, it is not clear that any time constants in the 8100 are actually lengthened or shortened by it. Rather, one explanation is that the 8100 contains a limiter with a predetermined time constant and a compressor with a predetermined time constant. The RELEASE TIME control determines what proportion of the two mechanisms to use.

When this control is turned fully clockwise, the 8100 implements a large proportion of compression and a small proportion of limiting. When turned fully counter-clockwise, the converse is true. The apparent effect to the user is that turning the control further clockwise makes the front panel meters move more slowly. This explanation receives circumstantial support from the fact that the fully counter-clockwise position on this control on the earlier Optimod 8000 bore the supplemental label "limit only".

As this is written without knowledge of what is contained in the 8100's encapsulated modules, the absolute accuracy of this model is not known. The model does appear sufficiently accurate for the purposes of this section.

The purpose in setting this control fully counter-clockwise is to obtain the minimum gain reduction in the 8100's analog-based compressor while utilizing that amount of gain reduction in the 8100's limiter to achieve the desired loudness.

Ideally, the 8100's compressor would exercise only sufficient gain reduction to compensate for the up to 3 dB of recombination error present in the output of any multi-band processor. This it does with the RELEASE TIME control set fully counter-clockwise.

The effect of turning the release time clockwise is to reduce the amount of limiting and increase the amount of compression. However, the compression function is accomplished in the AUDIO PRISM's digital processors. Further compression is unneccessary.

If the on-air product appears more active or strident than is appropriate for the format, a method of reducing this activity, preferable to increasing the release time, is simply to reduce the input gain of the Optimod. This will have the same effect to reduce the amount of limiting, but without adding to the amount of compression used.

REMOVING THE PHASE ROTATORS IN THE 8100

Each channel of the Optimod 8100 input circuit contains a phase rotator. Their purpose is to remove most asymmetry present in the input signal. It is recommended that the 8100 rotators be bypassed and that two TEXAR PR-1's (one for each channel) be installed in the AUDIO PRISM's.

Phase rotators work by changing the phase relationships between critical odd-order harmonics in the lower voice frequencies. The human ear is not phase sensitive and, if the rotator is properly designed, a listener cannot tell the output wave from the input wave. However, the peak-to-average ratio of a signal after phase rotation is very different from before rotation. While the human ear cannot distinguish between the two, an audio processor will react very differently to the signal after rotation than it would to the same signal before rotation.

Optimum performance requires that the AUDIO PRISM's and the Optimod work in harmony. Using the rotators in the Optimod, rotation takes place after the variable-gain stages in the AUDIO PRISM's but before the variable-gain stages in the Optimod. The AUDIO PRISM's and the Optimod are actually operating on two very different signals. A crude analogy might be to expect two churchgoers to stand side-by-side and sing in harmony while turned to very different pages in the hymnal. A very important rule for the use of phase rotators therefore is:

> A phase rotator, if used,
> should precede every variable-
> gain stage in the audio system.

This is, in fact, the location of the Orban phase rotators if the Optimod were used alone. Bypassing the Orban rotators and installing the PR-1's places the rotators first in line in the new system. The PR-1's also accomplish their task with less total phase shift. The Orban rotators utilize 540 degrees of rotation to accomplish their purpose while the PR-1's use only 360 degrees of rotation.

Refer in the 8100 Operating Manual to schematic drawing #60034, "SCHEMATIC, PCB, LEFT & RIGHT COMPRESSORS, CARD #3 & #4". Locate test point "TP1" at the top center of the drawing. To the left of that, locate connection "W" which feeds the panel meter.

To bypass the phase rotators, remove IC303 and IC403. (IC403 is in the same position as IC303, but located on Card 4). Connect a small jumper wire from TP1 to point "W" on Card 3 and on Card 4. Each of these connections passes through a plated-through hole, making placement of the jumper a simple matter. Simply solder the jumper into the two appropriate holes. If it is desired to return the 8100 to original condition at some future date, the jumpers can be removed and the chips replaced, leaving no permanent marks.

Notice that this change also removes the 30 Hz Highpass Filter from the circuit. The highpass function is already provided in the AUDIO PRISM, a second is unnecessary. You should not however, attempt to disable IC302B and IC402B. While they are no longer in the program flow, they continue to feed the Gate Control Sample at connection "V". This signal must continue to be present for the 8100 to operate properly.

OPERATION WITH THE ORBAN 8000

One of the most powerful but yet unfatiguing FM sounds possible today can be produced using the Orban 8000 limiter/stereo generator and two AUDIO PRISM's. Because of the near absence of discretionary adjustments on the 8000, set-up is simple, quick, and will produce a known good product.

After setting all other controls on the AUDIO PRISM's, set the outputs for equal level suitable to drive the telephone lines (STL's). Set the RELEASE TIME control on the 8000 fully counter-clockwise (limit only). Using music as a program source and with the METER switch in the G/R position, adjust the 8000's input attenuators for 3 dB of broadband gain reduction on <u>frequent</u> peaks. On <u>infrequent</u> peaks, the meter should indicate 5 dB of gain reduction.

Extensive field tests have shown that best results are obtained when the 8000 inputs are conscientiously maintained in this range. Less than 3 dB of gain reduction does not permit the 8000 sufficient range of control to maintain peak modulation. Program density and loudness will be lost. Greater than the 3 dB of gain reduction on frequent peaks will cause excessive action of the 8000's HFL (high frequency limiter). This will unneccessarily roll off high frequency response, noticeably dulling the program.

A <u>very</u> <u>significant</u> improvement in program quality and clarity is possible by replacing some of the integrated circuits in the 8000. The Texas Instruments BIFET series of operational amplifiers, which were not available when the 8000 was designed, are superior in many ways to those used in the original 8000 design. Later versions of the 8000 used the newer operational amplifiers in some locations.

These improvements can be completed in an afternoon and require less than $20 in parts. We <u>strongly</u> recommend them for all Orban 8000's to be used in conjunction with the AUDIO PRISM.

"RECHIPPING" THE ORBAN 8000

* * * * * NOTICE * * * * *

THE FOLLOWING CHANGES ARE NOT ENDORSED BY THE EQUIPMENT'S MANUFACTURER AND MAY AFFECT YOUR ABILITY TO HAVE FACTORY WORK PERFORMED ON YOUR UNIT.

[CONTINUED ON NEXT PAGE]

* * * * * NOTICE * * * * *

THE FOLLOWING CHANGES HAVE BEEN THOROUGHLY RESEARCHED AND TESTED. NUMEROUS ENGINEERS HAVE IMPLEMENTED THEM WITH EQUALLY SUCCESSFUL RESULTS. HOWEVER, TEXAR INCORPORATED SHALL IN NO MANNER BE LIABLE FOR DAMAGES, DIRECT, CONSEQUENTIAL, OR OTHERWISE, ARISING FROM USE OF THE FOLLOWING INFORMATION, WHETHER DUE TO MISUSE OR NEGLIGENCE ON THE PART OF EITHER PARTY.

Several schemes for upgrading the integrated circuits in the Orban 8000 have been published or circulated in the past few years. The following are informally known in many areas as the "Chris Hood modifications", named after the Pittsburgh-based designer of the STA-MAX™ who developed them.

PARTS REQUIRED:

- 8 TL071CP low-noise operational amplifier
- 2 TL072CP dual low-noise operational amplifier
- 2 capacitors, 0.1 uF, 25WVDC, Mylar
- 2 capacitors, 47 uF, 16 WVDC, electrolytic (See text)

The following table lists the device type of the amplifiers presently on the Limiter Board, and those of the devices that will replace them:

CIRCUIT PART NUMBER	OLD DEVICE TYPE NUMBER	NEW DEVICE TYPE NUMBER
IC201	301	TL071CP
IC202	301	"
IC203*	1556	"
IC204*	1556	"
IC205	709	"
IC206	709	"
IC209	4558	TL072CP
IC210	4558	"
IC211	709	TL071CP
IC212	709	"

*IC203 and IC204 need not be replaced if the high-pass filters have been bypassed (See page III-2 of the 8000 Operating Manual).

Refer to the diagram entitled "LIMITER BOARD ALIGNMENT POINTS" in the rear of the 8000 Operating Manual to locate the above devices. All of the replacement devices are plastic 8-pin DIP packages. It is possible some of the original devices to be replaced are metal cans with their leads formed to fit a standard 8-pin DIP pattern. Remove these as you would the other devices. Orient the replacement device using the index notch shown on the above mentioned diagram.

Removing components soldered directly to double-sided circuit boards can be difficult without the proper tools and knowledge. Only some 8000's were manufactured with sockets for all IC's. If yours was not, before attempting to remove components from the circuit board, familiarize yourself with the section entitled "Replacement Of Components in Printed Circuit Boards". It is on page V-4 of the 8000 Operating Manual.

Procede as follows:

[ ] Using a small pair of diagonal cutters, remove pins 1, 5 and 8 from each of the 8 TL071CP's. (Alternatively, one could simply bend them up so they will not make contact when inserted in the circuit).

[ ] Replace IC209 and IC210 with the two TL072CP's.

[ ] Using the remaining 8 IC's, replace IC201, IC202, IC203, IC204, IC205, IC206, IC211 and IC212.

[ ] Locate R263. Disconnect either end from the circuit board. Connect one of the mylar capacitors between the now vacant connection on the circuit board and the loose end of the resistor. There should now be a series RC network where the resistor was.

[ ] Locate R264. Disconnect either end from the circuit board and connect the other mylar capacitor as above.

These resistors sample the audio signal for the HFL stages. It is possible for a strong bass note, such as a bass drum, to "punch a hole" in the high-frequency band by falsely activating the HFL's. Decoupling the sample to the HFL's with the capacitors reduces this phenomenon.

[ ] Replace C605 and C606 with new units of the same value.

See following text for discussion of this step.

C605 and C606 are electrically in the power supply, but are mounted on the limiter board, just to the left of the two large heat sinks. While not specifically contained in the audio section, these capacitors can have a marked effect on audio performance.

Electrolytic capacitors sometimes deteriorate with age, particularly in the presence of higher ambient heat. Usually this is evidenced by a decrease in value as the electrolytic paste dries and becomes ineffective. Below a certain value, the capacitor no longer has sufficient filtering capacity to accomplish the task for which it was intended.

These particular capacitors, if their value changes significantly, can cause the power supply to oscillate. If this happens, a number of symptoms can appear, none of which would immediately indicate the underlying problem was in the power supply.

In a number of 8000's where the above rechipping procedure was performed, this power supply oscillation was observed. Replacing C605 and C606 with new units of the same type cured the oscillation.

The replacement of C605 and C606 in the rechipping procedure is new to the December 1985 revision of this manual. It was not present in earlier revisions.

VALID EVALUATION METHODS WHEN USING THE AUDIO PRISM IN CONJUNCTION WITH THE OPTIMOD 8100 OR THE OPTIMOD 8000

Naturally, when one purchases or contemplates purchasing a new piece of equipment, he (or she) would like to know the benefits that piece of equipment will produce for him (or her).

As dynamic acoustic performance is a very subjective matter, sometimes difficult to quantize in printed literature, and because the purchase price of a premium grade audio processor is not insignificant, additional user familiarization is frequently desirable.

Borrowing a page from the automobile industry, most audio processing equipment manufacturers and distributors will provide loaner units for evaluation, in much the same way as auto dealers provide a prospective customer with a car to "test drive".

As is the industry norm, TEXAR has such an evaluation program.

One logical method to determine the benefit which a piece of equipment produces (what engineers call the "delta function" and what business managers call the "marginal benefit") is to evaluate the system with the equipment in question in and then out of the circuit.

Care must be taken in how to structure such a test. There are two legitimate methods for evaluation, and one method which, although it appears to be valid, will produce erroneous results.

That method to be avoided is to connect the AUDIO PRISM's and the OPTIMOD as a system, and to then patch the AUDIO PRISM's in and out of the circuit. In a stereo system, this would naturally involve simultaneously inserting and then removing two patch cord plugs with each hand.

The impressions of such an A-B comparison are generally misleading. The fallacy in this comparison has to do with the fact that the human ear has a short term memory of less than a second. The time constants of the Optimod are on the order of several seconds. From the time the patches are removed or inserted, to the time that the Optimod time constants have adjusted to the new input conditions, several seconds will elapse.

The adaptive circuits in the Optimod will "feather the edges" of any abrupt changes, good or bad, making them less noticeable.

A lesser reason this method is not valid is the setting of the Optimod controls. As stated earlier, the control settings recommended for used on the Optimod in conjunction with AUDIO PRISM's are not appropriate for use with the 8100 alone.

Two comparison methods are valid. The most desirable one is to have two identical Optimod's configured in two separate systems. The first system should include an Optimod, adjusted for normal operation, sourced directly from the audio console outputs. The second system would feed the audio console outputs to the AUDIO PRISM's. Their outputs would be fed to the second Optimod, which is adjusted as described earlier in this manual.

The BNC outputs of the two Optimods would then be fed to a coaxial switch or relay, which would select between the two. In this configuration, neither Optimod input is changed as one switches from monitoring one system to the other. There is no "settling time" required for the Optimod time constants to stabilize, and the differences in performance heard are valid the instant the switch is changed.

This is an "ideal" situation; however, many stations do not have the luxury of having two identical Optimods.

The second valid method of evaluation is to use the competition as a reference point. Using this method, the Optimod is adjusted for its maximum performance while operating alone in the program chain. Several station personnel should compare the on-air signal to that of at least two other stations with similar music. They should make specific reference notes comparing relative loudness, spectral balance, clarity, and any other subjective quality which appears relevant.

It is important to use several different listening environments to gather this information. At a minimum, they should include a handheld portable with a small speaker, a high quality component system, and a car radio. Data from several additional car radios may be helpful.

Much of this first half of the evaluation may already be completed, as station personnel, in their daily market monitoring, are usually familiar with how the sound compares to that of other stations.

The AUDIO PRISM's are then installed in the program line between the console and the Optimod. They should be adjusted for proper operation and the Optimod controls reset to appropriate settings. The same station personnel then again compare the on-air sound with that of the same reference stations. That improvement between the original comparison to the competition and the second comparison is the benefit contributed by the AUDIO PRISMS.

There is a very small chance that one of the reference stations may make a change in its processing system during the course of your tests. In going from part one of this test to part two, if your on-air sound appears to have gained ground compared to one station, and lost ground with reference to another, one of the reference stations has changed, and the tests should be repeated.

This method of using other stations as reference points for comparison does not produce results as quickly as the two-Optimod method (several hours vs. several minutes); however, the results are equally valid. In fact, as the comparison between your on-air sound and that of the competition is what listeners will actually be witnessing, the results of this test may actually be more relevant.

BYPASS MODE OPERATION

The former PROOF MODE has been changed in function and is now called the BYPASS MODE.

The original AUDIO PRISM (units using the CX-1 PC board) PROOF mode was unlike that of most other discriminate processors. The PROOF mode froze the gain of all processors at their nominal levels, but kept all normal signal paths in the circuit. This was in contrast to the more common practice of driving the output amplifier directly from the input amplifier, bypassing all frequency-dependent circuits and VCA's. The underlying logic of the PROOF mode assumed that many circuit malfunctions could go undetected if they were not in the signal path when equipment performance measurements were performed.

Several years of experience with the AUDIO PRISM design have shown no known cases of internal circuits whose performance has degenerated. Over the same period, many instances have occurred of users misconnecting the patch panel while bypassing the AUDIO PRISM's for set-up purposes. Some users have inadvertently interrupted the program. Others have even gotten unintended signal sources on the air.

Further, many stations, including one major New York City FM, do not even have patch panels in their program lines, making the PINK NOISE set-up procedure almost impossible.

With this as background, the PROOF function has been changed to the more conventional BYPASS function, effective with the change to the CX-2 circuit board. The bandpass filter and the M-101 processor cards are now removed from the circuit when S102 is in the ON position. This permits the AUDIO PRISM to do two functions simultaneously: 1) adjust the spectral MIX LEVEL controls under pink noise conditions, while 2) maintaining the normal program flow from the studio to the limiting device which follows the AUDIO PRISM.

The gain of the BYPASS function is variable, allowing the user to adjust the BYPASS output level to be equal in level (but not in density) to the normal output level. This gain is controlled by R115. This control is located behind a small, unlabeled hole, located directly above the left threaded mounting for the escutcheon cover plate. Due to physical constraints, this control is recessed slightly more than an inch, making it not on an even plane with the rest of the front panel controls. Use a thin flat-blade screwdriver to access this control, such as the Xcelite "Greenie" which came with the AUDIO PRISM. While the control is not easily visible, a few seconds of experimentation will usually allow the user to seat the screwdriver in the adjustment slot. This control is a multi-turn potentiometer.

When in the BYPASS mode, only the INPUT GAIN control and R115 will have any effect on the output signal. The OUTPUT GAIN control is out of the circuit.

If your AUDIO PRISM is equipped with the EAGLE Intelligent Clipper, the EAGLE is also removed from the signal path in the BYPASS mode.

USE WITH OTHER EQUIPMENT

EXCITERS

A very few FM users have experienced difficulties with their exciters when installing the AUDIO PRISM's. In most cases, the AUDIO PRISM's were part of a general audio improvement program and were installed in conjunction with a composite clipper.

While all modern FM exciters employ a two-speed AFC loop, there are still some in use which have only a single-speed loop. When used with heavily processed audio, these exciters are sometimes unable to distinguish between a low bass note and a low-frequency AFC correction voltage.

A sudden bass note may unsettle the AFC loop causing a tearing sound on the air identical to that produced by multipath reception. In the case of multipath reception, the phase-locked-loop (PLL) in the receiver's detector is on the verge of unlocking. In this instance, it is the PLL in the exciter AFC which is on the verge of unlocking.

The only remedy short of changing the AFC circuit is to reduce the degree of processing.

The HARRIS MS-15 is one exciter with a single-speed AFC. Its look-alike successor, the MX-15, has a two-speed AFC. An update is available from HARRIS to upgrade the MS-15 to MX-15 specifications. This improvement should be implemented on any MS-15 which will be used in conjunction with the AUDIO PRISM and a composite clipper.

USE WITH OTHER EQUIPMENT

COMPOSITE CLIPPERS

The composite clipper is an additional piece of audio processing equipment which is becoming common in some markets. The composite clipper serves as an absolute final "brick wall" modulation limiting device going into the FM exciter. It has two uses which, while easy to lump together, are actually very different:

1) Compensate for "bounce" in composite STL's, and
2) Provide a small but noticeable increase in loudness of the on-air signal.

Four factors should enter into the decision whether to employ composite clipping or not:

1) Whether the station uses a composite STL,
2) Whether the station operates an SCA channel,
3) Station format, and
4) Degree of competition for loudness among other stations in the market.

All composite STL's exhibit "bounce". That is to say that three consecutive peaks of 100%, 100% and 100% delivered to the STL transmitter, may be returned as 95%, 105% and 100% at the output terminals of the STL receiver. The ratio of peak-to-average modulation levels is not as consistent at the output of an STL system as it is at the input. This can result in a small loss of modulation level.

This is not a criticism of the design of any one, or all, composite STL's. Rather, it is simply the physical consequence of attempting to pass a clipped waveform through a program channel with finite bandwidth.

The composite clipper, when installed between the STL receiver and the FM exciter, can restore the consistency to the ratio of peak-to-average modulation levels, and recover the lost modulation.

For proper operation, the composite clipper <u>must</u> be placed at the FM transmitter. Some stations have attempted to place the unit at the studio between the stereo generator and the STL transmitter. These installations have been counter-productive.

When used primarily as a loudness-enhancement device, the composite clipper, like many pieces of audio processing equipment, allows the broadcaster to make a small increase in the quantity of the audio while making a small compromise in the quality of the signal. When used in moderation, this capability is often desirable.

The word moderation is important. The following analogy is helpful:

> A few shakes of salt on corn-on-the-cob makes the corn more appealing. It adds a little bit of spark and excitement to the corn. But fifty shakes of salt does not make the corn ten times as appealing as did five shakes. In fact, fifty shakes would make the corn so distasteful that no one would want to eat it.

The same is true of composite clipping. A little may be desirable. A lot is worse than none at all.

One composite clipper is the model CP-803, manufactured by Modulation Sciences Incorporated (MSI), of Brooklyn, New York. This unit has two front panel indicators: a yellow LED labeled "Normal" and a red LED labeled "Heavy". When using this unit, we recommend adjustment so that the yellow LED is illuminated approximately 50% of the time and the red LED is never illuminated.

Composite clipping is more appropriate for some formats than for others. For those formats where high apparent loudness is important, composite clipping may be desirable. These formats include CHR (Contemporary Hit Radio), AOR (Album Oriented Rock), AC (Adult Contemporary) and Urban. Composite clipping is less likely to be appropriate for formats such as Classical or Easy Listening where faithful audio reproduction is of higher importance than apparent loudness. Composite clipping may or may not be appropriate for Country formats, depending on local market conditions.

Composite clippers generate harmonics of the clipped signal. For the station transmitting only their main channel program material, higher-frequency harmonics will fall harmlessly out of the passband of the listeners' receivers. But for the station implementing an SCA, some harmonics will fall within the SCA passband, resulting in main-to-subchannel crosstalk. This will be evidenced as a popping or frying sound in the SCA receiver.

The greater the degree of composite clipping, the greater the degree of crosstalk. How much crosstalk, if any, may be tolerated into the SCA subchannel depends on the program content in that subchannel. Computer data distribution services are less tolerant to crosstalk than are voices services, such as physicians' update services. Similarly, high speed data services (2400 and 4800 baud) are more sensitive than slow speed (300 and 1200 baud) services.

Lastly, the norm for the market should be considered when deciding whether to clip the composite and, if so, how much to clip. If your signal is head and shoulders above the rest, there may be little marginal benefit in composite clipping. Conversely, if most stations in the market are employing a significant amount of composite clipping, a station may have little choice but to attempt to keep up.

In all cases, adjustment of the composite clipper is critical.

THEORY OF OPERATION

The AUDIO PRISM was designed around the philosophy that future significant improvements in broadcast audio processing will have to do with quarter-hour maintenance.

Since the early 1960's, broadcasters have placed a high premium on ever-increasing modulation levels. Each successive generation of audio processor brought higher modulation levels. The time between successive generations also seems to have decreased significantly.

Higher modulation levels meant that the recovered audio at the receiver detector also had a higher value. This higher audio voltage could be used to more effectively mask noise at the receiver. This noise had three components: atmospheric noise over the propagation path, interference from other adjacent or co-channel stations, and thermionic noise within the receiver.

Regardless of the noise source, if one assumes it to be constant, and that the station can be comfortably listened to until the signal-to-noise ratio falls below a certain constant, if one doubles the modulating voltage on a given carrier, one can travel to the point where the carrier is only half as strong and still receive the signal.

[This linear relationship holds only for AM. A slightly more complex, non-linear relationship holds true for FM, although the end result is similar.]

The end effect of increasing modulation levels was to increase the outer fringe or distance to where the station could be comfortably heard. The greater this distance, the more people that were contained within the station's service area, and the greater the station's potential listening audience (cumes) was.

The primary purpose of audio processing prior to 1983 was to increase potential cumes.

Additionally, some broadcasters feel there is some psycological advantage or attraction to listeners to listen to the station which appeared to be the most powerful.

Today, modulation levels are only a percentage point or two away from their theoretical maximums. The room simply does not exist for significant additional increases in modulation levels. Stated another way: radio broadcasting is all cumed out.

The revenue-producing ability of a broadcast property is not directly related to the cume audience. Rather, it is nearly proportional to the average audience (frequently called "average-quarter-hour" audience).

Simple logic dictates that if we are to increase the <u>average</u> number of listeners while not increasing the <u>total</u> number of listeners, we must keep each listener for a <u>longer</u> period of time. The rating services, logically, call this parameter "time spent listening".

Much research has gone into the programming aspects of keeping listeners tuned-in for longer periods of time. While programmers constantly seek new positive ways to keep listeners interested, such as perhaps an improved music rotation, much scrutiny is also given to removing negative program elements, which tend to make listeners tune-out, such as perhaps an abrasive song. Time spent listening can be increased by removing program elements which listeners find fatiguing.

The AUDIO PRISM was designed with the intent of reducing the electronic fatigue factors which reduce time spent listening. Typically, these can be a brash or squashed sound, an un-natural sound, or similar processing artifacts. To accomplish this, the AUDIO PRISM employs <u>digital technology</u> in its control circuits. Previous generations of audio processors employed <u>analog</u> technology.

Analog processor could do two things: expand and compress. Digitally-controlled processors can do three things: expand, compress, and, if necessary, <u>do nothing</u>. While this may appear, on first inspection, to be a <u>small</u> point, it is not.

If the level is too high in an analog processor, the processor will compress. The moment the level is no longer too high, the analog processor will expand. Expand is simply the default value of what an analog processor does whenever it is not compressing.

A very accurate comparison is to observe what happens to a baseball thrown straight up in the air. After it has lost its reason to go up (the fact that you threw it), it does not simply float at that height until someone climbs a ladder and pushes it back down. As soon as it stops going up, it starts coming back down. Fall back to earth is simply the default value of what a baseball does when it is no longer going up.

The same is true of an analog processor. So long as the analog processor is fed a normal operating level, it is continually either increasing or decreasing gain. If it is not going up, then it is going down, even if it doesn't have a reason to. Up and down, up and down. It never rests. This translates into a "busy" sound and listener fatigue.

Imagine driving a car down a straight road, but the car has no steering wheel. Instead, it has a two position-switch. One position is labeled LEFT TURN and the other position is labeled RIGHT TURN. There is no center position marked STRAIGHT AHEAD. You must navigate your way down the road by flipping the switch back and forth from one position to the other.

The path the car will travel will be a wobbly one as you swerve left, then right, then left again trying to keep the car on the road. Without a center position labeled STRAIGHT AHEAD, you can only approximate a straight line by continually reversing your direction. At the end of the trip, any passengers in the car will have experienced considerably more movement than was necessary in their trip from point A to point B. Your passengers, like radio listeners, will eventually become fatigued by the continual (and needless) change in direction.

To increase loudness in an analog-controlled processor, one shortens the attack and recovery time constants. This increases the density, but makes the compression level go up and down all that much faster, and makes the signal more "busy" than it was before. Up goes listener fatigue and down goes time spent listening.

In an effort to prevent listener burnout, one can decrease the number of gain reversals in a given period by lengthening the time constants, but loudness suffers and the broadcaster has come full circle back to where he started from.

This is the classic "You can have loudness, or you can have quality, but you can't have both" dilemma that broadcasters have been running from since the modulation war was declared.

Using digital control, the broadcaster is no longer confined to having a default value of "expand". Now, the default value can be "maintain present gain". A digitally-controlled processor can be programmed to expand only when it has a reason. While maintaining an equal or higher modulation level, in any given period, the AUDIO PRISM will make only one-third to one-half as many AGC voltage reversals as will an analog processor.

The decision whether to expand, compress or do nothing is made on the M-101 processor card by an LM339N, CMOS comparator. Refer to drawing #8511, "M-101 DIGITALLY CONTROLLED AUDIO PROCESSOR". Two sections of the LM339N, U2B and U2C, form a window-comparator. When the voltage applied to pin 4 exceeds the compression reference voltage on pin 5, a compress action is initiated. When the voltage applied to pin 11 (which is connected to pin 4) falls below the expansion reference voltage on pin 10, an expand action is initiated. When the voltage at pins 4 and 11 is neither higher than that on pin 5 nor less than that on pin 10, no action is initiated and the processor remains at its present gain. This is known as being in the "dead band" or the permissible range.

One can increase the permissible range by increasing the spread between the voltages on pins 10 and 5. Similarly, for a given input signal on pins 11 and 4, increasing the voltage spread between pins 10 and 5 will decrease the number of decisions the circuit will make in a given period.

To be symmetrical, one might expect that one would move one reference voltage up while moving the other down. In the interest of simplicity, the voltage at pin 5 is held constant at 5.0 volts, while the voltage on pin 10 is moved up or down. The range of the voltage on pin 10 is from 2.5 to 4.5 volts.

The control which varies the voltage on pin 10 is the DENSITY control. Accordingly, it is a simple matter to change the character of the AUDIO PRISM with this control by changing the size of the window comparator's permissible area.

INTEGRATED CIRCUIT SUBSTITUTIONS

The AUDIO PRISM utilizes two families of TEXAS INSTRUMENTS Bi-fet op-amps: the TL071 family and the TL081 family. Each family has packages with one, two or four op-amps in them. The two families are pin-for-pin identical except that the TL071 family has a significantly lower signal to noise ratio. IC's of the TL071 family may be used in place of members of the noisier TL081 family, but not vice versa.

Some cross-reference sheets erroneously list National Semiconductor and Motorola equivalents to the TL071 family. These IC's are really equivalents for the TL081 family. Only Exar (Spelled like TEXAR without the "T") and Thompson-CSF (a European firm) manufacture exact replacements, and both carry the same TL071 family nomenclature as would the Texas Instruments part.

The table below lists permissible substitutions for op-amps in the AUDIO PRISM. Substitutions other than these may compromise the signal to noise ratio of the AUDIO PRISM.

NUMBER OF SECTIONS	TEXAS INSTRUMENTS NUMBER	NATIONAL NUMBER	MOTOROLA NUMBER
1	TL071	None	None
2	TL072	None	None
4	TL074	None	None
1	TL081	LF351	MC34001
2	TL082	LF353	MC34002
4	TL084	LF347	MC34004

The AUDIO PRISM uses many 4000-series CMOS devices. Most manufacturers choose to label their CMOS parts per the standard established by RCA, which developed the CMOS line. Motorola; however, chooses to add their customary "MC" prefix and a number "1" before the normal 4-digit JEDEC identifier. A Motorola MC14001 is equivalent to an RCA CD4001, a Motorola MC14050 is equivalent to an RCA CD4050, and so forth.

INTERNAL ADJUSTMENTS

For those who would seek to improve the performance of the AUDIO PRISM by experimenting with different time constants, note that the sealed trim pots inside the unit are not time constants. They are bias adjustments. Tampering with these will break the control loop of the processors and cause them to stop operating. For unusual applications where the installed time constants are very inappropriate, consult the factory directly for corrective action.

SEPARATE MUSIC AND VOICE PROCESSING SYSTEMS

A partial facility is implemented in the AUDIO PRISM to allow "post-mix voice injection". Some stations prefer to process their live announcer mike separately from their music and recorded sources. Typically, the end goal is to permit the use of a significant amount of equalization on the mike channel. If subsequently fed through a multi-band processor, the processor will tend to redistribute the spectral energy, negating the function of the equalizer.

The desired end goal is accomplished by feeding the mike channel into the program circuit at a point after the multi-band processing. Several stations have implemented this configuration, using the summing-point on the CX-1 board as the point to introduce the mike channel.

This was accomplished by connecting a fifth resistor (in addition to the existing four representing the four bands) to the inverting input of the summing amp. This practice has been formalized in the CX-2. A signal introduced at Pin 1 of connector Y103 on the CX-2 will be superimposed on the AUDIO PRISM's output.

The gain at this mixing port is determined by the value of R126. Lower values of R126 will produce higher gain. Solder pads are provided for R126 and its position is marked, although it is normally not inserted at the factory, to minimize the possibility of stray noise pickup.

A value of 33K is recommended for R126 if this facility is implemented, although the exact value is not critical. Values less than 10K are discouraged. If more gain appears necessary to establish the desired voice injection level, the level provided to the AUDIO PRISM should be increased.

As described earlier, this is a partial facility for post-mix injection, a convenience for those who would add this feature themselves to their AUDIO PRISM's. No provision has been made to get the post-mix signal into the AUDIO PRISM chassis. This task, as it did before, rests with the user making the modification.

The RF and transient protection integrity of the AUDIO PRISM should be preserved. In the past, the method of getting the signal into the chassis has been to "borrow" two (or one) of the STEREO STRAP lines on the rear panel. As few stations implement the stereo strap function, these conductors, with their attendant RF filters and surge protection, are usually available. Consult the schematics for the FP-1 and the MB-2. These conductors can be intercepted either at the cable connecting the FP-1 and MB-2, or on the bottom of the MB-2.

When done with reasonable care (!), implementing these post-mix injection modifications will not invalidate the AUDIO PRISM warranty.

In low-RF environments, acceptable performance may be had by bringing a single, unbalanced conductor into the AUDIO PRISM. The proper connection point is at the motherboard end of the cable which interconnects the CX-2 and the MB-2. A signal introduced on pin #1 is connected through to R126 on the CX-2. There are no traces on the MB-2 connecting to pin #1. This conductor "dead-ends" there.

In high-RF environments, a balanced input is recommended. A small balanced-to-unbalanced converter will have to be constructed for inside the AUDIO PRISM. This can be made from a TL071 and four resistors.

When implementing post-mix injection of the mike channel, in addition to the equalization, it is usually necessary to add some type of active processing to the mike channel. This processing would be located in the circuit after the mike preamp, but before it was fed to R126 in the AUDIO PRISM. Without separate processing of its own, the mike channel would sound underpowered when compared to the music channel. Multi-band processing is generally not required in the mike channel. A single-channel, fast limiter, and possibly some noise control circuitry, is generally all that is required. The Gain Brain and the Kepex by Valley People (Nashville, TN) are popular devices for these applications.

When implementing post-mix injection, the air staff will have little or no control of voice-over levels. As both the music and voice are processed prior to the summation point, which source dominates and by how much, is fixed by the output level of the voice channel limiter. For the station attempting to insure uniform voice-overs and air sound from a pre-occupied air staff, this can be a desired feature. For the station needing maximum flexibility for control by the air staff, this can be a hinderance.

One alternative is to place the announcer's mike fader in the line-level circuit between the output of the voice channel limiter and the post-mix input of the AUDIO PRISM.

Be certain that the sum of the voice injection and the outputs of the four bands do not cause clipping at the white, front panel, test point (TP205 on DB-2). The "Sum less than 20" rule of thumb described on Page 19 is not valid when using post-mix injection. The preferred method is to observe the waveform on TP205 with an oscilliscope.

There is one other consideration, not readily obvious, which must be kept in mind when contemplating separate voice channel processing. Similar provisions must be made on the back-up audio chain. More than one station has installed separate voice and music processing systems only to switch to the back-up system many months later, and wonder why the announcer's voice was not coming over the air.

Additionally, the voice post-mix input is not operational when the AUDIO PRISM is in BYPASS MODE. When doing pink-noise set-up on the AUDIO PRISM's where separate voice and music processing is employed, it will be necessary to switch to the back-up processing system, or confine adjustment to periods when no live announcing is necessary.

Separate channel processing is an involved procedure which affects daily operations in an ongoing fashion. These can be as significant as the hardware considerations. Generally, it is warranted only in larger markets when the operational inconveniences are outweighed by the larger potential rewards of the market.

APPENDIX A

The following is a tabulation of MIX LEVEL settings which have been used successfully in the field. The readings were taken with a Simpson 260 VOM. A Potomac Instruments AA-51 Audio Analyzer may be used in place of the Simpson 260.

THE FOLLOWING READING ARE IN dBm, NOT VOLTS!

USE THE INNERMOST SCALE ON THE SIMPSON 260.
USE THE OUTERMOST SCALE ON THE POTOMAC INSTRUMENT AA-51.

	LOW BAND	MID BAND	PRESENCE BAND	HIGH BAND
ORBAN 8100A *	4.0	4.0	4.0	3.0
ORBAN 8000(A)	4.0	4.0	1.0	-3.0
AM (Plate modulated)	0.0	0.0	7.0	11.0
AM (Not plate modulated)	1.0	1.0	7.0	10.0

For a discussion of setting the MIX LEVEL controls, see the text on pages 15 through 19, paying particular attention to the exceptions noted on page 17.

(CONTINUED ON NEXT PAGE)

FM users should also see the text on page 22, paying particular attention to the comments regarding composite clippers.

The above AM settings include pre-emphasis and assume the use of the TEXAR EAGLE™ AM Modulation Controller. For use with modulation controllers which have integral pre-emphasis, such as the TEXAR SUPER EAGLE™ or TEXAR DOUBLE EAGLE™, set the AUDIO PRISM for flat (4,4,4,4). Use the pre-emphasis available on the modulation controller.

TEXAR AUDIO PRISM™ ADJUSTMENT LOG

DATE AND TIME	MIX LEVELS				OTHER COMMENTS	INIT IALS
	LOW BAND	MID BAND	PRES BAND	HIGH BAND		

TEXAR USER SUGGESTION REPORT

All suggestions of how we may improve TEXAR products or TEXAR USERS' MANUALS are appreciated. The blanks below for your name, address and telephone number are optional. In the event we have additional questions regarding your suggestion, these enable us to contact you for more information on your application or idea. If for some reason you prefer not to tell us your name, please leave these blank. All suggestions, blind or otherwise, receive the same thoughtful consideration from our engineering department.

YOUR NAME:____

ADDRESS:____

____

____

TELEPHONE NUMBER WHERE YOU CAN
BE REACHED DURING THE DAY (______) ____________________

TEXAR PRODUCT YOUR SUGGESTION CONCERNS:______________________

THIS SUGGESTION PERTAINS TO: [ ] The product

[ ] The Users' Manual

****

TELL US HOW YOU FEEL WE CAN MAKE THE PRODUCT OR ITS MANUAL BETTER. YOU MAY USE ADDITIONAL SHEETS IF NEEDED:

TEXAR EQUIPMENT MALFUNCTION REPORT

CUSTOMER NAME (USUALLY A RADIO STATION CALL SIGN): ______________ R.M.A. NUMBER ________

ADDRESS TO WHICH REPAIRED EQUIPMENT SHOULD BE RETURNED:
(ZIP CODE IS ESSENTIAL. EQUIPMENT CANNOT BE SHIPPED TO A POST OFFICE BOX NUMBER. ALL EQUIPMENT IS RETURNED BY U.P.S. SURFACE UNLESS OTHERWISE REQUESTED.)

____

____

____

YOUR NAME (PLEASE PRINT): ______________________________

PHONE NUMBER WHERE YOU CAN BE REACHED DURING THE DAY:

(_____) ________________________

****

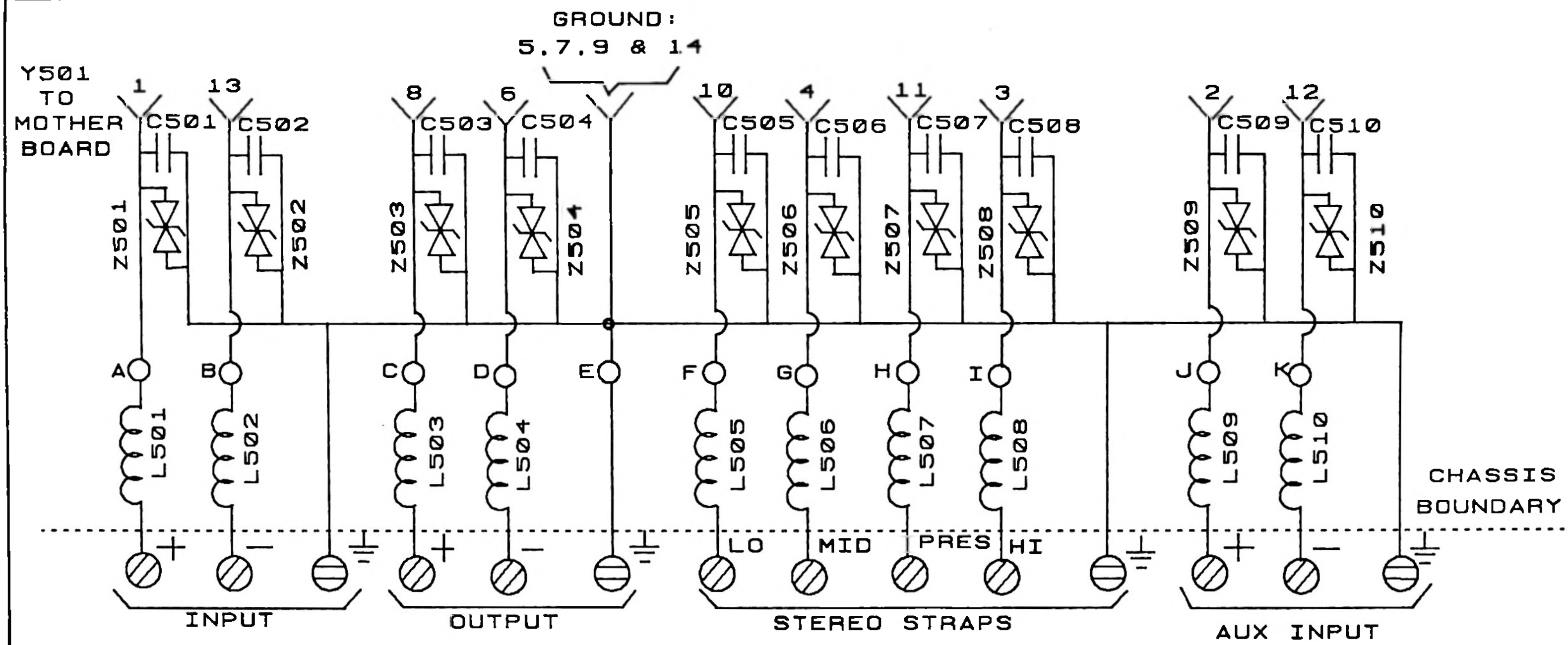
DEVICE OR CIRCUIT BOARD TYPE RETURNED ______________________

IF CIRCUITBOARD, REMOVED FROM WHAT TYPE TEXAR PRODUCT ______________________

SERIAL NUMBER OF PRODUCT ______________________

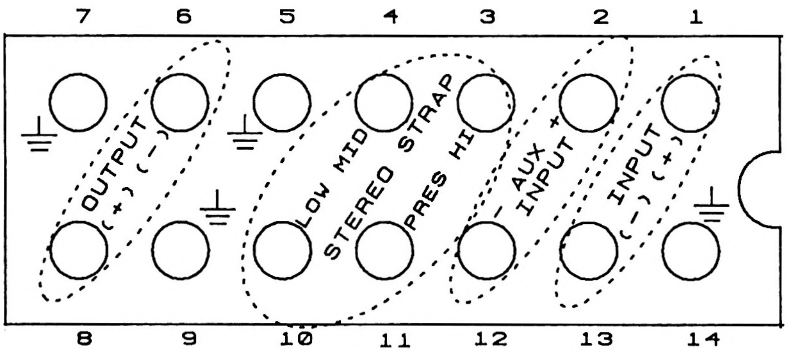
PLEASE DESCRIBE COMPLETELY THE PROBLEM YOU ARE HAVING WITH THIS EQUIPMENT. THE TECHNICIAN WHO REPAIRS YOUR UNIT MAY NOT BE THE SAME TECHNICIAN YOU SPOKE WITH ON THE TELEPHONE. TO INSURE THE FASTEST POSSIBLE RETURN OF YOUR EQUIPMENT, PLEASE DO NOT ASSUME THE TECHNICIAN IS ALREADY FAMILIAR WITH YOUR PROBLEM:

ADDITIONAL INFORMATION WHICH YOU FEEL MAY BE HELPFUL IN TROUBLE-SHOOTING YOUR EQUIPMENT:



TB501 - REAR APRON BARRIER STRIP

NOTE:
FULL IMPLEMENTATION SHOWN. SOME MODELS MAY USE LESS THAN THE FULL PROTECTION SHOWN.

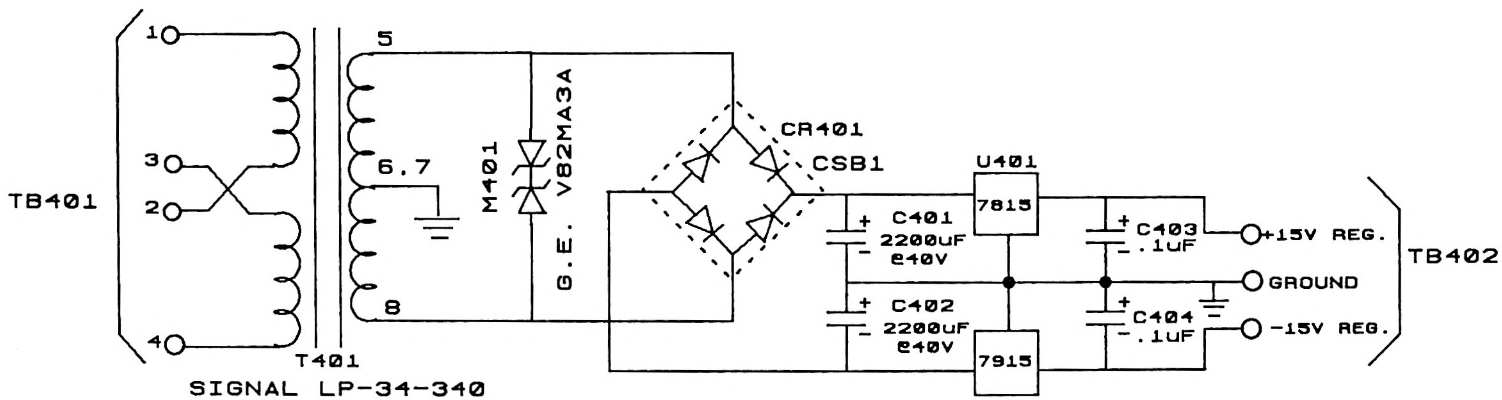


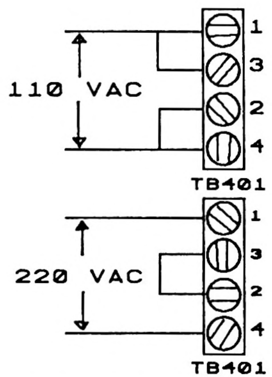
Y501
FP-1 <--> MB-1
INTERCONNECT
JUMPER

TEXAR
MONROEVILLE, PA 15146-1502

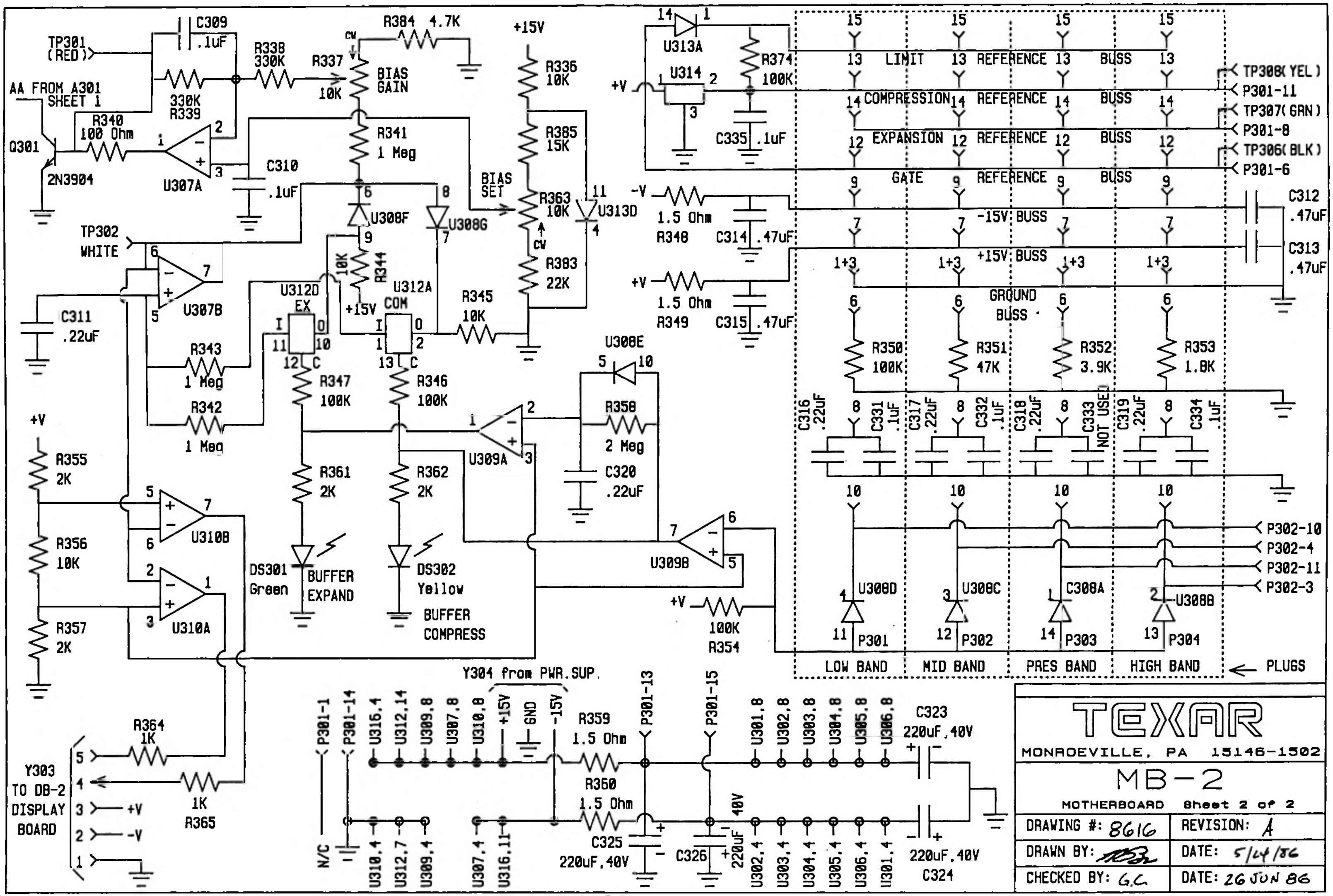
FP-1
FILTER AND PROTECTION BOARD

DRAWING #:	REVISION: A
DRAWN BY: MB	DATE:
CHECKED BY:	DATE: 8 MAR

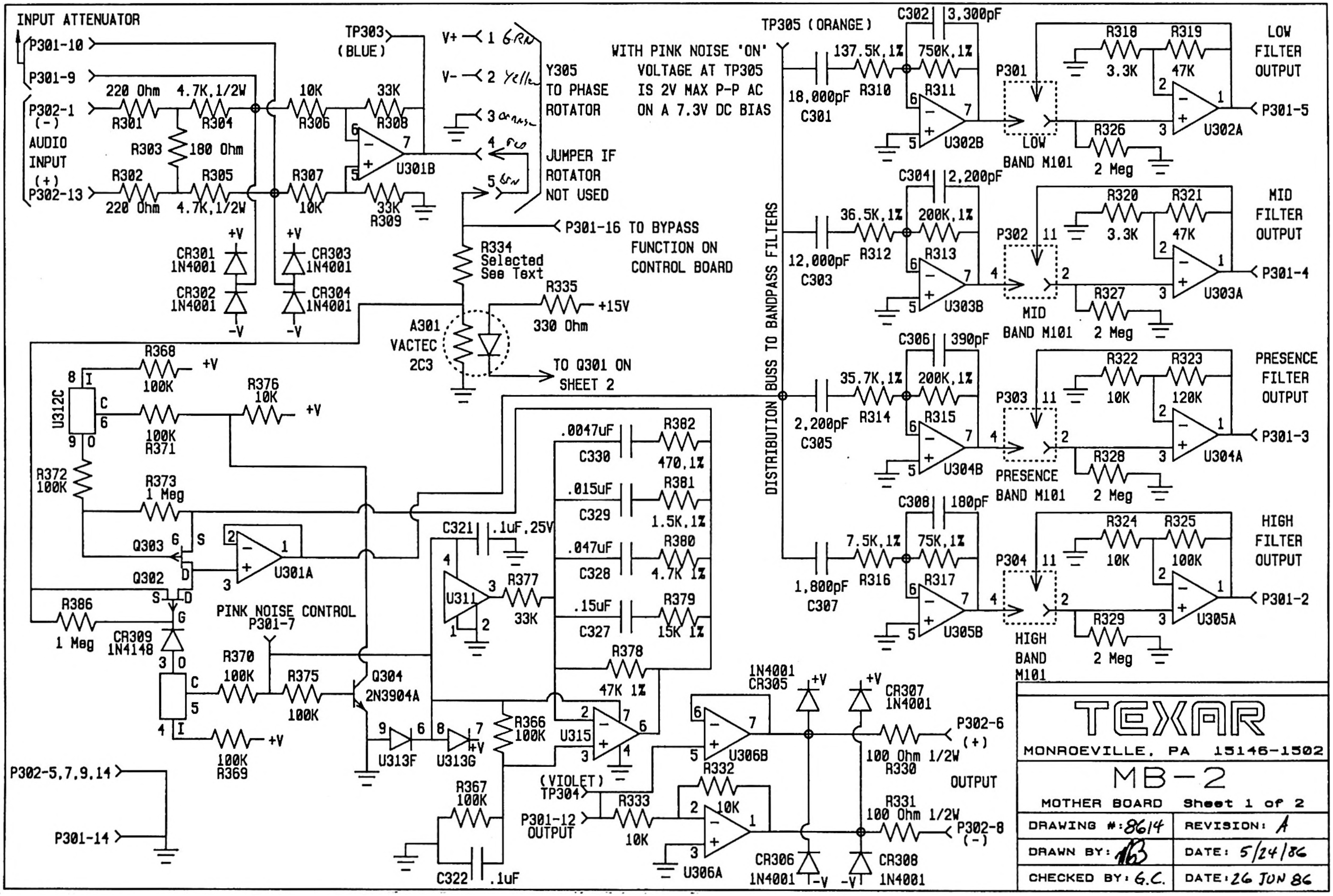




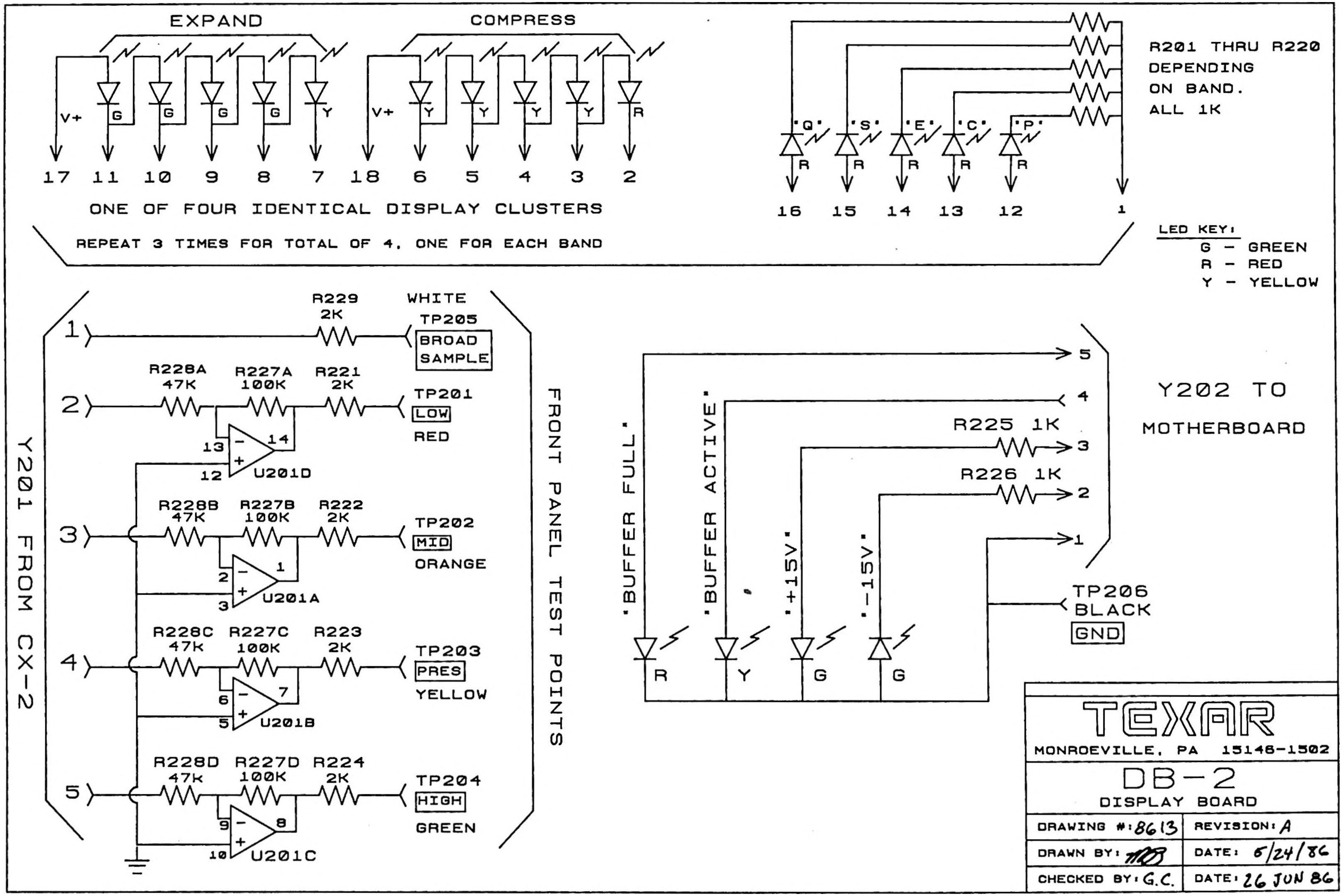
TEXAR	
MONROEVILLE, PA 15146-1502	
PS-1	
POWER SUPPLY	
DRAWING #: 8605	REVISION: A
DRAWN BY: MB	DATE: 4-7-86
CHECKED BY: GC	DATE: 8 MAR 86



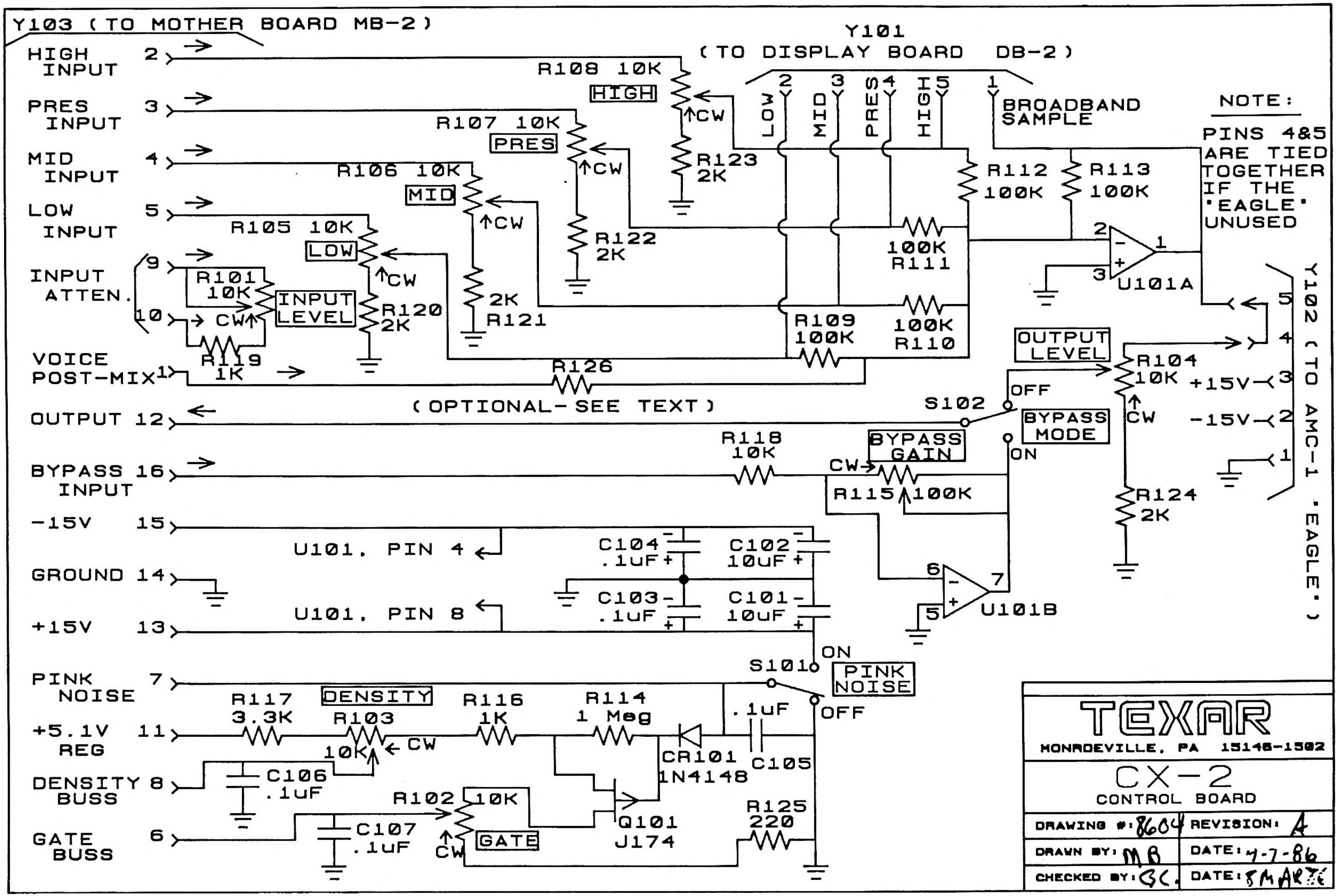
TP301 (RED)
AA FROM A301 SHEET 1
Q301 2N3904
C309 .1uF
R338 330K
R339 330K
R340 100 Ohm
U307A
C310 .1uF
R384 4.7K
R337 10K BIAS GAIN
R341 1 Meg
U308F
U308G
R344 10K
+15V
U312D EX
U312A COM
R345 10K
R347 100K
R346 100K
R336 10K
R385 15K
R363 10K BIAS SET
R383 22K
U313D
U313A
U314
R374 100K
C335 .1uF
TP302 WHITE
U307B
C311 .22uF
R343 1 Meg
R342 1 Meg
U309A
U308E
R358 2 Meg
C320 .22uF
R361 2K
R362 2K
DS301 Green BUFFER EXPAND
DS302 Yellow BUFFER COMPRESS
R355 2K
R356 10K
R357 2K
U310B
U310A
U309B
+V 100K R354
-V 1.5 Ohm R348
C314 .47uF
+V 1.5 Ohm R349
C315 .47uF
LIMIT REFERENCE BUSS
COMPRESSION REFERENCE BUSS
EXPANSION REFERENCE BUSS
GATE REFERENCE BUSS
-15V BUSS
+15V BUSS
GROUND BUSS
TP308 (YEL) P301-11
TP307 (GRN) P301-8
TP306 (BLK) P301-6
C312 .47uF
C313 .47uF
R350 100K
R351 47K
R352 3.9K
R353 1.8K
C316 .22uF C331 .1uF
C317 .22uF C332 .1uF
C318 .22uF C333 NOT USED
C319 .22uF C334 .1uF
P302-10
P302-4
P302-11
P302-3
U308D U308C U308A U308B
P301 P302 P303 P304
LOW BAND
MID BAND
PRES BAND
HIGH BAND
PLUGS
Y303 TO DB-2 DISPLAY BOARD
R364 1K
R365 1K
Y304 from PWR. SUP.
P301-1
P301-14
U316.4 U312.14 U309.8 U307.8 U310.8
+15V GND -15V
R359 1.5 Ohm
R360 1.5 Ohm
P301-13
P301-15
U301.8 U302.8 U303.8 U304.8 U305.8 U306.8
C323 220uF, 40V
C324 220uF, 40V
C325 220uF, 40V
C326 220uF 40V
N/C
U310.4 U312.7 U309.4 U307.4 U316.11
U302.4 U303.4 U304.4 U305.4 U306.4 U301.4
TEXAR
MONROEVILLE, PA 15146-1502
MB-2
MOTHERBOARD Sheet 2 of 2
DRAWING #: 8616
REVISION: A
DRAWN BY:
DATE: 5/14/86
CHECKED BY: G.C.
DATE: 26 JUN 86



INPUT ATTENUATOR
P301-10
P301-9
P302-1 (-)
AUDIO INPUT
(+) P302-13
R301 220 Ohm
R302 220 Ohm
R303 180 Ohm
R304 4.7K,1/2W
R305 4.7K,1/2W
CR301 1N4001
CR302 1N4001
CR303 1N4001
CR304 1N4001
R306 10K
R307 10K
R308 33K
R309 33K
U301B
TP303 (BLUE)
V+ 1 GRN
V- 2 Yellow
3 orange
4 Red
5 Grn
Y305 TO PHASE ROTATOR
JUMPER IF ROTATOR NOT USED
P301-16 TO BYPASS FUNCTION ON CONTROL BOARD
R334 Selected See Text
R335 330 Ohm +15V
A301 VACTEC 2C3
TO Q301 ON SHEET 2
WITH PINK NOISE 'ON' VOLTAGE AT TP305 IS 2V MAX P-P AC ON A 7.3V DC BIAS
TP305 (ORANGE)
DISTRIBUTION BUSS TO BANDPASS FILTERS
U312C
R368 100K +V
R371 100K
R376 10K +V
R372 100K
R373 1 Meg
Q303
Q302
U301A
R386 1 Meg
CR309 1N4148
PINK NOISE CONTROL P301-7
R370 100K
R375 100K
R369 100K +V
Q304 2N3904A
U313F
U313G
C321 .1uF,25V
U311
R377 33K
C330 .0047uF
R382 470,1%
C329 .015uF
R381 1.5K,1%
C328 .047uF
R380 4.7K 1%
C327 .15uF
R379 15K 1%
R378 47K 1%
R366 100K
U315
R367 100K
C322 .1uF
(VIOLET) TP304
P301-12 OUTPUT
R333 10K
R332 10K
U306A
U306B
1N4001 CR305
CR306 1N4001
CR307 1N4001
CR308 1N4001
R330 100 Ohm 1/2W
R331 100 Ohm 1/2W
P302-6 (+)
OUTPUT
P302-8 (-)
P302-5,7,9,14
P301-14
C301 18,000pF
R310 137.5K,1%
C302 3,300pF
R311 750K,1%
U302B
P301
LOW BAND M101
R318 3.3K
R319 47K
R326 2 Meg
U302A
LOW FILTER OUTPUT
P301-5
C303 12,000pF
R312 36.5K,1%
C304 2,200pF
R313 200K,1%
U303B
P302
MID BAND M101
R320 3.3K
R321 47K
R327 2 Meg
U303A
MID FILTER OUTPUT
P301-4
C305 2,200pF
R314 35.7K,1%
C306 390pF
R315 200K,1%
U304B
P303
PRESENCE BAND M101
R322 10K
R323 120K
R328 2 Meg
U304A
PRESENCE FILTER OUTPUT
P301-3
C307 1,800pF
R316 7.5K,1%
C308 180pF
R317 75K,1%
U305B
P304
HIGH BAND M101
R324 10K
R325 100K
R329 2 Meg
U305A
HIGH FILTER OUTPUT
P301-2
TEXAR
MONROEVILLE, PA 15146-1502
MB-2
MOTHER BOARD Sheet 1 of 2
DRAWING #: 8614
REVISION: A
DRAWN BY: HG
DATE: 5/24/86
CHECKED BY: G.C.
DATE: 26 JUN 86

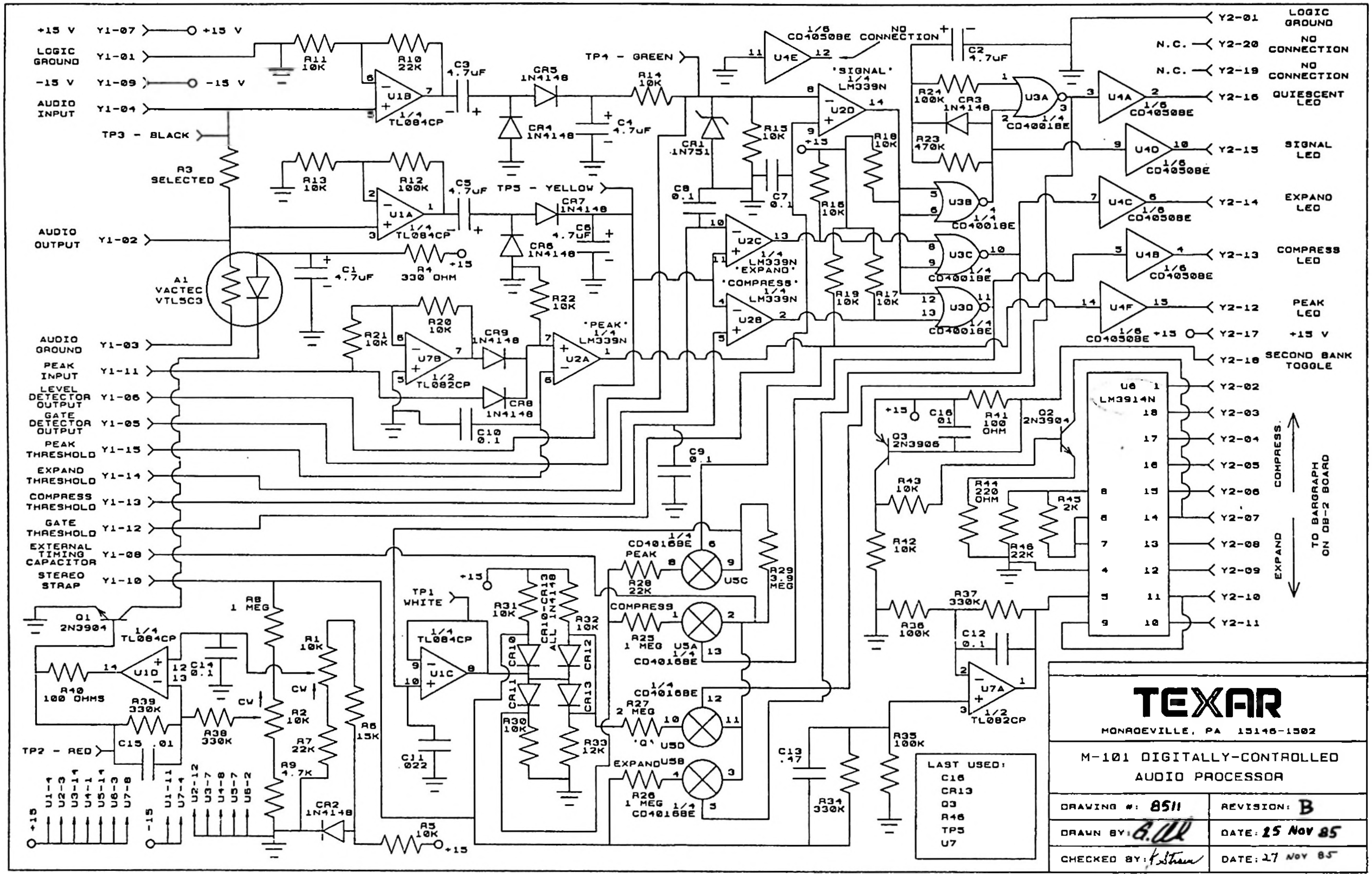


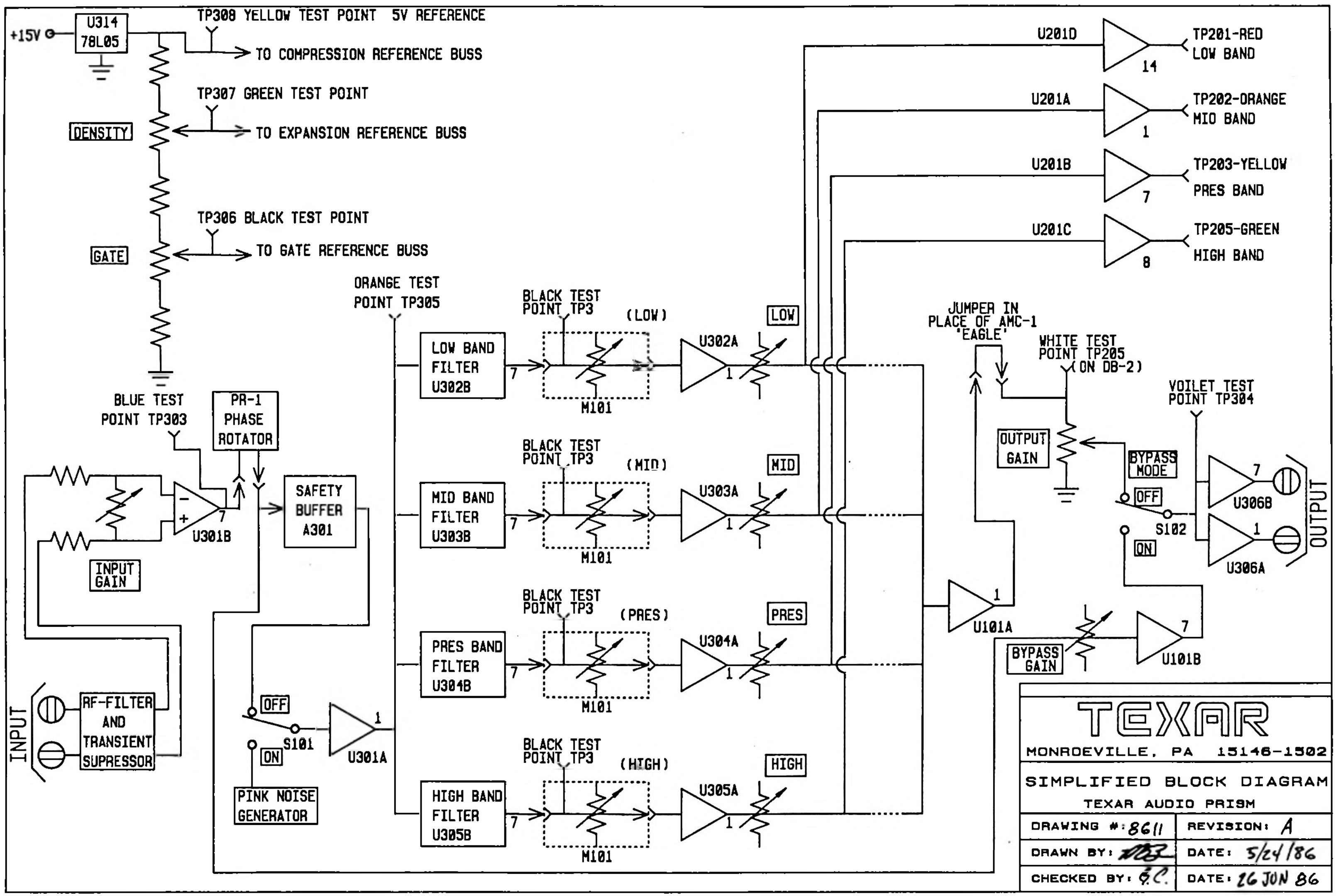
EXPAND
COMPRESS
V+
G
Y
R
17 11 10 9 8 7 18 6 5 4 3 2
ONE OF FOUR IDENTICAL DISPLAY CLUSTERS
REPEAT 3 TIMES FOR TOTAL OF 4, ONE FOR EACH BAND
'Q' 'S' 'E' 'C' 'P'
16 15 14 13 12 1
R201 THRU R220
DEPENDING
ON BAND.
ALL 1K
LED KEY:
G - GREEN
R - RED
Y - YELLOW
R229 2K
WHITE
TP205
BROAD SAMPLE
R228A 47K
R227A 100K
R221 2K
TP201
LOW
RED
U201D
R228B 47K
R227B 100K
R222 2K
TP202
MID
ORANGE
U201A
R228C 47k
R227C 100K
R223 2K
TP203
PRES
YELLOW
U201B
R228D 47k
R227D 100K
R224 2K
TP204
HIGH
GREEN
U201C
Y201 FROM CX-2
FRONT PANEL TEST POINTS
'BUFFER FULL'
'BUFFER ACTIVE'
'+15V'
'-15V'
R225 1K
R226 1K
TP206
BLACK
GND
Y202 TO
MOTHERBOARD
TEXAR
MONROEVILLE, PA 15146-1502
DB-2
DISPLAY BOARD
DRAWING #: 8613
REVISION: A
DRAWN BY:
DATE: 6/24/86
CHECKED BY: G.C.
DATE: 26 JUN 86

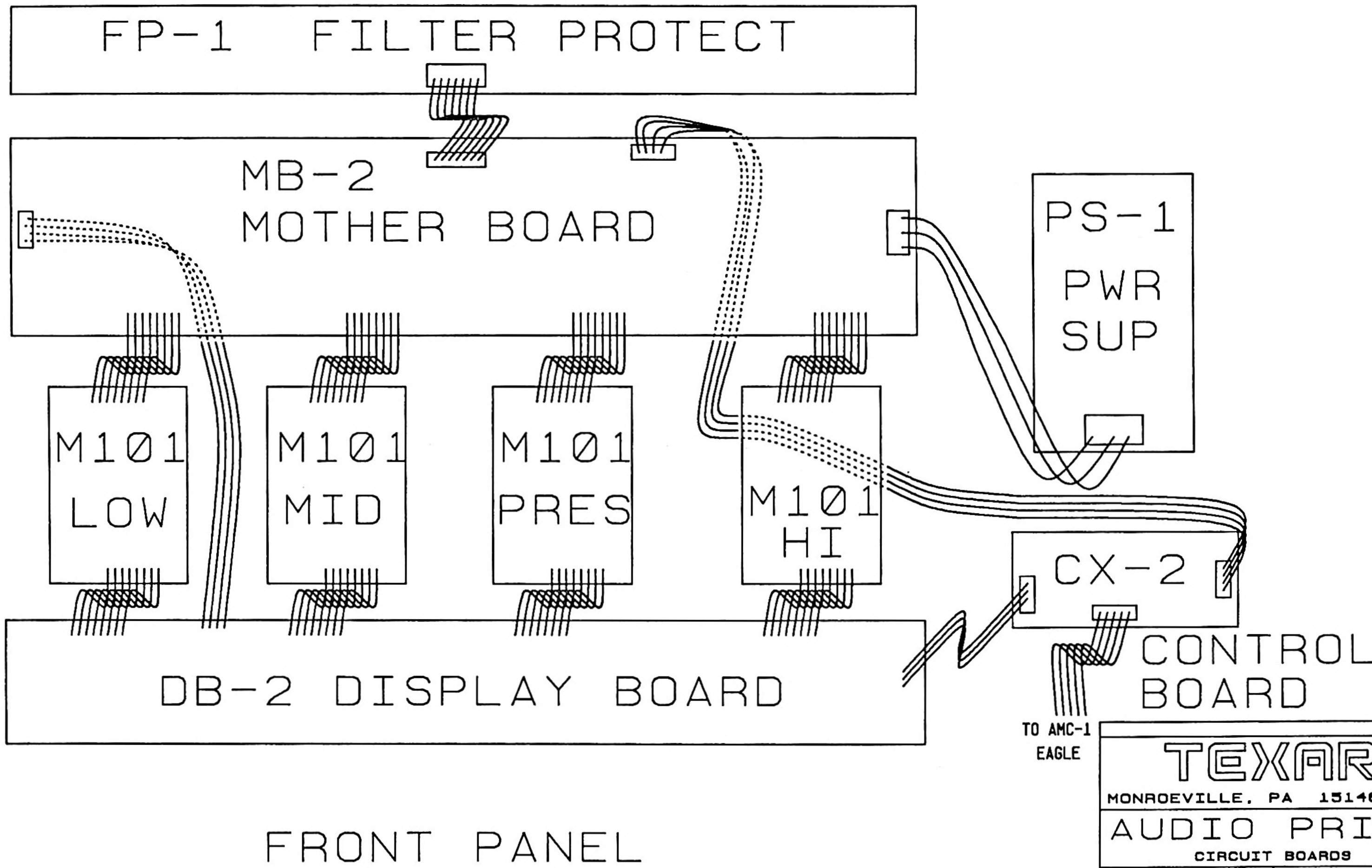

Y103 (TO MOTHER BOARD MB-2)
HIGH INPUT 2
PRES INPUT 3
MID INPUT 4
LOW INPUT 5
INPUT ATTEN. 9 10
VOICE POST-MIX 1
OUTPUT 12
BYPASS INPUT 16
-15V 15
GROUND 14
+15V 13
PINK NOISE 7
+5.1V REG 11
DENSITY BUSS 8
GATE BUSS 6
Y101
(TO DISPLAY BOARD DB-2)
LOW 2
MID 3
PRES 4
HIGH 5
BROADBAND SAMPLE 1
NOTE:
PINS 4&5 ARE TIED TOGETHER IF THE "EAGLE" UNUSED
R108 10K
HIGH
R123 2K
R107 10K
PRES
R122 2K
R106 10K
MID
2K R121
R105 10K
LOW
R120 2K
R101 10K
INPUT LEVEL
R119 1K
CW
R126
R109 100K
R110 100K
R111 100K
R112 100K
R113 100K
U101A
(OPTIONAL- SEE TEXT)
OUTPUT LEVEL
R104 10K
R124 2K
S102
OFF
ON
BYPASS MODE
R118 10K
BYPASS GAIN
R115 100K
U101B
+15V
-15V
Y102 (TO AMC-1 "EAGLE")
U101, PIN 4
U101, PIN 8
C104 .1uF
C102 10uF
C103 .1uF
C101 10uF
S101
PINK NOISE
R117 3.3K
DENSITY
R103 10K
R116 1K
R114 1 Meg
.1uF C105
CR101 1N4148
C106 .1uF
C107 .1uF
R102 10K
GATE
Q101 J174
R125 220
TEXAR
MONROEVILLE, PA 15146-1502
CX-2
CONTROL BOARD
DRAWING #: 8604
REVISION: A
DRAWN BY: MB
DATE: 4-7-86
CHECKED BY:
DATE:

Pin 7, rear connector

LM3914N




+15V
U314
78L05
TP308 YELLOW TEST POINT 5V REFERENCE
TO COMPRESSION REFERENCE BUSS
TP307 GREEN TEST POINT
DENSITY
TO EXPANSION REFERENCE BUSS
TP306 BLACK TEST POINT
GATE
TO GATE REFERENCE BUSS
ORANGE TEST POINT TP305
BLUE TEST POINT TP303
PR-1 PHASE ROTATOR
SAFETY BUFFER A301
U301B
INPUT GAIN
INPUT
RF-FILTER AND TRANSIENT SUPRESSOR
OFF
ON
S101
U301A
PINK NOISE GENERATOR
LOW BAND FILTER U302B
MID BAND FILTER U303B
PRES BAND FILTER U304B
HIGH BAND FILTER U305B
BLACK TEST POINT TP3
(LOW)
(MID)
(PRES)
(HIGH)
M101
U302A
U303A
U304A
U305A
LOW
MID
PRES
HIGH
U201D
U201A
U201B
U201C
TP201-RED LOW BAND
TP202-ORANGE MID BAND
TP203-YELLOW PRES BAND
TP205-GREEN HIGH BAND
JUMPER IN PLACE OF AMC-1 'EAGLE'
WHITE TEST POINT TP205 (ON DB-2)
OUTPUT GAIN
BYPASS MODE
VOILET TEST POINT TP304
S102
U306B
U306A
OUTPUT
U101A
U101B
BYPASS GAIN
TEXAR
MONROEVILLE, PA 15146-1502
SIMPLIFIED BLOCK DIAGRAM
TEXAR AUDIO PRISM
DRAWING #: 8611
REVISION: A
DRAWN BY:
DATE: 5/24/86
CHECKED BY:
DATE: 26 JUN 86



TEXAR
MONROEVILLE, PA 15146-1502

AUDIO PRISM
CIRCUIT BOARDS

DRAWING #: 8612 REVISION: A
DRAWN BY: [illegible] DATE: 6/24/86
CHECKED BY: G.C. DATE: 26 JUN 86

USERS' MANUAL

TEXAR EAGLE™

AM MODULATION CONTROLLER

APRIL 1985

REVISED APRIL 1986

REVISED JULY 1986

TEXAR
TEXAR INCORPORATED
616 BEATTY ROAD
MONROEVILLE, PA 15146-1502

(412) 856-4276

(412) 85-MICRO

USERS' MANUAL - TEXAR EAGLE™

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PARTS LIST	
SCHEMATIC	

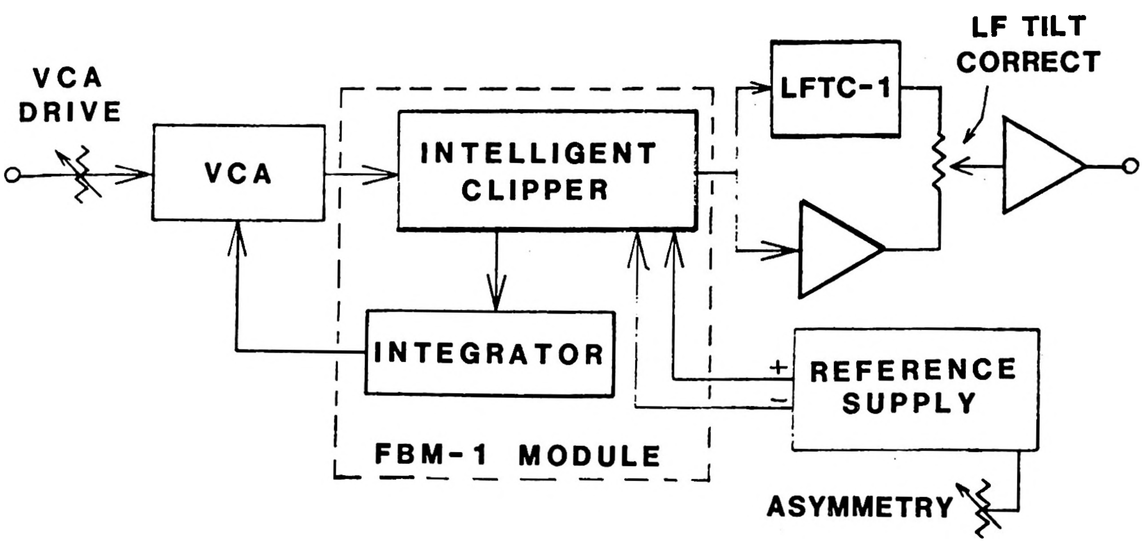
DESCRIPTION

The TEXAR EAGLE™ AM Modulation Controller is a high-quality final amplitude controller for use in AM broadcasting. It is an option to the TEXAR AUDIO PRISM™ Digitally Controlled Audio Processor. Together they comprise a complete monaural AM audio processing system.

The EAGLE includes a low-frequency tilt correction circuit to compensate for weaknesses in older transmitters, and variable asymmetry. LED's indicate the amount of instantaneous gain reduction and the degree of clipping action.

The EAGLE employs an Intelligent Clipper circuit; a hard clipper preceded by a VCA. An integrator monitors the current through the clipper diodes. If the conduction duty cycle becomes large enough that objectional distortion would result, the integrator decreases the control voltage to the VCA, which reduces the level fed to the clipper. This maintains the amount of clipping at a constant level just below where it would detract from the program material. Experience has shown that this arrangement produces denser modulation, with higher quality, than any other configuration available.

The block diagram of the EAGLE is shown below.



Reference voltages for the clipper are derived from a temperature-compensated National LM336Z Precision Voltage Reference. All voltages remain constant over a wide range of temperatures, insuring precision modulation control even in unheated transmitter buildings.

INSTALLATION

The TEXAR EAGLE is an option on the AUDIO PRISM™ and mounts inside the same enclosure, drawing bipolar supply voltages from the AUDIO PRISM. The EAGLE is mounted beneath the chassis cover. Access to the EAGLE's controls and indicators is through holes in this cover.

Modern, high-performance, AM clippers produce output waveforms with closely-controlled amplitude and phase parameters for maximum modulation. Any device connected between the clipper and the transmitter which disturbs the amplitude or phase relationships will also degrade the modulation performance. Transformers, including telephone repeat coils, are prime offenders at disturbing the phase response. For this reason, the EAGLE <u>must</u> be mounted at the transmitter; it cannot feed it through an STL or telephone lines. The TEXAR EAGLE is therefore intended primarily for the AM stations with co-located studios and transmitter. For split-site operation, the TEXAR SUPER EAGLE™ is preferred.

Mechanical and electrical considerations of installing the AUDIO PRISM are explained in the AUDIO PRISM USERS' MANUAL. The only additional considerations when using the EAGLE clipper will be protection of the adjustment access holes and the length of the input and output audio cables. Sufficient slack should exist that the unit can be slid out from the rack to expose the holes for the EAGLE adjustments. The unit should also be mounted so that foreign objects cannot fall through the access holes into the circuits below. In extremely hostile environments, tape may be placed over the holes when access to them is not required.

USE OF MODULATION ENHANCERS

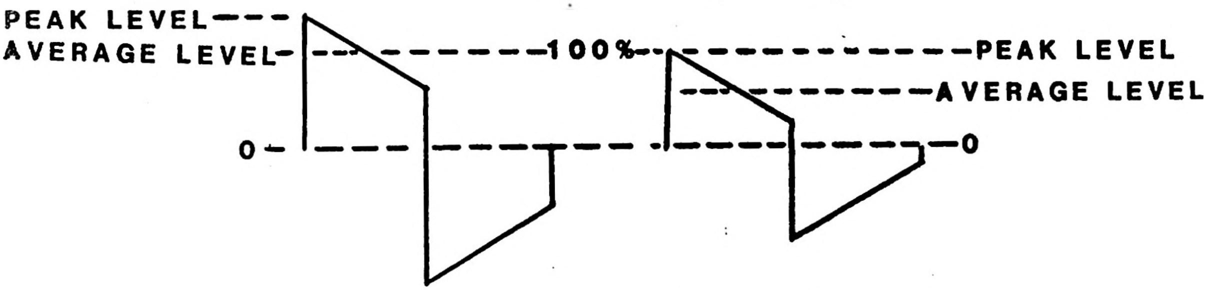
Any integral clipper in the transmitter should be disabled for proper operation of the EAGLE.

For the Harris MW-1, MW-5, MW-10 and MW-50, swing open the access door exposing the PDM circuit board. Place the toggle switch on the left wall of the exposed cavity in the "BYPASS" position. For the Continental 315R ("POWER ROCK") remove the service cover to the exciter card cage. Locate the PWM driver board. Move the IPL (Instantaneous Peak Limiter) toggle to the "off" position.

LOW FREQUENCY TILT CORRECTION

The LF TILT CORRECTION circuit alters the EAGLE's output in a way that compensates for the inaccurate waveform reproduction of some transmitters. The circuit is variable because some transmitters are more inaccurate than others. In the fully counterclockwise position of the LF TILT CORRECTION control, the circuit is out of the signal path, and there is no compensating action. Maximum compensation is obtained in the fully clockwise position.

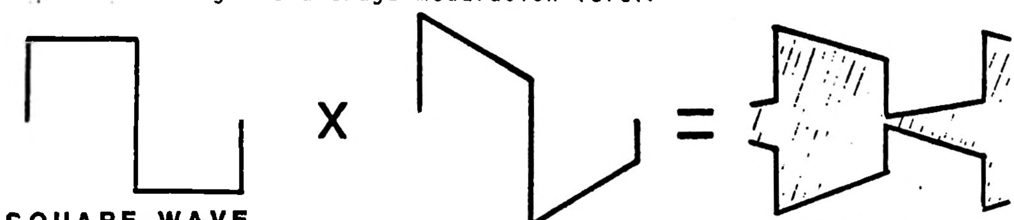
As shown below, where LF tilt is present, the leading edge of the output square wave has an amplitude substantially higher than that of its average value. As a result, the amplitude of the input signal must be reduced substantially to prevent overmodulation. This reduces a station's average modulation.



OVERMODULATION DUE TO LF TILT

LISTENER AREA LOST BECAUSE LEVEL REDUCED TO PREVENT OVERMODULATION

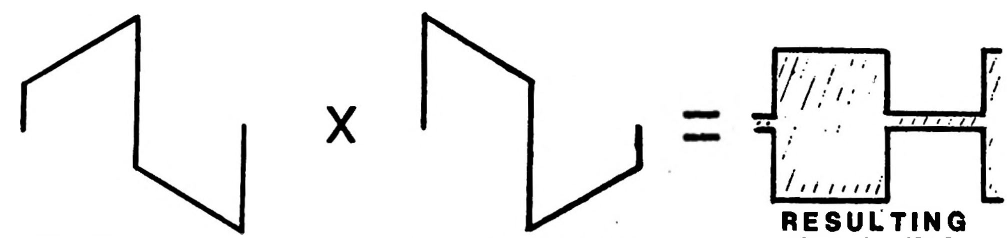
The Low-Frequency Tilt Correction circuit compensates for this transmitter weakness by predistorting the signal as shown below. It tilts the plateau of the input wave in the direction opposite that of the transmitter's response. The two wave tilts cancel each other and the transmitter envelope output is level as desired. The signal to the transmitter can then be increased, also increasing the average modulation level.



SQUARE WAVE INPUT

TRANSMITTER RESPONSE

RESULTING MODULATION ENVELOPE



PREDISTORTED INPUT WAVE

TRANSMITTER RESPONSE

RESULTING MODULATION ENVELOPE

It is important to note that the word "predistort" does not indicate the addition of audible distortion. The root word distort indicates that a signal or waveshape is in some way changed or altered from what it was. In most instances, these are non-linear changes which result in the generation of additional audio spectra at multiples of the fundamental frequency or at sum and difference frequencies of two fundamentals. These additional spectra constitute harmonic and intermodulation distortion respectively. Hence, the word distortion is normally associated with audible degradation. The process used here does not generate any additional audio spectra.

SETTING THE LF TILT CORRECT CONTROL

This control adjusts the characteristics of a compensating filter which follows the clipper diodes. As the filter characteristics are constant with amplitude, the following adjustments are made at reduced amplitude to minimize transient voltages within the transmitter during set-up.

Most newer AM modulation methods, such as pulse width modulation and solid-state Class D switching designs, have excellent square-wave response and will require little if any predistortion.

Connect an oscilliscope to an RF sample of the transmitter output. Adjust the scope for a "modulation envelope" display similar to the two on the far right on page 3. Trapezoid and "puckering circle" presentations are inappropriate for the following tests. The oscilliscope trace speed should be approximately 2 mS/division.

Some minor improvement in oscilliscope display stability may be had by connecting external sync to the scope; however, it is generally not necessary. The display will usually lock-up using normal sync as soon as modulation is applied to the transmitter.

Connect the input of the AUDIO PRISM to a sine wave oscillator. (DO NOT use square wave input for this procedure.) Set the oscillator frequency to 80 Hz. Frequency is critical. Use a frequency counter, if available, to insure the frequency is between 78 and 82 Hz. Adjust the oscillator output level while observing the amount of compression indicated by the LOW BAND M-101 card. Set the level so that the yellow LED under the letter "M" in compress is lit. Because the source is a single frequency, other bands will indicate considerably less compression. Depending on the setting of the gate control, the two higher bands may even enter the quiescent mode. This is normal.

For the purpose of setting the tilt corrector, advance the VCA DRIVE control so that the NORMAL LED on the EAGLE is lit all the time, but the HEAVY LED is not lit. This control will be set to its normal operating position described on later pages after adjustment of the tilt corrector is completed.

Observe the oscilliscope display. Advance the front panel OUTPUT GAIN until the modulation envelope indicates approximately 50% modulation. Verify this level with the modulation monitor.

*** CAUTION ***

> It is imperative that the modulation does not exceed 50% during the following steps. Higher modulation levels could cause permanent damage to the modulation transformer. The OUTPUT GAIN control will be adjusted in a later section for normal modulation levels.

Insert the eraser end of a pencil into the round hole above the LF TILT CORRECT control. This hole is labeled SETUP SWITCH on units after Serial Number 505. Use the soft eraser to depress the plastic switch directly below the hole.

This momentary contact switch removes the control voltage from the VCA, causing exaggerated clipping of the output. This makes the slope of the clipped wave much easier to read on the oscilliscope. Adjust the oscilliscope sweep speed for one or two complete cycles of the 80 Hz wave on the screen.

The LF TILT CORRECT control is very linear and predictable in its action. Rotate it through its complete range, while depressing the setup switch, to familiarize yourself with its action. Observe the transmitter's modulated RF envelope on the oscilliscope. Pay particular attention to the slope of the top of clipped waveforms.

Set the control to the position which produces a flat or level top on these waveforms. Turn the control in the direction of the lower side of the clipped waveform; if the wave is lower on the right, turn the control to the right, and vice versa.

After the proper adjustment is reached, remove the pencil from the setup switch.

SETTING THE VCA DRIVE CONTROL

The amount of VCA gain reduction is indicated by two LED's marked VCA ACTION. Illumination of the LED marked NORMAL indicates greater than 4dB (approximately) of gain reduction. Illumination of the LED marked HEAVY indicates greater than 10dB of gain reduction.

The VCA DRIVE control determines the trade-off between modulation density and signal quality. This control, more than any other, will determine the character of the on-air sound. Lower settings of this control will produce moderate modulation density with maximum quality. For the station where maximizing the service area is of great importance, higher settings of this control will produce greater signal penetration with some sacrifice of fidelity.

Occasional flickering of the HEAVY LED indicates light processing. Frequent flickering indicates a moderate amount of processing. Illumination of the HEAVY LED more than 20% of the time (20% on/80% off) qualifies as heavy processing.

The best setting of this control depends on a number of factors: the station's service area compared to that of its competitors, format, and amount of processing used by others in the market.

Advance the VCA DRIVE control, past the point where the NORMAL LED lights, to where the HEAVY LED flickers frequently. Observe the density and quality of the output. If this level of VCA drive is not optimum for your situation, adjust accordingly.

> Notice that the VCA DRIVE control adjusts the amount of gain reduction in the FET VCA. It does not control the degree of clipping. Clipping activity is a preset parameter of the FET Bias Module (FBM-1) and is not adjustable. For applications where this preset amount is not appropriate, replacement FBM-1 modules with greater or lesser clipper activities are available from the factory.

SETTING THE ASYMMETRY CONTROL

FCC regulations permit positive modulation of AM broadcast stations up to 125%. As negative modulation is limited to 100%, this is referred to as asymmetrical modulation. However, not every transmitter is capable of 125% positive modulation.* Some are not capable of positive peaks above 100%. Others are capable of some asymmetry, but less than the full 125%.

---

* For a full discussion of AM transmitter weaknesses, see Glen Clark, "AM Transmitter Techniques", Broadcast Engineering Magazine, December 1975, and C. B. Cox, "Enhancing AM Signal Coverage Through Improved Modulation", Procedings of the NAB Annual Broadcast Engineering Conference, 1974.

If you are not sure of the limitations of your transmitter, the following tests will determine what degree of asymmetry, if any, your transmitter will pass. Owners of newer transmitters employing pulse width modulation or solid-state Class-D switching schemes may be fairly certain that their transmitters will accept full 125% positive modulation and may skip the indented paragraphs which follow. Also, the engineer who does not desire to modulate asymmetrically may set the ASYMMETRY control fully counter-clockwise (symmetrical) at this time and proceed to the section SETTING THE AUDIO PRISM OUTPUT GAIN CONTROL.

If you are in doubt about the results you can achieve with your transmitter, consult the TEXAR Service Department. Our technicians have experience with most AM transmitters and can probably tell you what is normal for your make and model.

Determine if the phase polarity from the AUDIO PRISM to the transmitter is correct. Using music as a program source, drive the transmitter to 50% modulation with the ASYMMETRY control fully counter-clockwise. Verify the adjustment of your modulation monitor by switching the meter polarity switch from positive to negative. The meter should indicate similarly (within 5%) in both positions. (Some monitors have two meters which simultaneously read both polarities, and do not have such a switch). Now rotate the ASYMMETRY control fully clockwise. If the meter indicates higher when the meter polarity switch is set to positive than it does with it set to negative, the polarity is correct. If the meter indicates higher with the switch set to the negative position, reverse the polarity of the connections at the AUDIO PRISM output.

> Turn the ASYMMETRY control fully counter-clockwise. Connect an oscilliscope to a sample of the transmitter RF output. Still using music as a program source, adjust the scope for an envelope display. Increase the OUTPUT GAIN control of the AUDIO PRISM until 95% negative modulation is reached, indicated by the modulation monitor.
>
> Observe the location on the scope face of the positive peaks. Also notice the indication of the modulation monitor meter with the polarity switch in the positive position. Slowly advance the setting of the ASYMMETRY control. Observe whether the oscilliscope and modulation monitor indicate that the amplitude of the positive peaks are increasing as the control is turned. The action of this control, like the LF TILT CORRECTION control, is smooth and very linear. The increase in positive peaks indicated on the meter and scope display should be proportional to the position of the control.

Advance the asymmetry control until the point where further increase in its setting does not cause further increase in positive peaks. If this occurs at a positive modulation greater than 125%, you will be able to take full advantage of asymmetrical modulation allowed under FCC regulations. If this occurs at less than 125% positive modulation, this will be the maximum positive modulation which you may use.

Do not attempt to force positive modulation from a transmitter which does not naturally modulate asymmetrically. This practice will generate gross amounts of IMD in the transmitter, one of the most offensive types of signal degradation. Any small increase in coverage area is more than negated by listener irritation. Many transmitters, particularly those employing plate-modulation, simply are not capable of asymmetrical modulation.

Continuing the use of music as program source, observe the "+ CLIPPING" and "- CLIPPING" LED's on the EAGLE. Both should flicker about equally with the ASYMMETRY control fully counter-clockwise (symmetrical). As the control is advanced, the bias voltage on the positive clipper diode increases, letting more positive peaks pass. Because they pass rather than get clipped, the intensity of the "+ CLIPPING" LED will decrease as the ASYMMETRY control is advanced.

This incidentally serves as a good test of the EAGLE. If the LED's light equally at the symmetrical setting, and unequally when set for asymmetrical, it is a good indication that all parts of the EAGLE are working properly.

USE OF THE EAGLE WITH THE TEXAR PR-1 PHASE ROTATOR

It is recommended that the TEXAR PR-1 Phase Rotator be installed in all AUDIO PRISM's. The PR-1 implements an inaudible, linear process to remove asymmetry from the incoming program audio. As will be shown, this provides significant benefits, whether the station intends to transmit symmetrical or asymmetrical positive peak modulation.

It should be noted that the TEXAR PR-1 is acoustically transparent and does not detract from low frequency, transient response as do some phase rotators of other designs. The PR-1 employs a complex conjugate pair of poles near the frequency axis where some designs employ simple poles on the real axis. While these other designs produce anticipated flat response to steady-state, sinusoidal inputs, it can be demonstrated mathematically and acoustically verified that the bass transient response of the other designs is significantly compromised.

Where equal amplitude positive and negative modulation peaks are desired, the purpose of the PR-1 is obvious. However, even where asymetrical modulation is desired, the PR-1 is vital for optimum performance, although the logic is not as apparent.

There are two schools of thought regarding handling the polarity of asymmetrical program input, where an asymmetrical output is desired. The circuits associated with them are frequently called the "phase flipper" and the "phase rotator". The phase flipper senses the polarity of the input signal and, when necessary, switches an invertor into the circuit to maintain the desired polarity of the output signal. By contrast, the phase rotator removes any asymmetry from the input signal, producing a waveform with equal positive and negative peak amplitudes. If an asymmetrical output is desired, controlled asymmetry is reintroduced by the following clipper.

The phase flipper approach probably has more logical attraction on first consideration. Certainly it sounds simpler, which is something broadcast systems should be. Why remove asymmetry just to put it back in? In practice, the first method has two serious flaws, neither of which are obvious.

Inverting the program line polarity will also invert the phase of the announcer's headphones if an "off-air" monitor feed is used. As there is no such thing as absolute phase, only relative phase, music in the headphones will sound the same after the phase inversion as it did before the inversion. But that which the announcer hears while on live microphone, the headphone audio combining in his ear with the direct sound of his own voice, propagated through the tissues and bone of his head, will change. Here there are two sources to produce a relative phase. Electrically flipping the phase of one of the signals while the other remains unchanged will drastically change the character of the audio which the announcer percieves.

This effect is significant and disorienting. Most announcers find it very distracting while trying to work a live mike. One solution to this problem is to monitor console output instead of off-air, but because the processing equipment is then not included in the announcer's source, accurate voice overs and cross-fades are impossible to do. Newer versions of this approach use a sliding phase reversal circuit in place of the instant reversal. While this removes the "click" which the announcer previously heard at the moment of the phase reversal, it does not cure the rest of the problem of announcer disorientation.

The second non-obvious problem with the phase-flipper approach has to do with the degree of asymmetry of the input. U.S. FCC Rules limit positive peak modulation to 125%, or having a positive-to-negative ratio of 1.25 to 1. For a phase-flipper circuit to generate this ratio in the output without excessive clipping requires the input to already have the positive-to-negative ratio of exactly 1.25 to 1, no more, no less.

Consider the asymmetrical input wave which has 100% negative peaks and 200% positive peaks (not an uncommon occurance on male voice). The positive peaks will be reduced by a ratio of 0.625 to peak out at 125%; however, the negative peaks will also be scaled by 0.625 yielding 62.5% negative modulation. That leaves 37.5% of the legally permissible negative modulation unused. §100%-62.5% = 37.5%†

Even though the phase flipper has done its job, insured that the peaks were of the proper polarity, the modulation power delivered to the transmitter is 1.36 dB less than what it could be. §(125%+67.5%)/(125%+100%) = -1.36 dB†. This lost modulation can be recovered by not scaling the gain by 0.625 and instead simply clipping the positive peaks by 75% (200%-125%). However, this is a significant and audible degree of clipping. As will be shown below, this is significantly more clipping than would be required to convert a perfectly symmetrical input wave into one with 100% negative and 125% positive peaks.

Higher levels of input asymmetry underutilize the available modulation by even greater degrees or require even greater amounts of clipping. An input wave with 100% negative peaks and 300% positive peaks (still not an unlikely occurance) would produce audio power 2.6 dB less than permitted by law, or else require clipping 175% modulation of the positive-going wave to maintain maximum modulation §300%-125% = 175%†. §125/300 = 0.417; (125%+41.7%)/(125%+100%) = -2.6 dB†

The above analysis was based on U.S. standards. Foreign or international broadcasters may have different modulation limitations, and the equation numbers may change, but the point of the analysis remains valid.

While the phase-flipper approach may be appealing on first inspection, it has not shown itself to be a viable method of peak control. It has two inherent problems which cannot be overcome.

An asymmetrical input wave passed through a phase rotator has the phase relationship of critical odd-order harmonics modified in such a way that the output wave has equal positive and negative peak amplitudes. For FM applications, or AM applications where the transmitter is not capable of asymmetrical modulation, no further consideration of wave symmetry or polarity is necessary. Where the AM situation will permit asymmetrical modulation, the reference voltage to the positive clipper diode is simply raised so that the positive peaks are not clipped as much. Peaks that would otherwise have been truncated by the clipper in the normal course of modulation control are now permitted to pass to the transmitter.

This process takes place inside the control loop of the intelligent clipper. The result is that the total amount of clipping neither decreases or increases, satisfying the above claim that generation of full legal modulation by phase rotator method requires, on average, less clipping than does the phase flipper method.

The concept of phase rotator followed by intelligent clipper is deceptively simple, but powerful. When the rotator is properly designed, this combination requires less clipping and is the most acoustically correct method of producing full legal AM modulation.

SETTING THE AUDIO PRISM OUTPUT GAIN CONTROL

Good monitoring of the RF envelope is essential when adjusting the drive to the transmitter. Too little drive reduces modulation and coverage. Too much will cause distortion, interference and may evoke an FCC citation.

An oscilliscope RF envelope display is far superior to a modulation monitor. We have found few AM modulation monitors with peak flashers which accurately reflect the peak modulation. One peak flasher circuit relies on the gas ignition point of a neon bulb. Another drives the flasher circuit through a low-pass filter with very poor transient response. The filter "rings", which alters the peak amplitude of the signal to the flasher circuit. As a result, one can realistically tell very little about levels from the peak flasher of most modulation monitors. We have found only one brand of AM modulation monitor which accurately reflects the peak amplitude of an AM RF envelope.

Still using music as a program source, observe an RF envelope display on the oscilliscope. Increase the OUTPUT GAIN control to the point where the two closest parts of the envelope just begin to touch and form a faint line in the middle of the envelope. A bright line indicates too much modulation. A dimly lit area between the two closest points indicates insufficient modulation.

DISCLAIMER

Modern audio processing techniques produce very dense waveforms with peak-to-average ratios approaching one. Some plate-modulated transmitters manufactured before these modulation densities became common do not have sufficient safety margin for use with modern processors and may suffer modulation transformer or modulation reactor damage when driven heavily.

Frequently these models of transmitters have a reputation among engineers as having "light iron". This is not to say that all plate-modulated transmitters are weak in this area. Other models are well known for their near-indestructibility. Unfortunately, there is no simple way, short of driving it to the failure point, to tell if a particular model can sustain consistent heavy modulation. Members of both groups can be found equally among high power transmitters and low power transmitters, regardless of their age.

Asking the manufacturer is not a reliable way to determine which group a given transmitter falls into. If you are not sure of the limits of a particular unit, ask other engineers who have had experience with it. Such questions are usually welcomed and eagerly discussed at your local SBE Chapter Meeting.

The final judge of the suitability of the degree of processing shall be the user, who assumes all responsibility for its effects.

THEORY OF OPERATION

Operational amplifier U603B serves as a variable, active attenuator, transforming the high source impedance from the CX-1 board into zero source impedance at pin 7. JFET Q601 serves as a conventional shunt attenuator, with a range of approximately 15 dB. U603A restores the amplitude to a level usable by the FBM-1 module.

The output from the FBM-1 module splits to feed U603C and the Low Frequency Tilt Corrector (LFTC-1 Module). The amount of tilt correction in the module is fixed, and designed to be slightly more than that needed for the worst possible transmitter response. The outputs of U603C and the module are summed in R603, the LF TILT CORRECT control. U603D provides 6 dB of gain and a zero source impedance at the AMC-1 output, pin 4. This connects by the AMC-1's umbilical cable to the "hot" end of the CX-1 board OUTPUT LEVEL control (R104).

In the fully counter-clockwise position of R603, the unit's output is totally a function of the output of U603C. In the clockwise position, it is totally a function of the LFTC-1 output. Intermediate positions of the control provide a mix of the two and a lesser degree of correction. The gain of U603C is chosen to provide a constant average output level regardless of the position of R603. This minimizes the interaction between the adjustment of the output level and tilt correction controls, and greatly simplifies setup.

> Some transmitters, particularly those manufactured in the 1950's, require in excess of +12 dBm for 100% modulation. The gain of U603D is chosen to provide the maximum possible output level from the AUDIO PRISM consistent with the capabilities of the NE5532 output stage. Clockwise settings of both the LF TILT CORRECT and ASYMMETRY controls will increase the peak excursion of the voltage at the wiper of R603. In the event that heavy asymmetry is used with substantial amounts of tilt correction, the sum of these two may produce clipping in U603D.
>
> If your transmitter requires both, check the waveshape at the AMC-1 output. If the maximum excursion of positive peaks exhibits flattopping, remove R627. This reduces the gain of U603D from 6 dB to unity. Increase the AUDIO PRISM OUTPUT LEVEL control by a like amount.

The heart of the AMC-1 is the temperature-compensated LM336Z Precision Voltage Reference. All clipping and FET bias voltages are derived from it, insuring constant performance regardless of ambient conditions. The manufacturer rates the device's output as varying less than 0.8% over the range of from -55°C to 105°C.

U602D is configured as a fixed gain inverter and provides the negative clipper diode reference voltage at zero source impedance. The ASYMMETRY control (R602) varies the gain of U602D, the positive clipper diode reference voltage supply. Neglecting polarity, the range of R602 is such that the positive voltage can vary from 98% to 154% of the negative voltage.

R604 determines the quiescent FET bias voltage and is set to 2 dB past the transistor's knee. Normally there is no need to change this control from the factory setting unless Q601 is replaced.

U602B and U602C compare the audio signal at input to the FBM-1 module to the clipper reference supply voltages. If the signal voltage exceeds one or the other, the comparator output goes high, lighting the appropriate LED.

Similar comparators inside the FBM-1 module drive DS603 and DS604, which indicate the degree of FET gain reduction.

RETROFIT OR REMOVAL

The TEXAR EAGLE is normally supplied already installed at the factory. However, it is possible to field-install the EAGLE. For field installation, a new AUDIO PRISM lid, TEXAR Drawing #82004, Revision C, must also be ordered. The EAGLE cannot be mounted on the normal AUDIO PRISM lid.

All electrical connections from the EAGLE to the AUDIO PRISM are through a single five-conductor, ribbon cable, which connects via the CX-2 PC board (CX-1 on AUDIO PRISM's with serial number less than 505).

This installation should be done with AC power removed.

The following applies to later AUDIO PRISM's, equipped with the CX-2 board. For units equipped with the CX-1 board, consult the factory for field installation instructions. The CX-2 PC board is supplied with a short wire jumper in the position labeled Y102. This is along the very top edge of the CX-2. Being careful not to overheat the solder or lift a trace from the board, remove this jumper and clean the solder from the two holes. The EAGLE is supplied with a five-pin keyed socket (SAMTEC Part #SEP-10497-01) which mates to the keyed plug on the EAGLE connecting cable. This socket will have four female contacts and one male contact. Position the socket so that the male contact is on the right, farthest from the PINK NOISE toggle switch, and solder the socket in place.

Insert the mating plug into this socket. Place a sharp bend in the cable and fold it back over the top of the CX-2. Above the CX-2, place a diagonal fold in the cable so that the end connected to the EAGLE runs parallel to the front panel, over the high band M-101 card. Lay the remaining cable over the other M-101 cards. Being careful that the cable does not become pinched between the lid and the chassis or support posts, slowly set the lid in place and install the lid screws.

Connect power to the AC line cord. Immediately observe the two green, front panel, power supply LED's, "+15" and "-15". They should glow with equal intensity. If not, remove power, remove the lid and check for a pinched connecting cable.

When removing the lid on an EAGLE-equipped AUDIO PRISM, be careful to lift the lid gently and immediately disconnect the EAGLE cable from the CX-2. A sudden tug on the cable can snap the pins on the mating connectors.

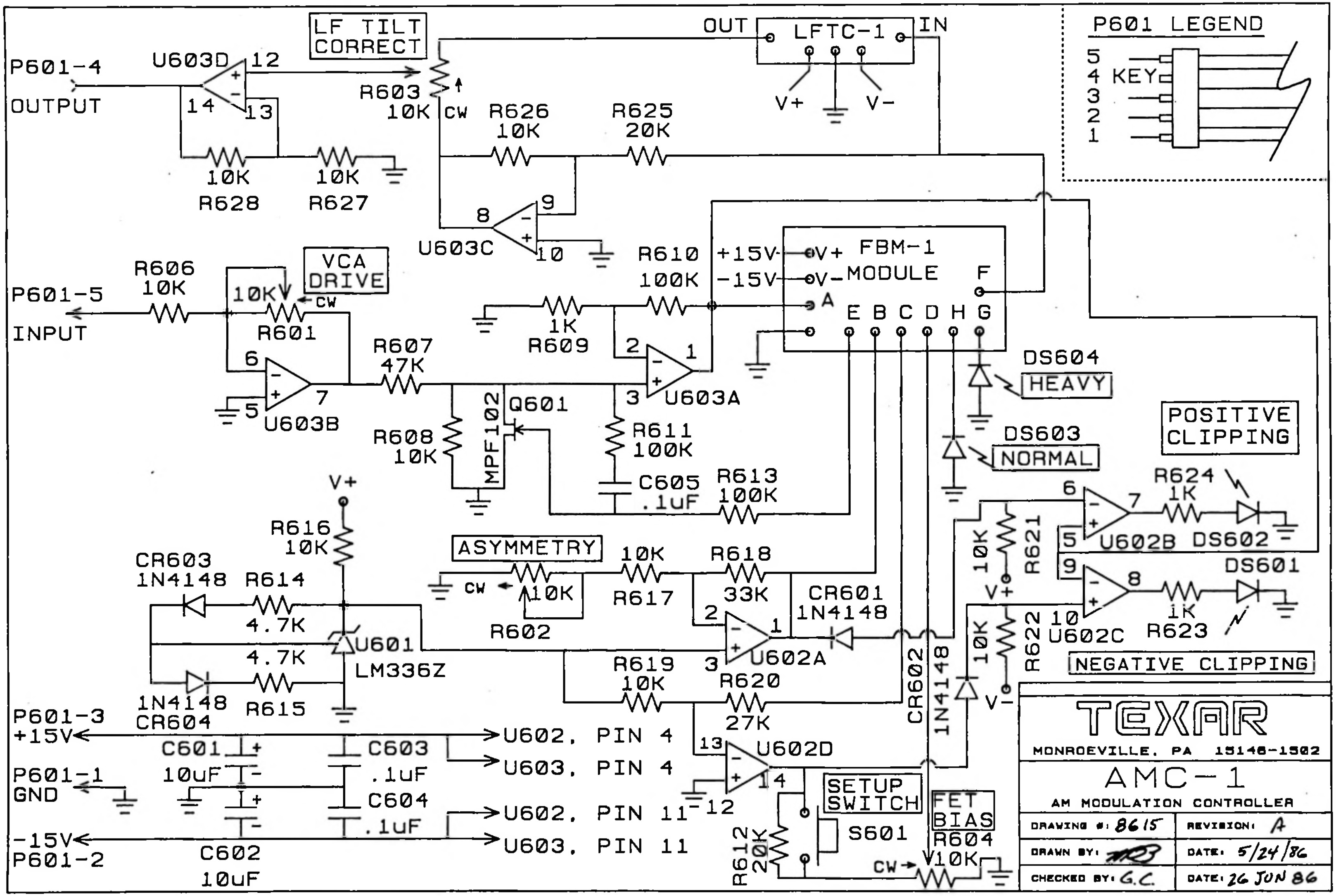
BYPASS MODE

The EAGLE is <u>not</u> in the program circuit when the front panel BYPASS MODE switch is in the ON position.

AMC-1 AM MODULATION CONTROLLER
REVISION A
PARTS LIST

PART NUMBER	DESCRIPTION	MANUFACTURER
C601-C602	10 uF @ 20 WVDC, TANTALUM	SPRG 196D106X9020JA1
C603-C605	0.1 uF @ 50 WVDC, CERAMIC, X7R	AVX MD015C104KAA
CR601-CR604	SILICON DIODE	1N914A
DS601-DS604	T-1 LED, RED	HLMP-1300
Q601	N-CHANNEL JFET	MOTOROLA MPF102
R601-R604	10 K LINEAR POTENTIOMETER	BOURNS 3386P
R605	NOT USED	
R606	10 K, 1/4 WATT	
R607	47 K, 1/4 WATT	
R608	10 K, 1/4 WATT	
R609	1 K, 1/4 WATT	
R610-R611	100 K, 1/4 WATT	
R612	20 K, 1/4 WATT	
R613	100 K, 1/4 WATT	
R614-R615	4.7 K, 1/4 WATT	
R616-R617	10 K, 1/4 WATT	
R618	33 K, 1/4 WATT	
R619	10 K, 1/4 WATT	
R620	27 K, 1/4 WATT	
R621-R622	10 K, 1/4 WATT	
R623-R624	1 K, 1/4 WATT	
R625	20 K, 1/4 WATT	
R626-R628	10 K, 1/4 WATT	
S601	SWITCH, SPST, N/O	ITT/SCHADOW 2810182
U601	2.5 VOLT PRECISION REFERENCE	LM336Z
U602-U603	QUAD OP-AMP	TL074CN
W601	AMC-1 P.C. BOARD, REV A	
X601	TILT CORRECTOR MODULE	TEXAR LFTC-1
X602	FET BIAS MODULE	TEXAR FBM-1

TEXAR, INC.
15 JULY 1986


LF TILT CORRECT
OUT
LFTC-1
IN
V+
V-
P601 LEGEND
5
4 KEY
3
2
1
P601-4
OUTPUT
U603D
12
14
13
R603
10K
CW
R626
10K
R625
20K
10K
R628
10K
R627
8
9
10
U603C
R610
100K
+15V
-15V
V+
V-
FBM-1
MODULE
F
A
E B C D H G
R606
10K
P601-5
INPUT
VCA DRIVE
10K
CW
R601
6
5
7
U603B
R607
47K
1K
R609
2
1
3
U603A
DS604
HEAVY
R608
10K
MPF102
Q601
R611
100K
C605
.1uF
R613
100K
DS603
NORMAL
POSITIVE CLIPPING
R624
1K
6
7
5
U602B
DS602
10K
R621
V+
9
8
10
U602C
1K
R623
DS601
10K
R622
V-
NEGATIVE CLIPPING
V+
R616
10K
CR603
1N4148
R614
4.7K
U601
LM336Z
4.7K
1N4148
CR604
R615
ASYMMETRY
CW
10K
R602
10K
R617
R618
33K
CR601
1N4148
2
1
3
U602A
CR602
1N4148
R619
10K
R620
27K
13
U602D
14
12
SETUP SWITCH
S601
FET BIAS
R604
10K
CW
R612
20K
P601-3
+15V
C601
10uF
C603
.1uF
U602, PIN 4
U603, PIN 4
P601-1
GND
C604
.1uF
U602, PIN 11
U603, PIN 11
-15V
P601-2
C602
10uF
TEXAR
MONROEVILLE, PA 15146-1502
AMC-1
AM MODULATION CONTROLLER
DRAWING #: 8615
REVISION: A
DRAWN BY:
DATE: 5/24/86
CHECKED BY: G.C.
DATE: 26 JUN 86

TEXAR INCORPORATED
616 Beatty Road
Monroeville, PA 15146-1502
(412) 856-4276
(412) 85-MICRO

March 23, 1987

Dear Texar RCF-1 Customer:

Enclosed please find the promised documentation and access panel for the RCF-1 which our records indicate was recently purchased by you. A significant amount of feedback from our customers like yourselves has allowed us to put much additional information in this manual. It has been three hole punched, so that it can be placed in your Audio Prism™ Users' Manual. If you have a recent manual, you should find that there is a red plastic divider at the rear of your manual for precisely this purpose.

Also enclosed are three capacitors and a technical update sheet which describes how to install them on your RCF-1. These capacitors alleviate a condition which was present in only a few RCF-1's; however, we recommend this update for ALL RCF-1's as a precaution.

Thank you for your interest in Texar products and your patience in getting the enclosed material to you.

Barry Honel

Barry Honel
Operations and Sales Manager

6. TEXAR RCF-1

TEXAR INCORPORATED

616 Beatty Road

Monroeville, PA 15146-1502

(412) 896-4276

MARCH 1987

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6. TEXAR REPLACEMENT CARD FIVE

6.1. INTRODUCTION

The TEXAR Replacement Card Five (RCF-1) is a performance-enhancement kit for the Orban Optimod 8100(A)*. It consists of three items: the RCF-1 circuit board, a removable, metal, access panel, and this User's Manual. Benefits of the RCF-1 are greater modulation density and increased bass response, or a much less processed sound while still maintaining the same loudness derived from the original Orban Card #5.

The RCF-1 was designed specifically for use in Optimod 8100's used in conjunction with two **TEXAR AUDIO PRISM's™**. Increased performance is accomplished by more closely matching the operational characteristics of the Optimod with those of the AUDIO PRISM's. Use of the RCF-1 in barefoot Optimod's (those without multi-band pre-processing) is strongly discouraged as the unit may not react favorably to all types of program conditions. Use of the RCF-1 in Optimod's preceeded by pre-processors other than the AUDIO PRISM have not been taken into account in the design, and will produce unknown results.

The RCF-1 will not work with the Orban Optimod 8000*.

6.2. INSTALLATION

Installation of the RCF-1 is quite simple and requires less than five minutes. Although adjustment of the RCF-1 and minor readjustment of the AUDIO PRISM's is likely to be neccessary, the air-sound is likely to be sufficiently close to the final product that the Optimod can be returned to the air as soon as the card installation is complete.

6.2.1 Recommended Equipment

Best operation of the RCF-1 is obtained with the following equipment:

1) An otherwise unmodified Orban Optimod 8100 or 8100/A configured for SINGLE-SITE operation. Some customers with user-supplied modifications to the Optimod Card #6 experienced unusual performance when the RCF-1 was installed. An exception to the prohibition of modifications is the removal of the Optimod phase rotators described on page 31 of the TEXAR AUDIO PRISM Users Manual (Editions January 1987 and later). Removal of the Op-

* Optimod 8100 and Optimod 8000 are registered trademarks of Orban Associates, Inc., San Francisco, CA.

timod phase rotators will not negatively affect the performance of the RCF-1, so long as the TEXAR PR-1 Phase Rotators are installed in the AUDIO PRISM's. The blue plastic jumpers on the Orban cards #6, #8 and #9 should be set for normal (non-XT mode) operation. See Appendix M of the Orban Optimod 8100 Operating Manual. Orban Broadcast offers a final output filter/clipper option card for the Optimod 8100(A), known as the "zero card". Use of this card in conjunction with the TEXAR RCF-1 has proven to meet with positive results from broadcasters who transmit one or more SCA channels.

The Optimod should NOT be in a split-site configuration. In the split-site configuration, the H-F Limiter Card (Card #6) will be located at the transmitter and the AGC Card (Card #5) will be located at the studio. Even minor phase and amplitude anomalies in the telephone lines or dual channel STL's can upset the critical peak level fed to the H-F Limiter Card by the AGC cards. This does not present a problem, as no reason exists to run split-site configuration with the **TEXAR AUDIO PRISM** / Optimod 8100(A) combination.

The purpose for split-site operation is to protect the telephone lines or STL's from being overdriven by program peaks. Placing the Orban accessory chassis at the studio provided level protection for these program circuits. However, as the RCF- 1 is intended to be used only in conjunction with two, studio-located, TEXAR AUDIO PRISM's, which will provide the program circuit protection, split-site operation of the Optimod is unneccessary.

Accordingly, the RCF-1 will not mount in the Orban split-site chassis. Please refer to the Orban Accessory Chassis Manual Supplement for instructions on returning the split-chassis Optimod to an original single-site unit.

2) A modern (two-speed AFC) FM exciter. While all modern FM exciters employ a two-speed AFC loop, there are still some exciters in use which have only a single-speed loop. When used with heavily processed audio, these exciters are sometimes unable to distinguish between a low bass note and a low-frequency AFC correction voltage.

A sudden bass note may unsettle the AFC loop causing a tearing sound on the air, identical to that produced by multipath reception. In the case of multipath reception, the phase-locked-loop (PLL) in the receiver's detector is on the verge of unlocking. In this instance, it is the PLL in the exciter AFC which is on the verge of unlocking. The only remedy short of changing the AFC circuit is to reduce the degree of processing.

The following is a list of exciters with which TEXAR engineers have had experience sufficient to form an opinion:

Acceptable for use with the RCF-1:

- Continental 802A
- Broadcast Electronics FX-30
- Harris MX-15

Unacceptable for use with the RCF-1:

- CCA FM-10E
- CCA FM-40E
- Harris/Gates TE-1
- Harris/Gates TE-3
- Harris MS-15*
- RCA BTE-15
- RCA BTE-115

6.2.2 Installation Procedure

The procedure for installing the RCF-1 is quite simple:

1) Remove the three stainless-steel hex-screws across the top edge of the Optimod 8100. Either a 1/16" or a 2mm metric hex driver will fit these screws correctly.

2) Tilt down the entire beige front panel (the cards cannot be removed through the smaller opening of the access door).

3) Turn off the AC power. DO NOT remove cards from the Optimod with power applied. The power switch is a white, plastic, toggle located on the left. (As transients produced during power-down at the Optimod's BNC, output connector have infrequently unlocked the AFC of the STL transmitter or the FM exciter which follow the Optimod, some engineers remove the output cable prior to power- down. Remember to reconnect it later.)

4) Remove the chocolate-brown or beige colored access panel by unlocking the Dzus fastners on the four corners. (Color of panel depends upon serial number of the Optimod 8100.) These are quarter-turn fastners. Tilt the top of the panel toward you, making sure it does not become caught on the upper lip. This can frequently be accomplished by placing a finger through one of the open holes in the panel and pushing downward. This should compress the two ribbon cables beneath the panel sufficiently for the panel to be removed.

5) Remove the original Orban Card #5 by pulling on the nylon card ejector on the top of the card. Remove it the rest of the way from the card cage and put it in a safe place. It will be necessary to reinstall the card, should it become necessary to return the Optimod for factory service at a later date. Orban will (understandably) not perform service on equipment which is not completely of their manufacture.

* The HARRIS MS-15 is one exciter with a single speed AFC. Its look-alike successor, the MX-15, has a two-speed AFC. The Harris MS-15 can be returned to the factory to be upgraded to a model MX-15 for a charge. Contact Harris for details. The improvement should be implemented on any MS-15 which will be used in conjunction with the **AUDIO PRISM** and the RCF-1.

6) Slide the TEXAR RCF-1 into the newly-vacant slot. Push gently but firmly to insure that it seats in the edge connector at the rear. **DO NOT FORCE THE CARD.** This could result in damage to the card or the connector, or result in a fracture of the mother board. If the card will not seat easily, remove it and investigate the source of the obstruction. There is some vertical variation among Optimods in the relative positioning of the edge connector and the card guide rails. If a verical alignment problems is present, clipping a corner of the RCF-1 edge connector with a pair of diagonal cutters may correct this problem. The cut should be about 1/8" and at a 45 degree angle, but should not include any of the gold fingers.

7) Without installing the new chocolate-brown-colored access panel, turn the power switch back on and temporarily tilt the beige front panel. Verify that the multimeter reads near 100% in the +15V and -15V positions of the rotary, meter-selector switch. Also verify that both LED's on the RCF-1 are illuminated and the gain reduction meters on the Optimod have returned to zero. If no program material is fed to the Optimod, these LED's should illuminate steady green. If program material continues to be fed to it, both may flicker occasionally to red and then return to their normal green. If both the meter and LED conditions are met, proceed. If not, investigate the offending condition.

8) Tilt the beige front panel back down and install the brown TEXAR access panel that was provided with the RCF-1. It should be tilted for insertion in the reverse manner of how the original was removed. Be certain, as you position the panel for final alignment, that DS1 and DS2, the two LED's on the RCF-1, align with the two holes labeled "MAIN FOLLOW BASS" and "BASS FOLLOW MAIN". Anchor the panel by clockwise rotating the four Dzus fasteners. These are identical to the fastners on the Orban panel and are also quarter-turn fasteners. When properly seated, the screwdriver slots should all be similarly aligned, parallel to the ground.

9) VERIFY THAT THE TWO RIGHT-HAND SWITCHES LOCATED ON CARDS #6 & #7 ARE IN THE UP ("OPERATE") POSITION AND THAT THE RCF-1 SWITCH IS TO THE LEFT (also "OPERATE"). Removal and replacement of the brown panels may have changed the switch settings. If this condition is not corrected at this time, it may result in improper operation or a needless trip back to the transmitter site.

10) Close the beige front panel and reinstall the three stainless-steel hex screws.

11) RECONNECT THE OUTPUT CABLE IF IT WAS REMOVED IN STEP 3 ABOVE.

6.3. ADJUSTMENT

6.3.1 The RCF-1 Controls

There are three variable controls, one switch, and two indicators on the RCF-1. The recommended initial settings of the variable controls are indicated on the access panel by a circle around the appropriate number. The controls function as follows:

CLIPPING - Regulates the level of the audio supplied to the clipper diodes. Clockwise rotation of this control increases the level fed to the clipper and the amount of clipping performed. Increased apparent loudness will result. Counter-clockwise rotation of this control will produce a less processed sound. The recommended inital setting is 7. While the CLIPPING control is properly labeled on all of the TEXAR-supplied access panels, on Revision 0 of the RCF-1, this control was silkscreen labeled as DENSITY rather than CLIPPING.

BASS BOOST - Controls the proportion of low-frequency program material to mid- and high-frequency material. Clockwise rotation produces more bass response. The recommended initial setting is 8.

INTERBAND COUPLING - Regulates the degree of corrective action taken by the INTERBAND COUPLING MODULE (IBC-1). Counterclockwise rotation causes the maximum amount of interaction between the MASTER and BASS control voltages. Experience has shown that a setting of 0 (maximum coupling) is normally optimal for all formats.

PROOF/OPERATE SWITCH - When in proof mode, inhibits action of the VCA's so that steady-state (tone) measurements may be performed.

COUPLING ACTION LED's - Indicate the corrective action taken by the IBC-1. These LED's will glow a steady green with no audio applied. A corrective action taken on the MASTER control voltage by the IBC-1 will cause the MASTER FOLLOW BASS LED to flash red momentarily. A red flash of the BASS FOLLOW MASTER LED indicates a corrective action taken by the IBC-1 on the BASS control voltage. Both LED's will become less active with more clockwise settings of the INTERBAND COUPLING control. The COUPLING ACTION LED's do not serve a significant purpose during set-up and adjustment. Their primary purpose is as a diagnostic tool. See the TROUBLESHOOTING section of this manual.

There is no GATE control on the RCF-1. There is need for only one noise gate in an audio system and multiple gates will "fight" with each other. This function is already provided in the AUDIO PRISM's.

The CLIPPING and BASS BOOST controls do not act instantaneously. They do not directly regulate the VCA control voltages. Rather, they control the threshold voltages fed to the MASTER and BASS PARC modules. As a result, it will take several seconds for changes in these controls to be reflected in the operation of the unit. The proper procedure is to make an exploratory change in one of these controls, and then to wait several seconds for the VCA gains to stabilize before evaluating the change.

Few parallels should be drawn between the operation of the RCF-1 controls and those of the original card #5, other than that clockwise rotation of both CLIPPING controls will result in a more processed sound. While there are obvious outward similarities, the internal operation of the two cards is very different. One should also not attempt to draw correlations regarding relative knob position between between the RCF-1 and the original card #5. A 12 o'clock setting of the RCF-1 CLIPPING control does not necessarily produce the same degree of clipping as a 12 o'clock setting of the original CLIPPING control. Further, while the INTERBAND COUPLING and the BASS COUPLING controls have similar-sounding labels, they are not equivalent. The BASS COUPLING control served as a de facto BASS response control. The INTERBAND COUPLING control does not affect spectral balance.

The movement of the left three edgewise front-pane meter will, with the RCF-1 installed, differ significantly from the meter action experienced with the original card #5 installed. In particular, the LIMITING meter will now be much less active, seldom moving past 2 dB of gain reduction. This is perfectly normal. Additionally, if significant amounts of BASS BOOST are used, the BASS G/R meter will deflect less than before.

6.3.2 Adjusting The System

Set the INTERBAND COUPLING control to 0, the CLIPPING control to 7, the BASS BOOST control to 8, and the H-F LIMITING control to 5. Adjust the INPUT ATTENUATOR's for between 6 and 13 dB of gain reduction.

Fine tuning the RCF-1 to fit your format is very easy. The following are some suggestions which may aid you in reaching the sound you desire.

The INTERBAND COUPLING control produces little aural change. As stated earlier, this control DOES NOT provide for bass enhancement as the BASS COUPLING control did on the origional Orban card #5. While the recommended setting on the IBC control is zero, there may be certain situations that sound slightly more dynamic if this control is rotated from the zero setting. Classical formats or very un-processed stations may find a setting of one to five to be slightly less restrictive. Any setting above five on the IBC control will, for the most part, remove the IBC function from the processing algorithm.

Most stations are looking for a more open, less processed sound. This is much more true today than just a few years ago, mainly due to the advent of digital Compact Discs (CD's) and renewed interest in listener fatigue and quarter-hour-maintanance. The CLIPPING control on the RCF-1 can provide high loudness with either a dense tight 'electric' sound or an open dynamic sound. Note however, that undesirable distortion may result if large amounts of clipping are simultainiously used with high levels of gain reduction. Turning the CLIPPING control more toward zero will increase gain reduction and will decrease clipping. A CLIPPING control setting between 4 - 7 will deliver a sound that 'feels' controlled and consistant.

Used judiciously, this control is very powerful and can maximize loudness, but some cautions are in order:

It is recommended that the following CLIPPING positions are not exceeded, based on the amount of indicated TOTAL gain reduction:

	Gain Reduction	CLIPPING
•	UP to 6db	10
•	6db - 8db	8
•	8db - 10db	7
•	10db - 13db	6

TEXAR recommends that the Optimod input attenuators be adjusted for a minimum of 6dB - 8dB and a maximum of 10dB - 13dB peak gain reduction, read on the TOTAL meter. The RCF-1 was specifically designed to operate in this range, unlike the Optimod without the RCF-1 installed.

A good, solid bass is even more important today than before. With the cleaner high-end from better recordings, the bass can get lost in the sizzle. The RCF-1 BASS BOOST control has the power to overcome this problem and is very effective.

A setting of 4 on the BASS BOOST control will effectivly remove any dynamic bass enhancement and will sound closest to the origional program material. A setting greater than 5 will increase the dynamic enhancement of the bass.

The BASS BOOST control, like the CLIPPING control, can be abused. Changes in BASS BOOST should be made very slowly and a long listening session should be made prior to furthur changes. We recommend you listen for a few days on several different types of radios after you make a change to this control. You should be able to get a very solid, punchy bass sound for your format by careful adjustment of this control.

The only limitation we have found on the use of the BASS BOOST control becomes apparent when using nearly the maximum recommended gain reduction and the CLIPPING control setting is higher than that recommended. If your format and market demand a situation as described, you should not exceed a BASS BOOST setting of 7. A setting higher than that will cause low frequency IM distortions to become apparent.

The RCF-1 has no control that directly adjusts the high frequency response. The H-F LIMITING control on card 6 can be used to produce some mild high end differences. With settings low in number (eg. 1 - 3), distorion products will be minimized, but a slightly restricted hi-end may occur. Cymbals may sound less exciting and limited in nature, but sibilance will be greatly reduced. If your source material is marginal, you may have to use lower settings of the H-F LIMITING control.

For stations looking for a very unprocessed sound, and who have exceptional source material (eg. CD's) and a very clean audio chain, a higher setting of H-F LIMITING may be used. This will open up the highs and create a less restricted feel. Adjustment of the H-F LIMITING control is very subjective and only close listening will indicate what setting should be used.

Minor readjustment of the Audio Prism mix levels will likely be necessary for optimum performance with the addition of the RCF-1. Some early TEXAR advertisments for the RCF-1 indicated that no re-adjustments of the mix levels was required; however, experience has shown this not to be the case. In particular, greater levels of PRESENCE and HIGH band injection may be required to properly offset the greater bass of the RCF-1. Consult Appendix A, on page 50, of the January 1987 AUDIO PRISM Users' Manual. Spectral mix is market dependant. What works in New York may not be appropriate for Athens, Ga. Long listening and minor control changes are your best bet for precisly adjusting the audio processing chain. TEXAR Factory assistance is available to TEXAR product owners who have question on the setup of the RCF-1.

6.4. THEORY OF OPERATION

The TEXAR RCF-1 is a direct, functional replacement for the original Orban Card #5, but with modified operating algorithms. It draws its operating power from the Optimod, and has the same pin-out and protocol as the original Card #5 (See the upper left corner of the RCF-1 schematic, TEXAR Drawing #87001, for call-out of the edge-connector pin-out).

During its normal lifetime, the RCF-1 will require no recalibration or adjustment of any but the front-panel controls.

Refer to TEXAR Drawing #87002. Card 5's purpose is to generate AGC control voltages to regulate the MASTER and BASS band VCA's, located on Orban Cards #3 and #4.

The RCF-1 samples the output of the left and right channel MASTER band VCA's and processes them in the MASTER PERCEPTUAL ATTACK AND RECOVERY COMPUTER ("MASTER PARC module"). The MASTER PARC module, in turn, produces a control voltage. The left and right MASTER band samples enter the RCF-1 on edge connectors pins "M" and "N", respectively, and enter the MASTER PARC module on pins 4 and 6. The uncorrected MASTER control voltage leaves the MASTER PARC MODULE on pin 7. The MASTER PARC MODULE is the upper of three black modules mounted on the RCF-1.

The left and right channel BASS band samples enter the RCF-1 on edge connector pins "R" and "S", and enter the BASS PARC MODULE on pins 4 and 6. The uncorrected BASS control voltage leaves the BASS PARC MODULE on pin 7. The BASS PARC MODULE is the lower of the three black modules.

The uncorrected MASTER and BASS control voltages are then fed to the INTERBAND COUPLING module (IBC-1), the center one of the three black modules. Under most conditions, the IBC-1 will simply pass the control voltages through to its output with no change. However, under transient program conditions, such as a bass drum beat or cymbal crash, where unduly harsh program clipping could occur, the IBC-1 will modify the control voltages to produce a more desirable audio product. The IBC-1's operation differs significantly from that of the original Orban BASS COUPLING circuit in a number of ways. One is that the IBC-1 can modify both MASTER and BASS control voltages, where the BASS COUPLING circuit can modify only the bass control voltage.

The "corrected" control voltages are then converted to control currents to drive the VCA's on cards #3 and #4. The corrected MASTER control voltage is fed from pin 4 of the IBC-1 module to voltage follower U4A. A sample of the MASTER control voltage feeds edge connector pin "X", which drives the edgewise front-panel, gain-reduction, meter labeled COMPRESSION. Linear-voltage-to-log-current conversion is accomplished in differential pair U7B. The MASTER control current exits the RCF-1 on edge connector pin "F".

A sample-and-hold circuit consisting of U5B drives voltage follower U5C, which provides voltage to the front panel TOTAL gain-reduction meter, via pin "Y" on the edge connector. (There is no lead to drive the front panel LIMITING meter, as it is multiplexed from the two existing lines by the Optimod metering PC board.)

The corrected BASS control voltage is fed from pin 4 of the IBC-1 to voltage follower U4B, which drives the front panel TOTAL BASS gain reduction meter via edge connector pin "W". The output of voltage follower U4B is also linear-voltage-to-log-current converted in differential pair U7A.

U6B and U6A maintain the desired static currents through the MASTER and BASS differential pairs, respectively. R26 and R32 are factory adjustments which set the static currents to produce the desired, quiescent gain in the VCA's when no AGC voltage is present.

U1 is a thermally-compensated, precision, voltage reference which supplies a precise 2.50 volt for reference use on the RCF- 1. Its output is voltage followed by U2B. This voltage drives U5A and U5D which provide -1.200 and +1.200 volts to the clipper diodes located on Orban Cards #3 and #4. The 2.50 volt source also feeds variable-gain amplifier U2D and subsequently U3A, which provide the variable CLIPPING reference voltage to the MASTER PARC module. The 2.50 volt reference also drives variable-gain amplifier U2A, which supplies the INTERBAND COUPLING reference voltage to the IBC-1 module. Variable-gain stage U2C samples the CLIPPING reference voltage and provides, through voltage-follower U3B, a BASS BOOST reference to the BASS PARC module. Notice that the voltage at the output of U3B is dependant not only on the position of R15, the BASS BOOST control, but also on the position of R10, the CLIPPING control. The physical significance of this is that, as the CLIPPING control is varied, the clipping depths of the MASTER and BASS bands will track up and down together, in proportion to each other. The BASS BOOST control will affect only the ratio of BASS to MASTER band program material.

To place the RCF-1 in PROOF mode, switch S1 places +15 volts on the inputs to voltage-followers U3A and U3B, forcing the CLIPPING and BASS BOOST reference voltages to high levels. This places the threshold for VCA voltage generation in the PARC modules at an unreachable level, insuring that no AGC action will take place. The PROOF mode switch also places +15 volts on the input to voltage follower U5D, pushing the clipper diode bias voltages on edge connector pins "H" and "J" from their normal levels, to +15 volts and -15 volts, respectively. This also is an unreachable level, insuring that no clipping will occur.

6.5. TROUBLESHOOTING

If you should experience problems with the RCF-1/Optimod 8100 combination, the first step is to determine if the problem is contained in the RCF-1 or the Optimod itself. This can easily be accomplished by temporarily replacing the RCF-1 with the original Orban Card #5. If the condition persists, the problem lies in the Optimod itself. If the condition is corrected, the problem is indeed in the RCF-1.

***** IMPORTANT *****
The TEXAR RCF-1 is not an Orban Broadcast product. DO NOT call Orban Broadcast for assistance with TEXAR's RCF-1 Replacement Card Five. For factory assistance, contact TEXAR's Technical Support Department at (412) 856-4276.

Should your Optimod need to be returned to the Orban factory for repairs or re-alignment, do not ship the Optimod with the RCF- 1 installed, as it will not conform to the protocol of the Orban alignment and test fixtures. The original Orban Card #5 should be installed for such a return.

A simple test for the likely proper functioning of the RCF-1 is to observe DS-1 and DS-2, the LED's labeled MAIN FOLLOW BASS ("MFB") and BASS FOLLOW MAIN ("BFM"). With the INTERBAND COUPLING control fully counterclockwise (setting "0") and active program material applied to produce nominal compression(about 8dB), these LED's should occassionally flicker from their normal green color to a momentary red color, and back. While not a complete and thorough test, the above condition is a statistically good indication that the RCF-1 is operating properly. The presence of this condition; however, does not neccessarily indicate that either the RCF-1, or the rest of the audio system, is properly adjusted. Note also that the BFM indicator will be the more active of the two, changing color much more frequently then the MFB indicator. This is a normal condition.

If a problem is verified to exist and it has been isolated to the TEXAR RCF-1, the card extender will be invaluable in locating the condition. The card extender is normally stored in an empty slot in the Optimod, located to the left of Card #3 and against the power supply partition. Remove the brown access panel as described earlier in the INSTALLATION section of this manual. Being certain to turn the power switch off, remove the RCF-1. Remove the card extender from the storage slot and place it in the slot vacated by the RCF-1. Insert the RCF-1 in the card extender with the components facing to the left (the number "5" on the card ejector should be upright). Turn the power back on. Consult the attached TEXAR Drawing #87001, schematic for the RCF-1.

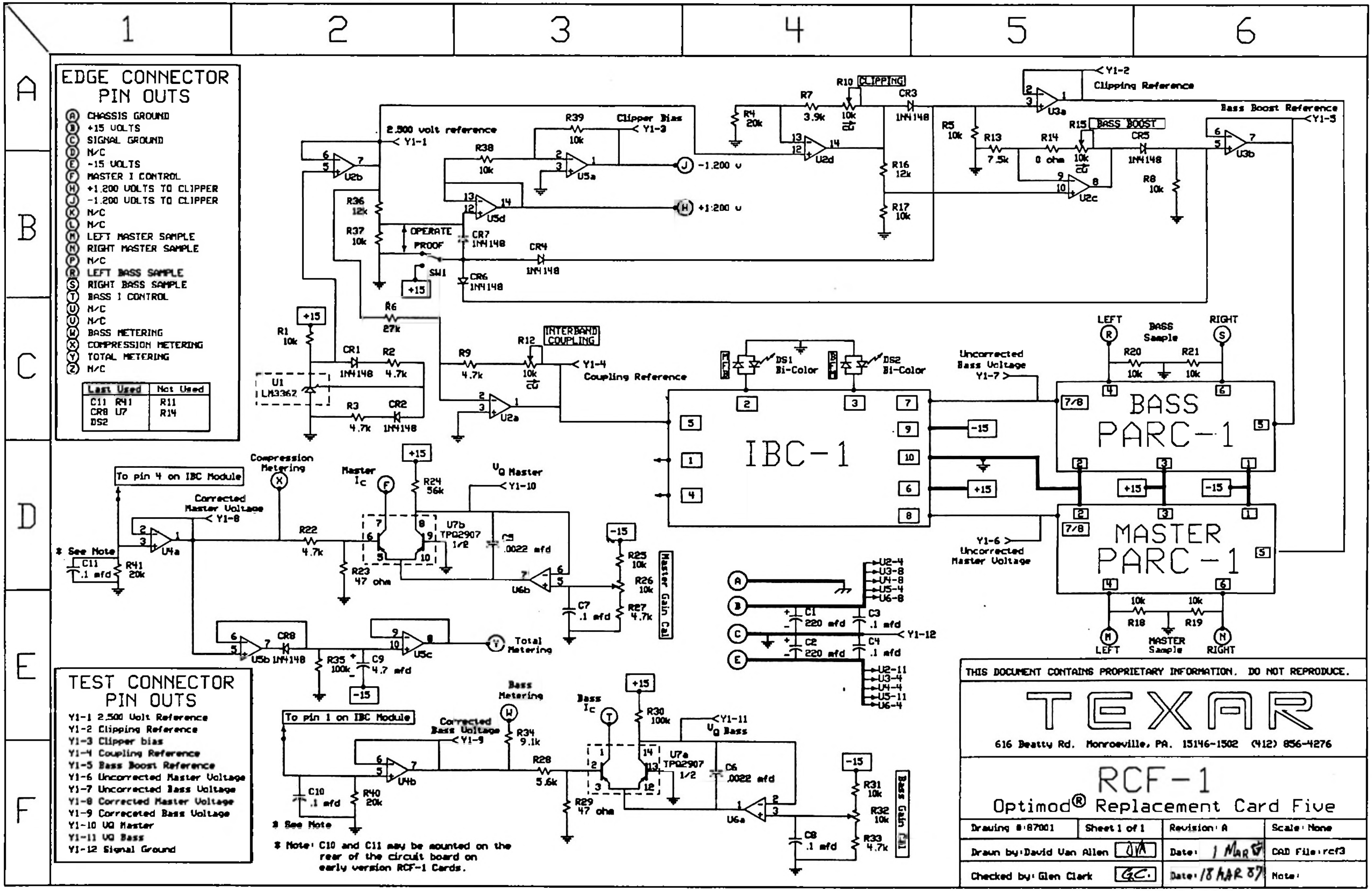
To allow convenient troubleshooting, eleven operational voltages of the RCF-1 plus ground are brought out to test and alignment connector Y1. Pin-out of the connector is described in the lower left corner of the above-referenced schematic. Pin 1 of the connector is closest to the Optimod. Pin 12 is closest to the user. While scope "hook probes" conveniently grasp the L-shaped pins for troubleshooting, frequently the ground clip of the scope probe is too large to connect without shorting pin 12 to pin 11. A convenient place to connect this lead to ground is at the top lead to C2, one of the electrolytic capacitors at the top rear of the RCF-1, and the one closest to the user.

Make sure the PROOF/OPERATE switch is in the OPERATE (left) position. Proceed to check the following voltages. If any are not within the recommended ranges, consult the schematic for the logical location of the defective component. Verify that there is +2.50 volts present at Pin 1 of Y1. Connect the meter or oscilliscope to Y1-2. This voltage should vary from +2.45 volts to +3.70 volts as you rotate the CLIPPING CONTROL from the left stop on the potentiometer to the right stop. (The CLIPPING control, R10, was silkscreened on early RCF-1 boards as the DENSITY control.) Verify that -1.20 volts is present at Y1-3. The voltage on Y1-4 should vary from -0.4 volts to -1.25 volts as the INTERBAND COUPLING CONTROL is rotated from full left rotation to full right. Place the CLIPPING control in its full counterclockwise position. The voltage on Y1-5 should vary from +0.85 volts to +2.60 volts when the BASS BOOST control is rotated from full left to full right. Tolerence for these measurements is plus or minus 10%.

You should observe nearly ground potential at Y1-6 and Y1-7 when no program material is applied. With program material applied, you should notice time-varying, negative-going voltages on both of these. The exact voltage will vary depending on the amount of compression, but it will be on the order of several volts. These are the "uncorrected" control voltages. The BASS control voltage, found on Y1-7, should move somewhat more slowly that the MASTER control voltage. The waveform on Y1-8 should mimic the voltage on Y1-6 nearly identically. As described earlier under THEORY OF OPERATION, these voltages are unequal only during the momentary actions of the Interband Coupling Module (IBC-1). The voltage on Y1-8 will be unequal to the voltage on Y1-6 only when the BASS FOLLOW MAIN (BFM) LED changes color from green to red. Similarly, the voltage on Y1-9 should be nearly identical to the voltage on Y1-7. They will be unequal only when the MAIN FOLLOW BASS (MFB) LED changes from green to red.

Y1-10 and Y1-11 are provided for factory calibration of R26 and R32. They provide little troubleshooting information and may be ignored.

Further diagnosis can be accomplished by a qualified technician and consulting the THEORY OF OPERATION section, earlier in this manual. Consult TEXAR's factory technical support department for further assistance.



EDGE CONNECTOR PIN OUTS

- A CHASSIS GROUND
- B +15 VOLTS
- C SIGNAL GROUND
- D N/C
- E -15 VOLTS
- F MASTER I CONTROL
- H +1.200 VOLTS TO CLIPPER
- J -1.200 VOLTS TO CLIPPER
- K N/C
- L N/C
- M LEFT MASTER SAMPLE
- N RIGHT MASTER SAMPLE
- P N/C
- R LEFT BASS SAMPLE
- S RIGHT BASS SAMPLE
- T BASS I CONTROL
- U N/C
- V N/C
- W BASS METERING
- X COMPRESSION METERING
- Y TOTAL METERING
- Z N/C

Last Used	Not Used
C11 R41 CR8 U7 DS2	R11 R14

TEST CONNECTOR PIN OUTS

- Y1-1 2.500 Volt Reference
- Y1-2 Clipping Reference
- Y1-3 Clipper bias
- Y1-4 Coupling Reference
- Y1-5 Bass Boost Reference
- Y1-6 Uncorrected Master Voltage
- Y1-7 Uncorrected Bass Voltage
- Y1-8 Corrected Master Voltage
- Y1-9 Corrected Bass Voltage
- Y1-10 VQ Master
- Y1-11 VQ Bass
- Y1-12 Signal Ground

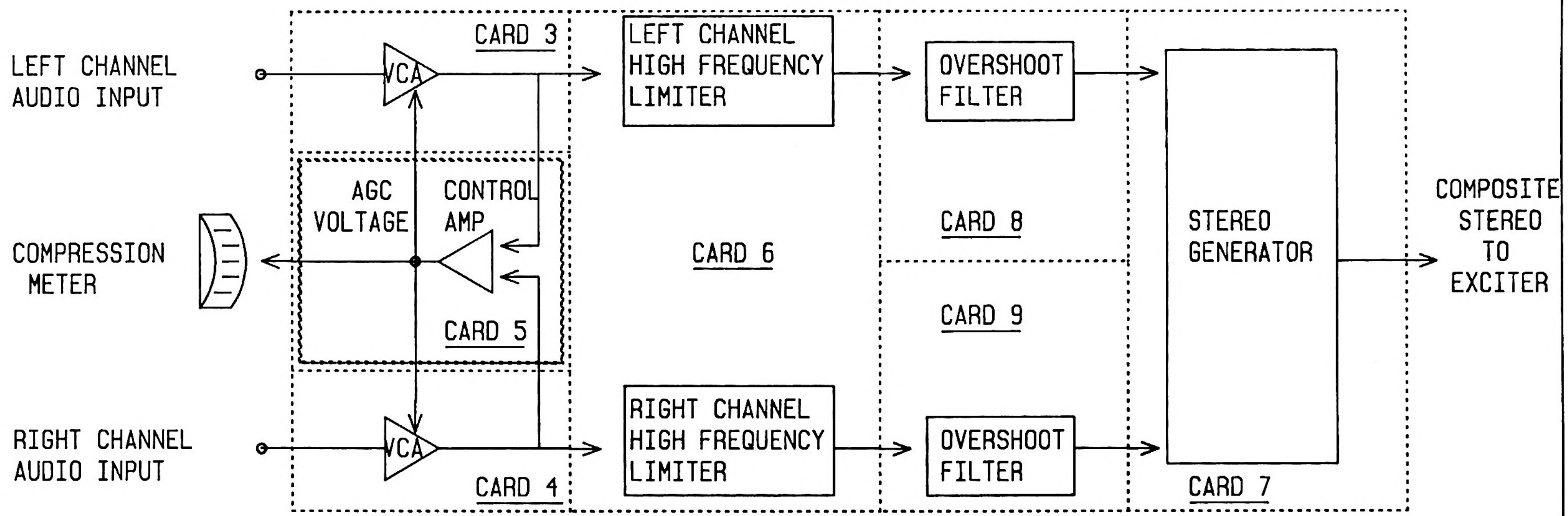
* Note: C10 and C11 may be mounted on the rear of the circuit board on early version RCF-1 Cards.

TEXAR
616 Beatty Rd. Monroeville, PA. 15146-1502 (412) 856-4276

RCF-1
Optimod® Replacement Card Five

Drawing #:87001	Sheet 1 of 1	Revision: A	Scale: None
Drawn by: David Van Allen DVA		Date: 1 Mar 87	CAD File: rcf3
Checked by: Glen Clark GC		Date: 18 MAR 87	Note:

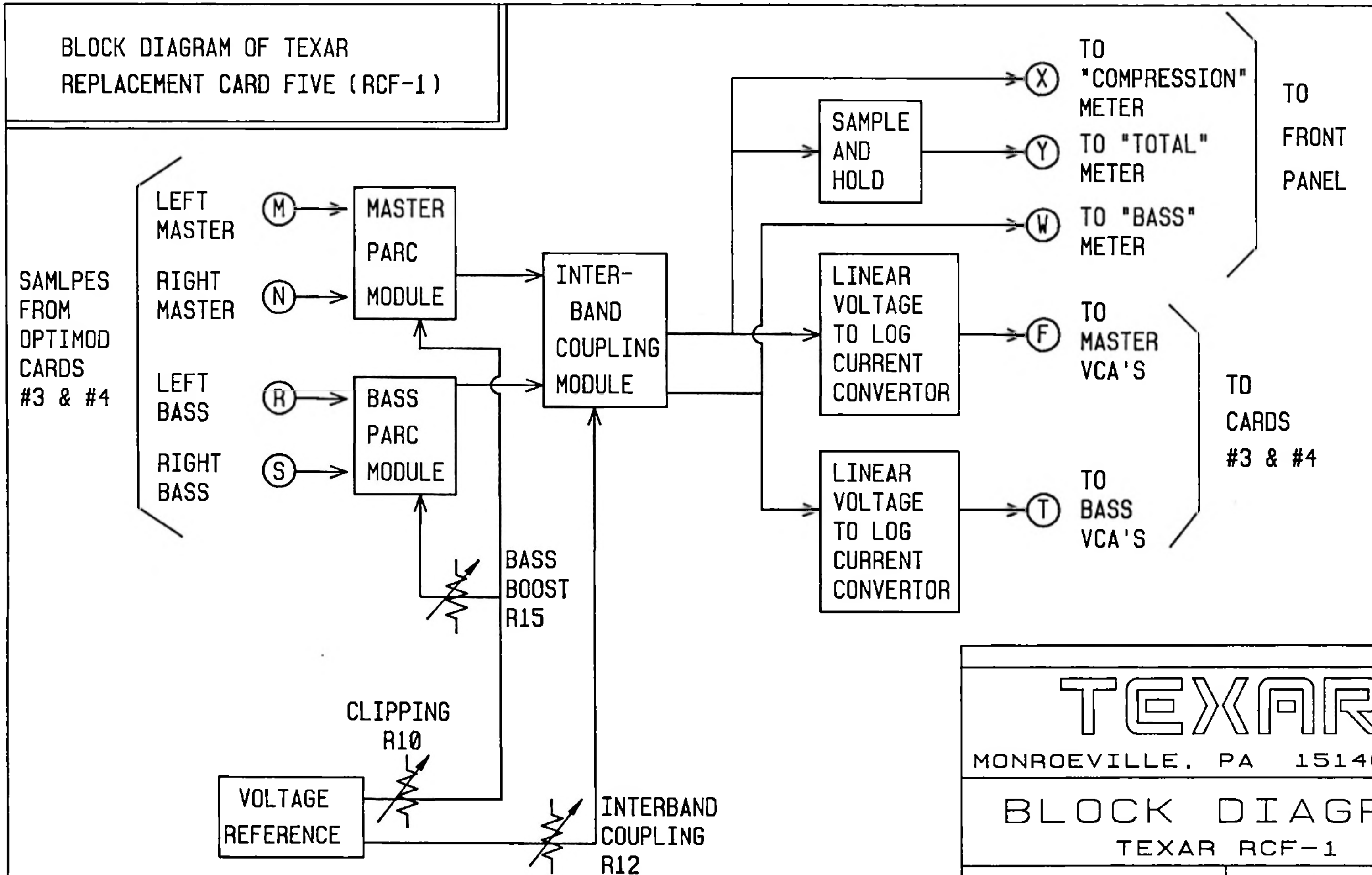
BLOCK DIAGRAM OF ORBAN OPTIMOD 8100 *



TEXAR	
MONROEVILLE, PA 15146-1502	
BLOCK DIAGRAM ORBAN OPTIMOD 8100*	
DRAWING #:87002	REVISION: A
DRAWN BY: MB	DATE: 1 Mar 87
CHECKED BY: G.C.	DATE: 8 MAR 87

* OPTIMOD 8100 IS A REGISTERED TRADEMARK OF ORBAN ASSOCIATES INC., SAN FRANCISCO, CA.

BLOCK DIAGRAM OF TEXAR
REPLACEMENT CARD FIVE (RCF-1)



TEXAR
MONROEVILLE, PA 15146-1502

BLOCK DIAGRAM
TEXAR RCF-1

DRAWING #: 87003	REVISION: A
DRAWN BY: MCB	DATE: 1 Mar 87
CHECKED BY: G.C.	DATE: 18 MAR 87

TEXAR TECHNICAL UPDATE

Product affected: RCF-1

Update release date: March 23, 1987

Condition addressed: "Fuzzy" sound produced by some RCF-1's, particularly when operated at low amounts of gain reduction

Some TEXAR RCF-1's produce a "fuzzy edge" on program material when operated as recommended, particularly when operated at low levels of gain reduction. In some cases, this condition also worsened steady-state distortion measurements. Laboratory research showed this was due to a diode operating in the "knee" area of its conduction curve.

Enclosed are three capacitors whose addition corrects this condition. They should be soldered, as described later, to the back side of the RCF-1 PC board. While this condition was apparent only on SOME RCF-1's, the modification is strongly recommended for ALL RCF-1's as a precaution. It should also be noted that this condition, if it does exist, is present only when the RCF-1 is mounted in the Optimod* card cage. It does NOT occur when the RCF-1 is operated via the card extender. For greater accessibility, RCF-1's are aligned and quality-control tested at the TEXAR factory in that very extender. As a result, this condition was not detected prior to shipment of the offending cards. We regret any inconvenience this may have caused you.

We will be happy to install these capacitors in your RCF-1 at no charge at the factory. However, as the amount of work is small and this would require being without the card for a period of time, most stations have indicated a preference to make the improvements themselves. If you would prefer to have these changes performed at the factory, please call the TEXAR Technical Support group (at 412-856-4276) for a RETURN MERCHANDISE AUTHORIZATION (RMA) number.

If you choose to perform these changes yourself, proceed as follows:

1) Lay the RCF-1 flat on the workbench with the component side down and the nylon card ejector in the upper, left-hand corner. The label "RCF-1" should be visible, upright, in the upper, left-hand corner, etched in the solder layer.

* Optimod is a registered trademark of Orban Associates, Inc. San Francisco, CA.

2) Connect one of the capacitors provided (all three are identical) between Y1, Pin 6 (Pin 1 is to the right; Pin 12 is to the left) and U4, Pin 3 (Pin 1 is in the upper, right-hand corner of U4 when viewed as described in Step 1 above).

3) Connect a second capacitor from U4, Pin 3 to the leftmost terminal of toggle switch S1. This is a ground connection. You should notice a heavy PC trace going from this switch terminal to Y1-12 above it.

4) Connect the third capacitor from U4, Pin 5 to this same ground on the toggle switch.

5) Re-install the RCF-1.

Physical placement of the capacitors is moderately critical. Do not attempt to place the capacitors at other locations on the RCF-1 that appear to be electrically equivalent.

Please feel free to call TEXAR Technical Support if you have any qusetions about this modification. The TEXAR factory will be operating at reduced capacity until April 6th, as many of our personnel will be manning the Exhibit booth at the Dallas, National Association of Broadcasters Convention. We will be able to give your call more immediate attention if you place your call after that date.

This sheet may be placed in the Users' Manual to signify that the modification has been completed.

Modification completed by: Brian A. CHASE

Date completed: 5-12-87

RCF-1 Serial #: ____________________