PRISM II AM

User's Manual

Gentner Broadcast Systems
1825 Research Way
Salt Lake City, Utah 84119
Phone: (801) 975-7200
Fax: (801) 977-0087

Prism II AM User's Manual

Printed in the United States of America. Gentner Communications Corporation reserves price and specification privileges.

Information in this manual is subject to change without notice.

Table of Contents

1

Introduction

Congratulations on your purchase of your new Prism II AM audio processor. The Prism II AM is a high performance four-band audio processor for AM broadcast use. The Prism II AM utilizes four intelligent, digitally controlled processor cards to achieve high apparent loudness while producing few processing artifacts.

Please refer to this manual if you have any questions or problems regarding the installation or use of your Prism II AM. If you cannot find an answer in this manual please contact:

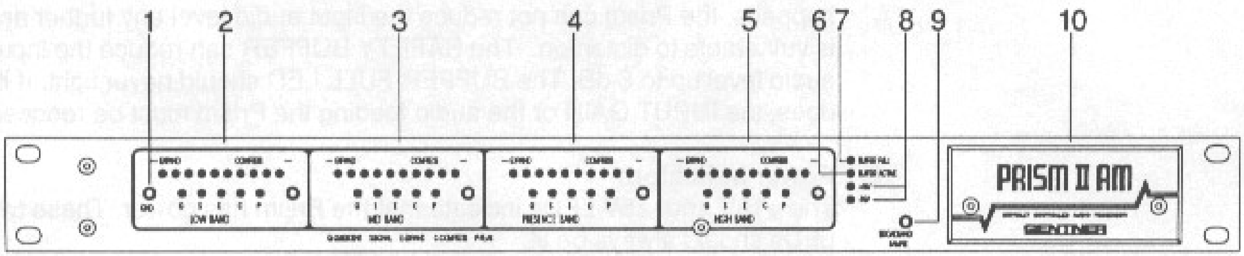
Gentner Broadcast Systems
1825 Research Way
Salt Lake City, Utah 84119
Telephone: (801) 975-7200
Facsimile: (801) 977-0087

Features and Benefits

- ***Digital Processing Control:*** Yields high apparent loudness with reduced listener fatigue.
- ***AMC-3 Final Limiter Card:*** Provides peak control and NRSC-1 compliance.
- ***Multiband Processing:*** Allows heavy processing in one band without "punching holes" in the rest of the audio spectrum.
- ***Individual Mix Level Controls:*** Allows the equalization of each band individually, without a separate program equalizer.
- ***Phase Rotator:*** Built-in for consistent loudness.
- ***Four Band Display:*** Front panel LEDs show the status of each band and the degree of expansion or compression.
- ***Density Control:*** Adjusts the amount of processing required by the station's format.
- ***Adjustable Gating Control:*** Provides "single ended" noise reduction.

Front Panel

The front panel provides bar graph displays, metering test points, and user adjustable controls. Each of the four bands has its own bar graph display. Removing the cover plate exposes various adjustable controls and mode switches.



1. **GND Connection**
   Electrical ground for connection of the ground terminal or the negative terminal of your test and measurement equipment.

2. **LOW BAND Display**
   Shows expansion and compression activitiy on the LOW BAND. For a more detailed description of this display refer to "Narrow Band Display" on page 1-10.

3. **MID BAND Display**
   Shows expansion and compression activitiy on the MID BAND. For a more detailed description of this display refer to "Narrow Band Display" on page 1-10.

4. **PRESENCE BAND Display**
   Shows expansion and compression activitiy on the PRESENCE BAND. For a more detailed description of this display refer to "Narrow Band Display" on page 1-10.

5. **HIGH BAND Display**
   Shows expansion and compression activitiy on the HIGH BAND. For a more detailed description of this display refer to "Narrow Band Display" on page 1-10.

6. **BUFFER ACTIVE Indicator**
The BUFFER ACTIVE LED indicates that the input audio is high enough that one of the narrow bands is reaching maximum compression. The SAFETY BUFFER compensates for this high level by reducing audio level being sent to the four bands. It is normal for the BUFFER ACTIVE LED to occasionally turn on. However, if it is frequently turning on, you should reduce the INPUT GAIN or reduce the audio level feeding the Prism.

7. **BUFFER FULL Indicator**
If the input level is high enough that it requires full attenuation by the SAFETY BUFFER, the BUFFER FULL LED turns on. When this happens, the Prism can not reduce the input audio level any further and is vulnerable to distortion. The SAFETY BUFFER can reduce the input audio level up to 6 dB. The BUFFER FULL LED should never light. If it does, the INPUT GAIN or the audio feeding the Prism must be reduced.

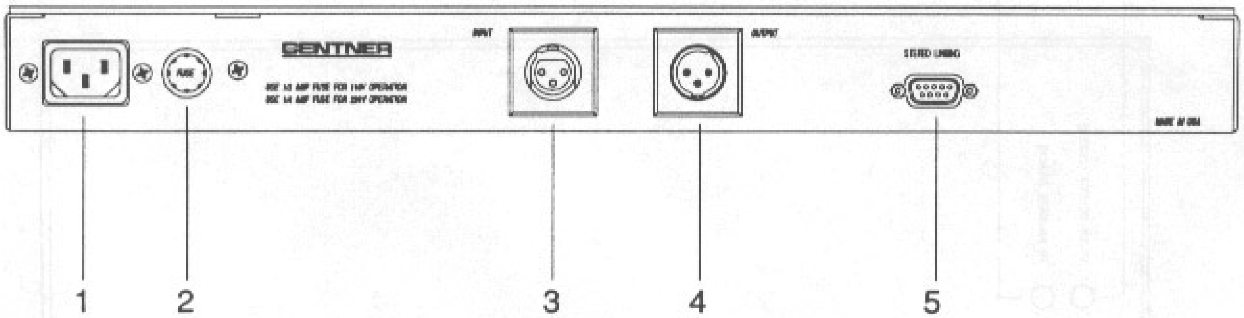
8. **Power Indicators**
The +15V and -15V LEDs indicate that the Prism has power. These two LEDs should always be lit.

9. **BROADBAND SAMPLE**
Contains the sum of the four audio bands to provide the complete spectrum of processed audio. This makes a convenient test point for your spectrum analyzer or RTA.

10. **Control Panel Cover Plate**
This removable plate covers various setup controls. After the setup process is complete, the Cover Plate should be replaced to protect the settings from being changed.

Rear Panel

The interconnection between the Prism II AM and the audio equipment is done via two XLR type connectors. Each of the conductors connected between the XLRs and the circuitry are RFI-shielded and transient-suppressed inside the unit.



1. **Power Entry Module**
Connect the supplied power cord between this module and the line voltage source. The Prism is shipped from the factory to be used with a 110 V power source. If you need 120 V operation, refer to "AC Line Voltage Select," in Chapter 2 for instructions on changing the input voltage.

2. **Fuse Holder**
Use a 1/2 amp slow-blow type fuse for 110 V operation or a 1/4 amp slow-blow type fuse for 220 V operation.

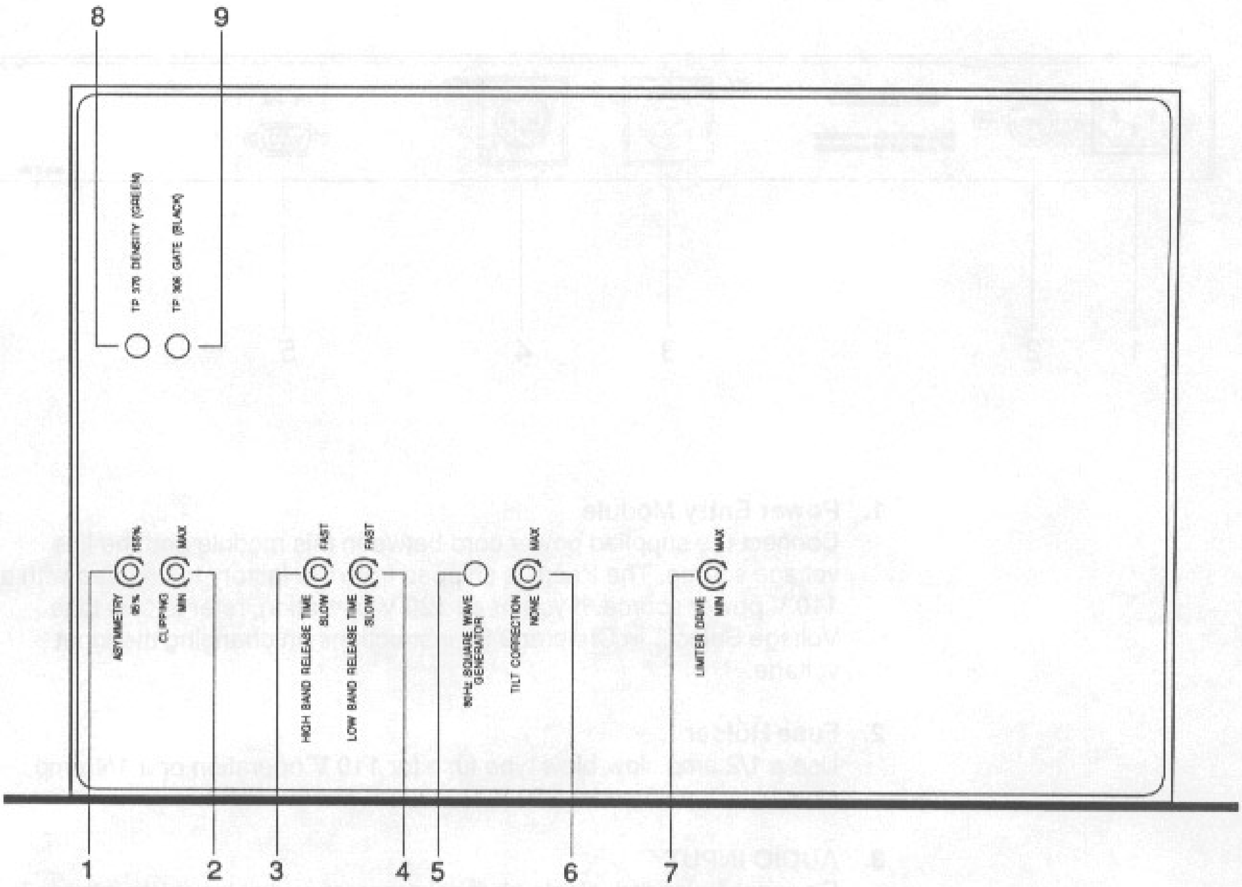
3. **AUDIO INPUT**
Recieves balanced, single-channel program audio. Input impedance is 600 ohm resistive terminating. Input level is 0 dBm to +12 dBm nominal.

4. **AUDIO OUTPUT**
Contains processed program audio. Output impedance is 200 ohms. Output level is +10 dBm nominal.

5. **STEREO LINKING**
Provides no function for the Prism II AM since the Prism II AM is not normally used in stereo applications.

Top Cover

The Top Cover provides access to the controls for the AM Modulation Controller (AMC-3) circuit. Also provided are measurement points for the DENSITY and GATE controls.



1. **ASYMMETRY CONTROL**
   Allows positve clipping threshold adjustment with respect to the negative threshold for asymmetrical modulation.

2. **CLIPPING CONTROL**
   Sets clipping level in the limiter section. Setting the control fully counter-clockwise causes minimal audio clipping. As the control is turned clockwise, more clipping action is acheived. This increases the coverage and loudness but causes more artifacts to appear in the audio.

3. **HIGH BAND RELEASE TIME**
   Sets the release time for the High band limiter. Fast release times yield higher average modulation and loudness while slow release times yield gentler limiter action for a more natural sound.

4. **LOW BAND RELEASE TIME**
   Sets the release time for the Low band limiter. Fast release times yield higher average modulation and loudness while slow release times yield gentler limiter action for a more natural sound.

5. **80 HZ SQUARE WAVE GENERATOR**
   This switch activates an 80 Hz square wave generator for use in setting up TILT CORRECTION.

6. **TILT CORRECTION**
   Causes the AM Modulation Control (AMC-3) board to compensate for the inaccurate waveform reproduction of some transmitters. Some transmitters are more inaccurate than others and the position of this control depends on your transmitter.

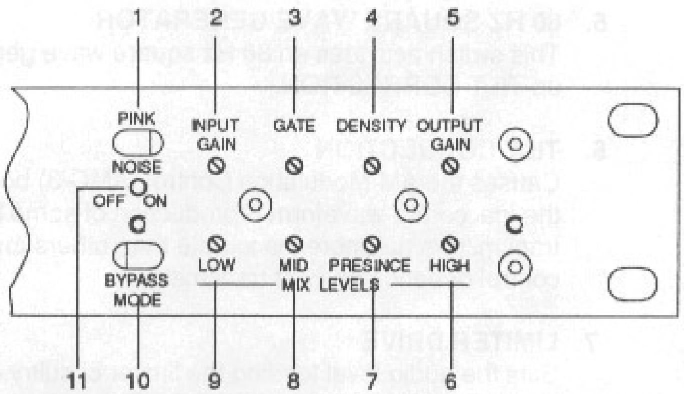
7 **LIMITER DRIVE**
   Sets the audio level feeding the limiter circuitry of the AM Modulation Controller (AMC-3). LIMITER DRIVE determines the trade-off between modulation density and signal quality.

8. **DENSITY Test Point**
   This test point provides a DC voltage that can be measured to aid in setting up the DENSITY control. This allows the DENSITY control to be set to an exact numerical setting rather than a visual approximation.

9. **GATE Test Point**
   This test point provides a DC voltage that can be measured to aid in setting the GATE control. This allows the GATE control to be set to an exact numerical setting rather than a visual approximation.

Control Panel

The CONTROL PANEL provides the adjustments and mode switches that control the audio processing. A removable plate covers this panel when adjustments are completed.



1. **PINK NOISE Switch**
   This switch turns on a pink noise generator for calibration purposes. The switch is recessed behind the panel to prevent accidental use.

2. **INPUT GAIN Control**
   Provides adjustment for the input gain of the Prism II AM to match the level of the audio being fed to the INPUT connector.

3. **GATE Control**
   Sets the level at which the Prism II AM gates on. The GATE control affects each of the four bands simultaneously. This control should be set at a point above the ambient noise or tape hiss level.

4. **DENSITY Control**
   Establishes the aggressiveness of the processing. It provides dynamic noise reduction, loudness control, and a gentle average level control. Lower settings provide a more "open sound" but modulation density is lower. At higher settings, the dynamic range of your source material is reduced, thus increasing modulation density. The effects of the DENSITY control setting can be seen on the bar graph displays. If the DENSITY control is at a lower setting, the fluctuation between the expand and compress LEDs will be slower. As the DENSITY control is increased, the fluctuation between the expand and compress LEDs will increase in frequency. As does the GATE control, this control adjusts all four bands simultaneosly.

5. **OUTPUT GAIN Control**
   Sets the audio level at the OUTPUT connector.

6. **HIGH MIX LEVEL**
   Provides adjustment for the amount of HIGH BAND audio fed to the mixing junction. Increasing the HIGH MIX LEVEL inceases the amount of HIGH BAND audio in the broadband processed audio.

7. **PRESENCE MIX LEVEL**
   Provides adjustment for the amount of PRESENCE BAND audio fed to the mixing junction. Increasing the PRESENCE MIX LEVEL inceases the amount of PRESENCE BAND audio in the broadband processed audio.

8. **MID MIX LEVEL**
   Provides adjustment for the amount of MID BAND audio fed to the mixing junction. Increasing the MID MIX LEVEL inceases the amount of MID BAND audio in the broadband processed audio.

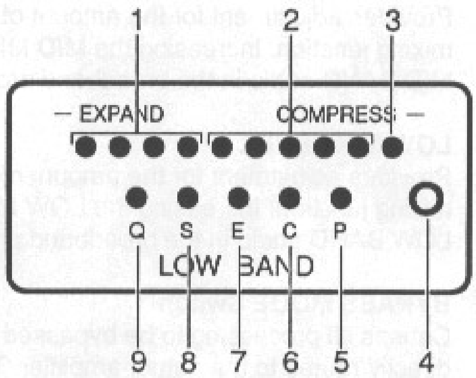
9. **LOW MIX LEVEL**
   Provides adjustment for the amount of LOW BAND audio fed to the mixing junction. Increasing the LOW MIX LEVEL inceases the amount of LOW BAND audio in the broadband processed audio.

10. **BYPASS MODE Switch**
    Causes all processing to be bypassed and allows the input audio to be directly routed to the output amplifier. This allows all controls on the front panel and on the top cover to be adjusted using the pink noise while maintaining a program flow from the studio to the transmitter.

11. **BYPASS LEVEL Control**
    Used to match the output audio level when in the BYPASS MODE, to the output audio level when operating in the normal mode.

Narrow Band Display

The action of each of the four digitally controlled processors is completely described by the NARROW BAND displays. Both the immediate action being performed and the range of action are displayed. Each band also has a test point which provides a sample of the audio in that band.



BAR GRAPH DISPLAY Ten LEDs are arranged in a bar graph fashion along the top of each band's display area. This bar graph is made up of Expand LEDs, Compress LEDs and a Compression Limit LED. Each band has a range of 20 dB from maximum expansion to maximum compression. Each LED represents approximately 2 dB of gain change.

During normal operation, the Expand and Compress LEDs flash back and forth. The activity of these LEDs is controlled by the DENSITY control and the program material.

If the input audio signal is too high, the entire bar graph may blank out. This indicates that the SAFETY BUFFER has been driven beyond its control range and that the input level needs to be reduced. This does not reflect an equipment malfunction or damage to the unit.

1. **EXPAND METER**
   These LEDs are green and indicate expansion.

2. **COMPRESS METER**
   These LEDs are yellow and indicate compression.

3. **COMPRESSION LIMIT LED**
   This LED is red and indicates that that the maximum amount of compression is being applied and that any increase in level will activate the safety buffer.

4. **NARROW BAND SAMPLE**
   Contains the audio signal of the processed narrow band.

5. **PEAK LED**
   Signifies that an audio peak has occurred which cannot be quickly controlled by the COMPRESS circuit. When these types of peaks occur, the PEAK circuit is activated to clip the peak. Under normal operation this LED flickers frequently but does not remain lit for any sustained period.

6. **COMPRESS LED**
   Indicates that the processor is decreasing gain. During normal operation, the COMPRESS LED and the EXPAND LED flash back and forth. The degree of activity is controlled by the DENSITY control.

7. **EXPAND LED**
   Indicates that the processor is increasing gain. During normal operation, the EXPAND LED and the COMPRESS LED flash back and forth. The degree of activity is controlled by the DENSITY control.

8. **SIGNAL LED**
   Indicates audio presence within a given band. This function is frequently called the "gate." The signal must reach the gating threshold, as set by the GATE control, for the SIGNAL LED to turn on. The Prism expands and compresses audio only when the SIGNAL LED is lit.

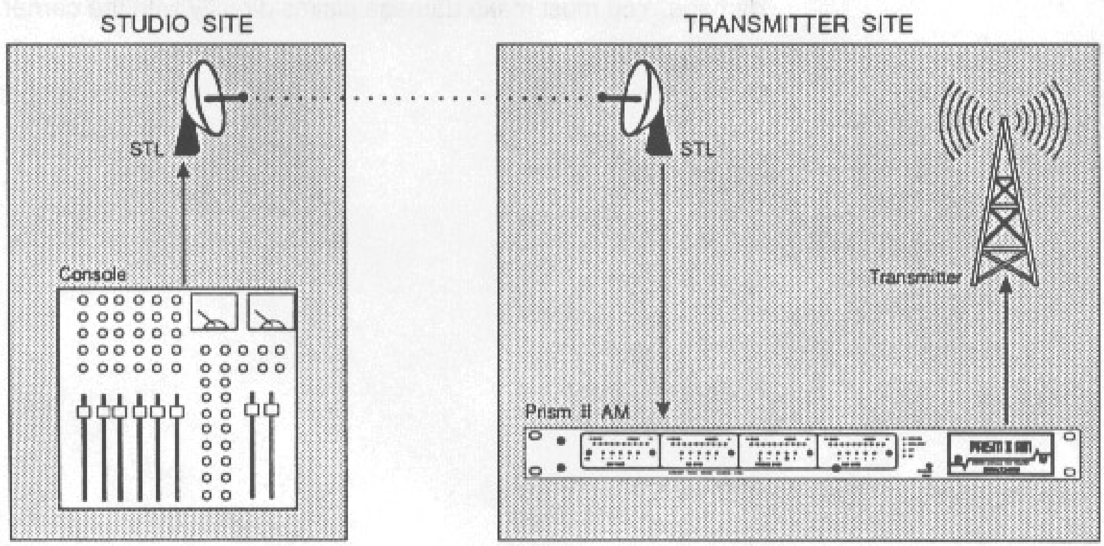
   If the signal in a particular band falls below the gating threshold, the SIGNAL LED will go out and the gain of the band will freeze. The processor will hold 1.7 seconds without changing gain. If the signal has not returned by the end of the 1.7 seconds, the QUIESCENT LED will turn on and the gain will slowly recover to the QUIESCENT position.

9. **QUIESCENT LED**
   When no signal is present in a band or if the audio is below the GATE level, the QUIESCENT LED will turn on and the LED bar graph will settle to the center position. Since the number of LEDs is even, there is not an exact middle point, so the display settles on the first yellow LED.

2 *Installation*

The Prism II AM should be mounted at the transmitter site. Devices such as transformers, telephone lines, and STL's connected between the Prism II AM and the transmitter can disturb the amplitude or phase relationships and degrade the modulation performance and peak control.

If you wish to pre-process and limit audio for maximum density without overdriving your transmission lines, you may want to try using the Prism Li at the studio along with the Prism II AM at the transmitter. Contact the Gentner Sales department at 801-975-7200 for details.



The installation procedure consists of setting the internal jumpers to the correct setting for your application, configuring the power supply for the proper power source, mounting the Prism in a rack, making the audio connections, and setting the controls to their starting points.

Once you have completed all the processes in this chapter, the Prism is ready for operation in your audio chain. When you have the Prism operating in your audio chain, refer to Chapter 3 for information on all the controls and how to create your own sound.

Unpacking

Carefully unpack your shipment and check for damage. Verify that all the parts listed below are included in your package. If anything is missing, please notify Gentner immediately.

- ☐ Prism II AM unit
- ☐ This manual
- ☐ Power cord
- ☐ Rack screws and cups
- ☐ Warranty card
- ☐ "Greenie" screwdriver
- ☐ Male XLR connector
- ☐ Female XLR connector

If you notice any damage to the shipment, notify your shipping carrier immediately. Retain the original boxes and packing material for inspection by the carrier. Gentner is not responsible for shipping damage. You must make damage claims directly with the carrier.

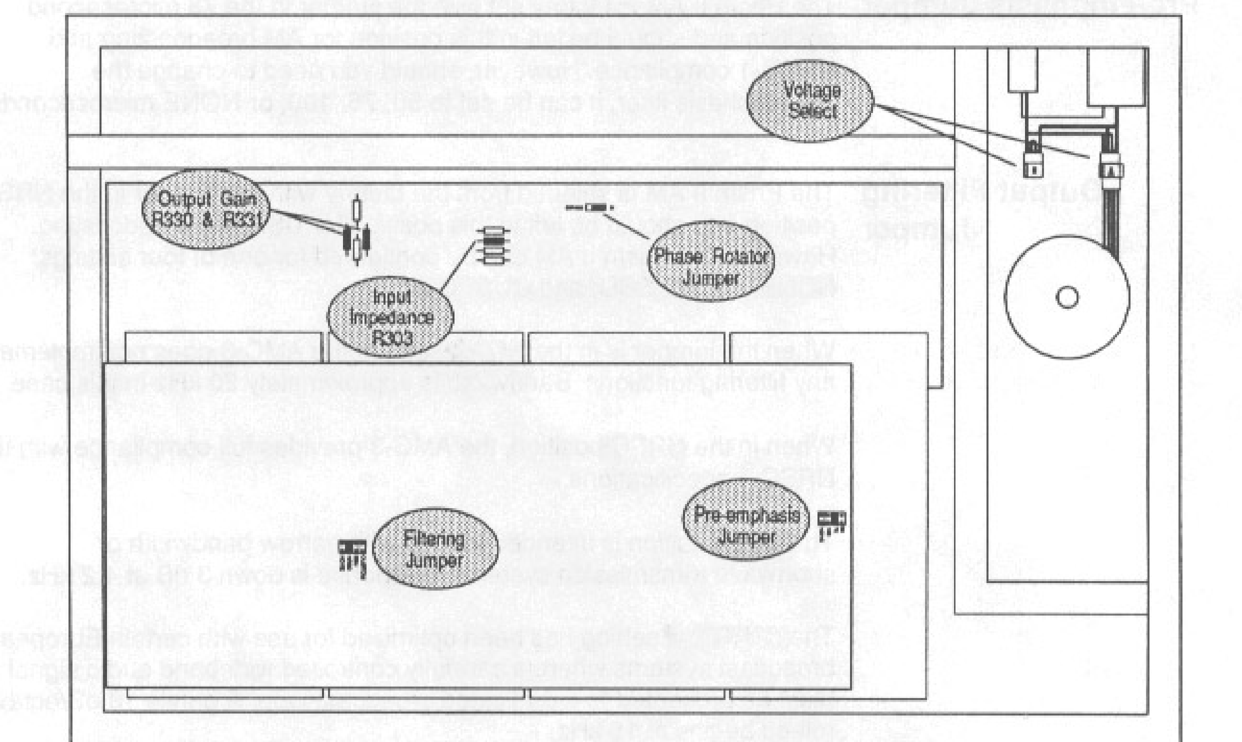
Internal Settings

Inside the Prism chassis are several options that, depending on your application, may need to be changed. Below are the default settings as shipped from the factory.

Phase Rotator Jumper	IN
Pre-emphasis	75 µs
Output Filtering	NRSC
AC Line Voltage	120 VAC
Input Impedance	600 ohms
Output Gain	+10 dBm

The location of each of these options is shown in the diagram below. To access them, the top cover of the Prism must be removed by removing eighteen screws holding down the cover.

IMPORTANT: Before removing the top cover, disconnect the Prism II AM from AC power to avoid electrical shock.



Phase Rotator Jumper

The Prism is shipped from the factory with the Phase Rotator enabled (labeled as 'IN' on the circuit board). It is recommend that the Phase Rotator is left enabled for all applications.

If you wish to disable the Phase Rotator, move the jumper to the BYPASS position. This is the left-hand position with the jumper covering pins 1 and 2.

The Phase Rotator increases processing efficiency by insuring that positive and negative peaks are of equal amplitude.

Although most recorded program material and live female voice signals are fairly symmetrical, live male voices tend to be asymmetrical. The Phase Rotator implements an inaudible, linear process to remove asymmetry from the incoming program audio. This provides significant benefits whether the station intends to transmit symmetrical or asymmetrical positive peak modulation.

It should be noted that the Phase Rotator is acoustically transparent and does not detract from low frequency, transient response as do some phase rotators. The Phase Rotator employs a complex conjugate pair of poles near the frequency axis where some designs employ simple poles on the real axis.

Pre-emphasis Jumper

The Prism II AM is factory set with the jumper in the 75 microsecond position and should be left in this position for AM broadcasting and NRSC-1 compliance. However, should you need to change the pre-emphasis filter, it can be set to 50, 75, 100, or NONE microseconds.

Output Filtering Jumper

The Prism II AM is shipped from the factory with the jumper in the NRSC position and should be left in this position for USA AM broadcasting. However, the Prism II AM can be configured for one of four settings: NONE, NRSC, EBU, and CUSTOM.

When the jumper is in the NONE setting, the AMC-3 does not implement any filtering functions. Bandwidth is approximately 20 kHz in this case.

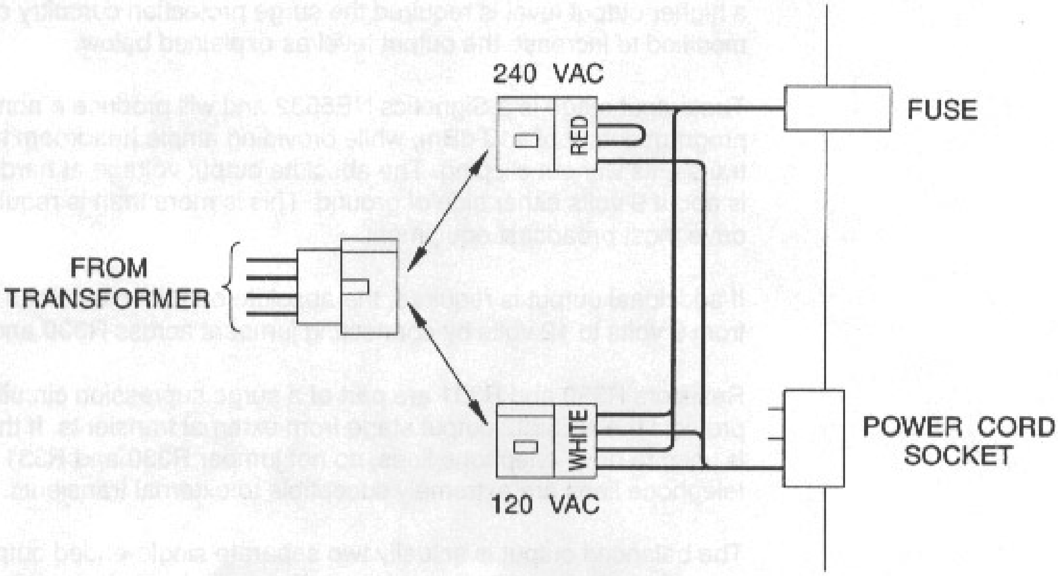
When in the NRSC position, the AMC-3 provides full compliance with the NRSC-1 specifications.

The EBU position is intended for use with narrow bandwidth or shortwave transmission systems. Response is down 3 dB at 4.2 kHz.

The CUSTOM setting has been optimized for use with certain European broadcast systems where a carefully controlled wideband audio signal must be presented to a dedicated broadcast loop. A gentle 18 dB/octave roll-off begins at 15 kHz.

AC Line Voltage Select

The Prism II AM is shipped from the factory ready to use with a 120 VAC 50/60 Hz power source (unless otherwise marked). The Prism can be changed to accept a 240 VAC 50/60 Hz power source. To change the power input, follow the procedure below.



- ☐ DISCONNECT THE PRISM II AM FROM AC POWER. Unplug the electrical power cable from the rear panel.
- ☐ Remove the lid by removing the eighteen flat-head machine screws.
- ☐ Locate the AC power wiring harness. Looking from the front of the unit, the harness is on the right-hand side, towards the rear.
- ☐ For 120 VAC operation, connect the harness to the white molex connector.
- ☐ For 240 VAC operation, connect the harness to the red molex connector.
- ☐ Replace the lid and the eighteen machine-head screws.
- ☐ Replace the external fuse with a 1/4 amp slow-blow type fuse for 240 VAC operation.

Input Impedance

All Prisms are shipped from the factory with the input wired for 600 ohms resitive, terminating. If your program audio source is already terminated, the input can be converted to 10K (nominal) bridging, by removing R303 on the motherboard.

Output Gain

The Prism II AM can provide an output level of up to +10 dBm nominal. If a higher output level is required the surge protection curcuitry can be modified to increase the output level as explained below.

The output stage is a Signetics NE5532 and will produce a nominal program output of +10 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 is required to drive most broadcast equipment.

If additional output is required, the absolute output voltage can be raised from 9 volts to 12 volts by connecting jumpers across R330 and R331.

Resistors R330 and R331 are part of a surge supression circuit which protects the NE5532 output stage from external transients. If the output is used to drive telephone lines, do not jumper R330 and R331 since telephone lines are extremely suceptible to external transients.

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. However, do not ground either of the output terminals.

Initial Settings

To begin with, put the controls at the settings noted below. These settings will provide a starting point for operating the Prism II AM. Then once the Prism II AM is installed and operating, you can fine-tune the controls.

Control Panel Settings

Remove the Control Panel Cover Plate, using a Phillips screwdriver, to expose the controls. Using a small, flat-bladed screwdriver, put the controls in the following positions.

INPUT GAIN . . . . .fully counter-clockwise
GATE .9 O'clock
DENSITY . . . . .fully clockwise
OUTPUT GAIN . . . . .fully counter-clockwise
LOW MIX . . . . .12 O'clock
MID MIX . . . . .12 O'clock
PRESENCE MIX . . . . .12 O'clock
HIGH MIX . . . . .12 O'clock
PINK NOISE . . . . .off
BYPASS MODE . . . . .off

Top Panel Settings

These controls are accessed through holes in the top cover. Using a small, flat-bladed screwdriver, put the controls in the following positions.

ASYMMETRY . . . . .fully counter-clockwise
CLIPPING . . . . .fully clockwise and back off 1/8 turn
HIGH BAND RELEASE TIME . . . . .fully counter-clockwise and back off 3/4 turn
LOW BAND RELEASE TIME . . . . . .fully counter-clockwise and back off 3/4 turn
TILT CORRECTION . . . . .fully counter-clockwise
DRIVE . . . . .fully counter-clockwise and back off 1/2 turn

Mounting in the Rack

The unit is completely self-contained in a single 1¾" 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.

Mount your Prism II AM in a standard 19" equipment rack, making sure not to block any of the ventilation holes. Insure that a free flow of air gets to the unit while it is operating. Gentner recommends and operating environment between 32 and 110 degrees Fahrenhiet.

The Prism II AM should not be mounted in a strong induction field.

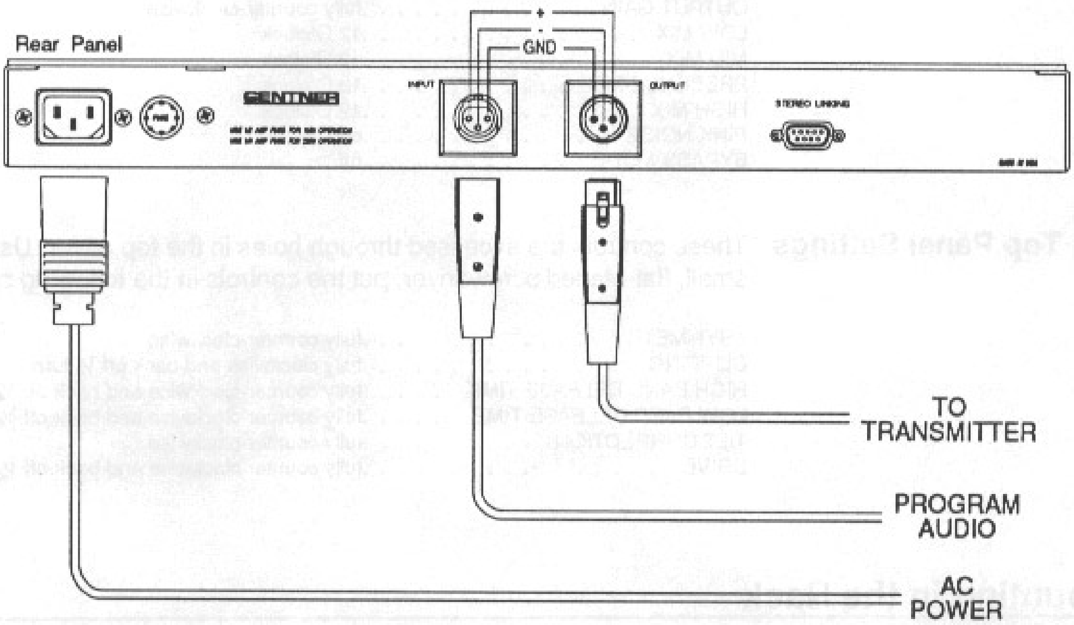
Note: Keep in mind, when you mount your Prism II AM, that may need access to the controls on the top. You can either leave enough rack space above the Prism II AM that you can reach in with a screwdriver, or you can leave a service loop in the wiring to allow you to pull the Prism II AM out of the rack.

Connections

☐ Connect a cable between the output of your audio console and the INPUT of the Prism II AM.

☐ Connect a cable between the OUTPUT of the Prism II AM and the input of the transmitter.

☐ Connect the Prism II AM to power using the supplied power cord. Make sure that the Power Supply is configured for the correct power source prior to connecting power.



Use of Modulation Enhancers

Any integral clipper in the transmitter should be disabled for proper operation of the Prism II AM.

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 expose 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.

For all other transmiters, consult the documentation that came with the transmitter.

Adjustments

After making the audio connections and applying power, you are ready to adjust the controls to the recommended settings. Only the adjustments neccessary for operation are outlined below. All other adjustments are more of a subjective nature and can be adjusted after you have become more familiar with your Prism II AM. Refer to Chapter 3 for detailed information about all of the controls.

Adjust the controls outlined in the steps below.

MIX Levels

☐ Put the PINK NOISE switch in the ON position.

☐ Connect the negative terminal of a VOM, such as a Simpson 260, to the GND test point (black) on the front panel of the Prism II AM.

☐ Connect the positive terminal of the VOM to the LOW band test point (red) and adjust the LOW MIX control for an output of 4.0 dBm.

☐ Move the positive terminal of the VOM to the MID band test point (orange) and adjust the MID MIX control for an output of 4.0 dBm.

☐ Move the positive terminal of the VOM to the PRESENCE band test point (yellow) and adjust the PRESENCE MIX control for an output of 2.0 dBm.

☐ Move the positive terminal of the VOM to the HIGH band test point (green) and adjust the HIGH MIX control for an output of -1.0 dBm.

☐ Put the PINK NOISE switch in the OFF position.

TILT CORRECTION

Note: During this adjustment it is imperative that the modulation does not exceed 50%. Higher modulation levels could cause permanent damage to the modulation transformer. To prevent damage to the transmitter, begin with the front panel OUTPUT GAIN control fully counter-clockwise.

☐ Connect an oscilloscope to an RF sample of the transmitter output and set the trace speed to 2 mS/division.

☐ Using the eraser end of a pencil (or similar object) press and hold down the 80 Hz SQUARE WAVE GENERATOR switch, accessed through the top cover.

☐ Turn up the OUTPUT GAIN control until the modulation envelope indicates approximately 50% modulation. Verify this level with a modulation monitor.

☐ Adjust the TILT CORRECTION control for a flat or level top on the oscilloscope waveform.

INPUT GAIN

☐ Apply program audio to the Prism II AM at a normal level.

☐ Adjust the INPUT GAIN control so that the average position of the MID band LED bar graph is the fourth LED from the right end. This is the yellow LED under the letter "O" in the word "COMPRESS."

Note: 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 bar-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 generalizations, however, and exact levels will depend on the instant program input.

☐ The BUFFER ACTIVE and the BUFFER FULL LEDs should not be illuminated.

OUTPUT GAIN

☐ Apply typical program audio to the Prism II AM at a normal level.

☐ Connect an oscilloscope to an RF sample of the transmitter output.

☐ 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 too little modulation.

Note: The OUTPUT GAIN is effected by the adjustment of the CLIPPING control and the ASYMETRY control. If either of these two controls is adjusted, the OUTPUT GAIN may need to be re-adjusted.

Ready for Operation

Your Prism II AM is now ready for operation in your broadcast audio chain.

3 Creating Your Own Sound

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

Since the early 1960s, 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 that it is constant and the station can be comfortably listened to until the signal-to-noise ratio falls below a certain constant — if the modulating voltage on a given carrier is doubled, you 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.

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, and the more people contained within the station's service area, the greater was the station's potential listening audience (cumes).

The primary purpose of audio processing in the past was to increase potential cumes. Additionally, some broadcasters felt there was some psychological advantage or attraction to listeners to the station which appeared to be the most powerful.

Today, modulation levels are only a percentage point or two away from their permitted maximums. The room simply does not exist for significant additional increases in modulation levels.

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 *average* number of listeners while not increasing the *total* number of listeners, we must keep each listener for a longer 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, much scrutiny is also given to removing negative program elements, which tend to make listeners tune out. Time spent listening can be increased by removing program elements which listeners find fatiguing.

The Prism II AM 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 unnatural sound, or similar processing artifacts. To accomplish this, the Prism II AM employs *digital technology* in its control circuits. Previous generations of audio processors employed *analog technology.*

Analog processors could do two things: expand and compress. Digitally controlled processors can do three things: expand, compress, and, if necessary, *do nothing.* While this may appear, on first inspection, to be a small 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. As 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.

To increase loudness in an analog controlled processor, you shorten the attack and recovery time constants. This increases the density but makes the compression level go up and down all that much faster, making 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.

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 Prism II AM will make only one-third to one-half as many AGC voltage reversals as will an analog processor.

Input Gain Control

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 bar graph) 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 different when driven hard than when driven at 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 "dense." 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 a conventional processor's action, they are correct.

By contrast, the acoustic sound of the Prism II AM does not change with changes in absolute input level. The charge and discharge rates of the timing capacitors of the Prism II AM'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 recommended level.

Gate Control

Observe the operation of the SIGNAL LEDs during normal program audio. 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. The above observations are only guidelines. Music and voice are moving targets. They are best described statistically, so it is not surprising that the Prism II AM'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 LEDs 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 will usually be about nine o'clock.

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 through an access hole in the chassis lid on the left hand side. There are two holes — TP306 is the rightmost 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 0.5 volts. These voltages are DC.

This control can be fine tuned without disturbing subsequent adjustments. If you find 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 source material are missing on the air, decrease the GATE control position.

Density Control

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 lower settings it provides dynamic noise reduction, loudness control, and a gentle average level control. Rotating this control clockwise provides all of the above functions, plus a denser average modulation level.

The setting of the DENSITY control can be quantized exactly, as the density reference voltage appears on TP307, the GREEN test point on the motherboard. This test point is accessible through an access hole in the chassis lid on the left hand side. There are two holes — TP307 is the rightmost of the two (the other is TP306, the GATE reference buss).

The normal setting for radio application (fully clockwise) will produce approximately 4.25 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 of the Prism II AM. This action is adjusted by the VCA DRIVE control on the AMC-3 AM Modulation Controller board.

Output Gain Control

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

Using music as a program source, observe an RF envelope display on an oscilloscope. 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.

Mix Levels

The MIX LEVEL controls are 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 called the "spectral mix."

Setting the spectral mix on a multi-band processor has typically been a time-consuming project, largely because there are few reference points to go by. To properly adjust any system, you must have some type of quantizable feedback from that system. The subjective nature of the human ear makes it anything but a laboratory instrument.

One repeatable and easily interpreted method of setting multi-band spectral mix levels is to use a Real Time Analyzer (RTA). On the Prism II AMs, the recombined output of the four bands is available on the front panel on the white test point labeled BROADBAND SAMPLE.

A calibration and setup circuit is included in the Prism II AM 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 Prism II AM's internal precision PINK NOISE generator and these test points, it is possible to accurately quantize the spectral mix. This is instrumental in arriving at a chosen mix, 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 U305B). As a result, matching physical positions of the MIX LEVEL control shafts does not indicate "flat" response. Judgments regarding the spectral mix should be made only by measuring the voltages appearing at the front panel test points.

The PINK NOISE switch automatically removes the DENSITY control from the circuit and forces the DENSITY buss to 5 volts for pink noise adjustments. 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.

Because of the pulsating nature of pink noise, particularly in the two lower frequency bands, an analog meter (one with a mechanical pointer) must be used for these measurements. A digital readout meter cannot be used.

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.

The following items should be kept in mind as the mix levels are adjusted.

- 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.
- AM stations operating into particularly narrow band directional antenna systems may find additional coloration is desirable to compensate for the bandwidth of the antenna.
- AM stations using the NRSC jumper setting on the AMC-3 board may want to reduce the HIGH band MIX LEVEL and perhaps the PRESENCE band mix. Center frequencies for these bands are 11.7 kHz and 2,040 Hz respectively. The NRSC pre-emphasis curve provides for approximately 10 dB of boost at 9.5 kHz and 3 dB of boost at 2,122 Hz.

Antenna Patterns

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, and 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 or she 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 extent of his coverage, but desiring improvements in the signal's fidelity, 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 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 "Computer-Optimized Directional Antenna Patterns Improve AM Coverage," are available from Glen Clark & Associates, (404) 499-1392.

Spectral Mix Limit

CAUTION : Care should be exercised when adjusting the spectral mix not to operate all four controls near the top of their range. The gain of the summing amplifier (U101A 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 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. The output stage can produce voltage swings to approximately 9 volts either side of ground (12 volts with the surge suppression 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.

An approximate rule of thumb is to add the mix levels at the front panel test points and see if they total 20 dB or greater. If they do, clipping is likely and the settings should be reduced. Also, any one band should not be 6 dB or higher.

If you have arrived at a desired spectral mix by trial-and-error and 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.

For example: Front panel voltages of 6, 7, 8, and 5 dBm are encountered at the test points. The sum of the values is 26 and two of the values are above 6 dB, 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.

Prism II AMs have a front panel test point marked BROADBAND SAMPLE. This is the output of U101A. 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 +/–13V.

Performance Testing

There is 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 aberrations relates to the gate circuit. For tones of certain amplitudes, when the tone is in the center of one particular band, the corresponding M-102 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-102 processor cards for both bands will expand to capture the signal. Unlike the previous case where only one band provided significant output, now both M-102s 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 Prism II AM, but is a function of all multi-band processors.

It is best to make adjustments and draw information only when the Prism II AM 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.

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.

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.

Bypass Mode Operation

The proof function causes the bandpass filter and the M-102 processor cards to be removed from the circuit when the BYPASS switch is in the ON position. This permits the Prism II AM to do two functions simultaneously: adjust the spectral MIX LEVEL controls under pink noise conditions while maintaining the normal program flow from the studio to the limiting device which follows the Prism II AM.

Adjusting The Gain Of The Bypass Mode

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, so that it is not on an even plane with the rest of the front panel controls. Use a thin flat-bladed screwdriver to access this control, such as an Xcelite "Greenie." 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 the BYPASS GAIN control will have any effect on the output signal. The OUTPUT GAIN control is out of the circuit.

Note: The phase of the output audio is reversed 180 degrees when in the BYPASS mode.

Clipping Control

The CLIPPING control and the OUTPUT GAIN are adjusted interactively to set the modulation level of the transmitter. Reference must be made to the stations modulation monitor or calibrated oscilliscope to set these two controls. The correct setting of the CLIPPING control depends on the processing goal. Clockwise rotation increases the amount of clipping. Fully counter-clockwise rotation allows program peaks to pass with minimal clipping. Maximum loudness and coverage requires use of as much clipping as can be tolerated.

Note: The OUTPUT GAIN control and the ASYMMETRY control must be re-adjusted any time the CLIPPING control is adjusted.

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%.

The ASYMMETRY control modifies the action of the clipper circuit. As the control is rotated clockwise, the positive side of the modulating wave can pass with greater amplitude than the negative half. This allows AM stations to modulate up to 125% positive and greater if the transmitter equipment is capable of this.

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 paragraphs which follow. Also, the engineer who does not desire to modulate asymmetrically may set the ASYMMETRY control fully counterclockwise (symmetrical).

Note: The amount of asymmetry available from the processor depends on the amount of clipping in use, so the setting of the CLIPPING control should be set before adjusting the ASYMMETRY control.

Determine if the phase polarity from the Prism II AM to the transmitter is correct. Using music as a program source, drive the transmitter to 50% modulation with the ASYMMETRY control fully counterclockwise. 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 is higher with the switch set to the negative position, reverse the polarity of the connections at the Prism II AM output.

Turn the ASYMMETRY control fully counterclockwise. Connect an oscilloscope 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 Prism II AM 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 oscilloscope and modulation monitor indicate that the amplitude of the positive peaks are increasing as the control is turned. The action of this 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 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.

High Band Release Time

The AM Modulation Controller (AMC-3) board splits the audio into two bands—a High band and a Low band. The release time constant for the High band can be varied by adjusting the High Band Release Time control.

Rotating the control clockwise quickens the release time of the limiter. Fast release times yield higher average modulation and more loudness. However, increased distortion is a byproduct.

Rotating the control counter-clockwise slows down the release time of the limiter. Slow release times yield gentler limiter action for a more natural sound. The range is from 1100 milliseconds to 100 milliseconds.

If maximum coverage and loudness is desired, adjust the control clockwise until the point of objectional artifacts (distortion) is reached, then slightly back of the control in the counter-clockwise direction.

Low Band Release Time

The AM Modulation Controller (AMC-3) board splits the audio into two bands—a High band and a Low band. The release time constant for the Low band can be varied by adjusting the Low Band Release Time control.

Rotating the control clockwise quickens the release time of the limiter. Fast release times yield higher average modulation and more loudness. However, increased distortion is a byproduct.

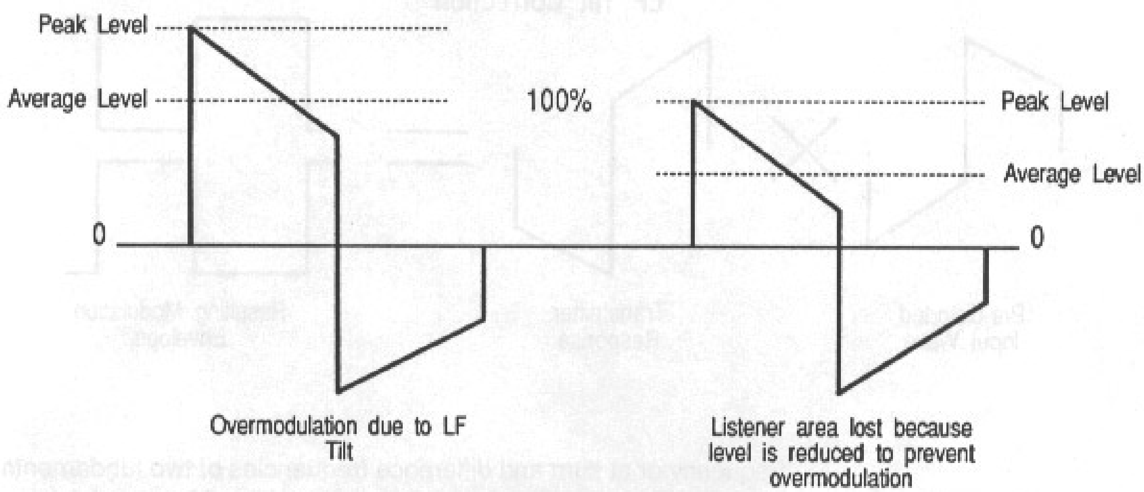
Rotating the control counter-clockwise slows down the release time of the limiter. Slow release times yield gentler limiter action for a more natural sound. The range is from 1100 milliseconds to 100 milliseconds

If maximum coverage and loudness is desired, adjust the control clockwise until the point of objectional artifacts (distortion) is reached, then slightly back of the control in the counter-clockwise direction.

Low Frequency Tilt Correction

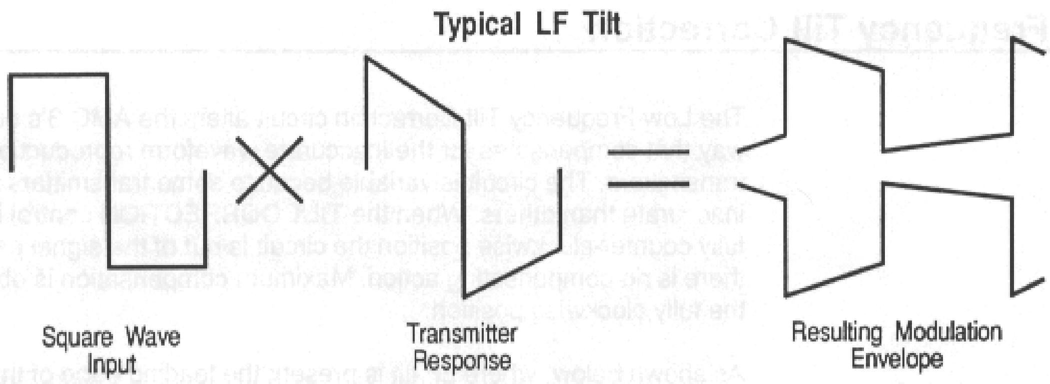
The Low Frequency Tilt Correction circuit alters the AMC-3's output in a way that compensates for the inaccurate waveform reproduction of some transmitters. The circuit is variable because some transmitters are more inaccurate than others. When the TILT CORRECTION control is In the fully counter-clockwise position the circuit is out of the signal path and there is no compensating action. Maximum compensation is obtained in the fully clockwise position.

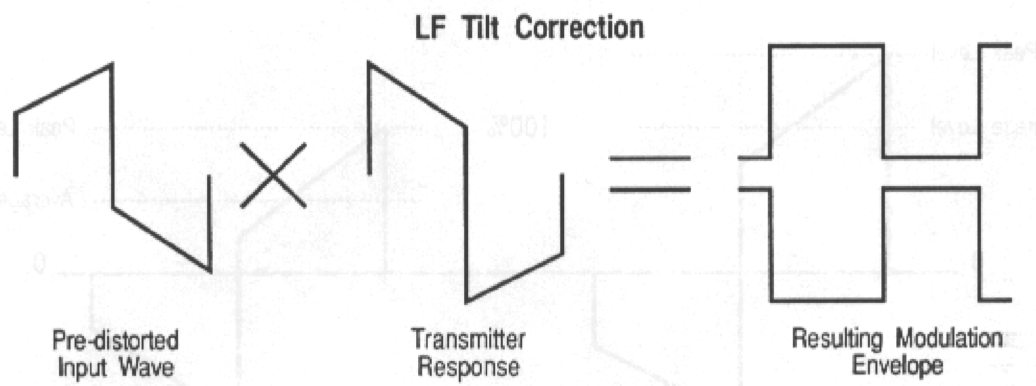
As shown below, where LF tilt is present the leading edge of the output square wave has an amplitude substantially higher than that of its average value. As a result, the amplitude of the input signal must be reduced substantially to prevent overmodulation. This reduces a station's average modulation.



The 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.

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.

Limiter Drive Control

The LIMITER DRIVE control determines the trade-off between modulation density and signal quality. 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.

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

Notice that the LIMITER DRIVE control adjusts the amount of gain reduction in the VCA. It does not control the degree of clipping. Clipping activity is adjustable using the CLIPPING control.

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 modem 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 into which group a given transmitter falls. 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.

Phase Rotator

The purpose of the Phase Rotator is to increase the modulation level by insuring that positive and negative peaks are of equal amplitude. Leaving the Phase Rotator enable is recommended for all applications, even whee asymmetrical AM modulation is desired. This apparent contradiction will be explained later in this section.

Although most recorded program material and live female voice signals are fairly symmetrecial, live male voices tend to be asymmetrical. The Phase Rotator 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 Phase Rotator is acoustically transparent and does not detract from low frequency transient response as do some phase rotators of other designs. The Phase Rotator 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 flast 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 Phase Rotator is obvious. However, even where asymmetrical modulation is desired, the Phase Rotator is vital for optimum performance.

There are two schools of thought regarding handling the polarity of symmetrical program input where an asymmetrical output is desired. 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 may not be 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. What 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 perceives.

This effect is significant and disorienting. Most announcers find it very distracting while trying to work a live microphone. 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 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 occurrence 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 under-utilize 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 occurrence) 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 previous 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. The two inherent problems described above cannot be overcome.

The phase rotation, described below, is a much more viable solution.

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 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 modulation.

4

TROUBLESHOOTING

Many test points are provided in the Prism II AM to facilitate troubleshooting.

Two keys to quick and accurate troubleshooting are provided by Drawing #8611, "Simplified Block Diagram" and Drawing #8612, "AUDIO PRISM Circuit Boards" in Appendix D. These drawings provide an overview of the entire system at a glance. Drawing #8611 is a functional block diagram and Drawing #8612 is useful as a location diagram.

This troubleshooting section is only intended to serve as a general guide to the system. This section provides non-technical persons with enough information to make basic checks and technical persons with the overview necessary to make quick sense of the schematics.

Power Supply

With power applied, check the front panel LEDs labeled "+15V" and "–15V." They should both be lit at a normal brightness and with equal intensity. If one is extinguished or dim, disconnect AC power input to the unit immediately. Disconnect the load from the power supply by disconnecting plug P2 from the connector labeled "J2" of the PS-2A power supply.

Re-apply power and measure the +15V and –15V output at J2. If they are normal, then one of the other boards in the system is probably loading down the power supply. Isolate the defective board and replace it. Another possibility is that one of the voltage regulators on the PS-2A power supply board is weak. Replace it if necessary.

If the +15V and/or –15V outputs of the PS-2A power supply are abnormal, repair or replace the power supply.

Audio Processor Bands

Visually observe the action of the LEDs for each band on the front panel while program audio is being fed to the unit. If any band appears to act abnormally with respect to the other bands, swap the M-102 boards around. If the problem follows the M-102 board, then the M-102 audio processor board is probably defective and should be replaced.

NOTE: All M-102 audio processor boards are identical. Under normal operation any M-102 board may be exchanged with any other M-102 board in the system.

Symptom: No Output

Connect a signal source to the input of the Prism II AM and move the front panel BYPASS switch (S102) to the ON position. Make certain that the PINK NOISE switch is in the OFF position. Make certain that the Phase Rotator is in the circuit by placeing the jumper at Y301 of the mother board in the "IN" position. Adjust the front panel BYPASS GAIN (R115) control to the 12 o'clock position. Adjust the four front panel MIX LEVEL controls to the 12 o'clock position. Adjust the INPUT GAIN control (R101) and the OUTPUT GAIN control (R104) to approximately 12 o'clock positions and monitor the output of the Prism II AM. If no signal is present at the output, replace U101 and/or U301 as required. Move the front panel BYPASS switch (S102) to the OFF position and proceed.

Using an AC or audio voltmeter or oscilloscope and Drawings #8611 and #8612 (Appendix C), inject program material into the input of the Prism II AM as a test signal, and follow the signal flow through the system as described below. Ensure that the test signal is present at each of the test points described below before proceeding to the next test point.

Monitor the BLUE test point (TP303) on the MB-2 motherboard. If no signal is observed, check the input circuitry on the FP-1 Filter Protection board for continuity. Also check for proper operation of U301 and replace it if necessary.

Monitor the ORANGE test point (TP305) on the MB-2 motherboard. If no signal is observed, move the Phase Rotator jumper at Y301 to the BYPASS position. If the test signal now appears at the ORANGE test point (TP305), the PR-1 Phase Rotator is defective and must be replaced. U401 on the motherboard is the only active device in the Phase Rotator circuit.

Each of the M-102 audio processing boards has a BLACK test point (TP3). Check for the test signal at each of these points. If the test signal fails to appear at the BLACK test point (TP3), check for a defective M-102 board or U302, U303, U304, or U305 as required.

Monitor the RED, ORANGE, YELLOW, and GREEN test points on the front panel of the unit (TP201, TP202, TP203, and TP204 respectively). If the test signal fails to appear at any of these test points, check U302, U303, U304, or U305 as required. Check U201 as required. Also check the associated MIX LEVEL pots (R105, R106, R107, R108) if necessary.

Monitor the WHITE test point (TP205) located on the front panel of the Prism II AM. It is labeled "Broadband Sample." If the test signal fails to appear at this point, replace U101.

Monitor the VIOLET test point (TP304) located on the MB-2 motherboard. If the test signal fails to appear at this point, disconnect the cable from the connector labeled "Y1" of the AMC-3 board and jumper pins 4 and 5 of the cable. If the test signal now appears at the VIOLET test point on the front panel, the AMC-3 card is defective and must be replaced. If the test signal appears at this point but there is still no output from the unit, replace U306.

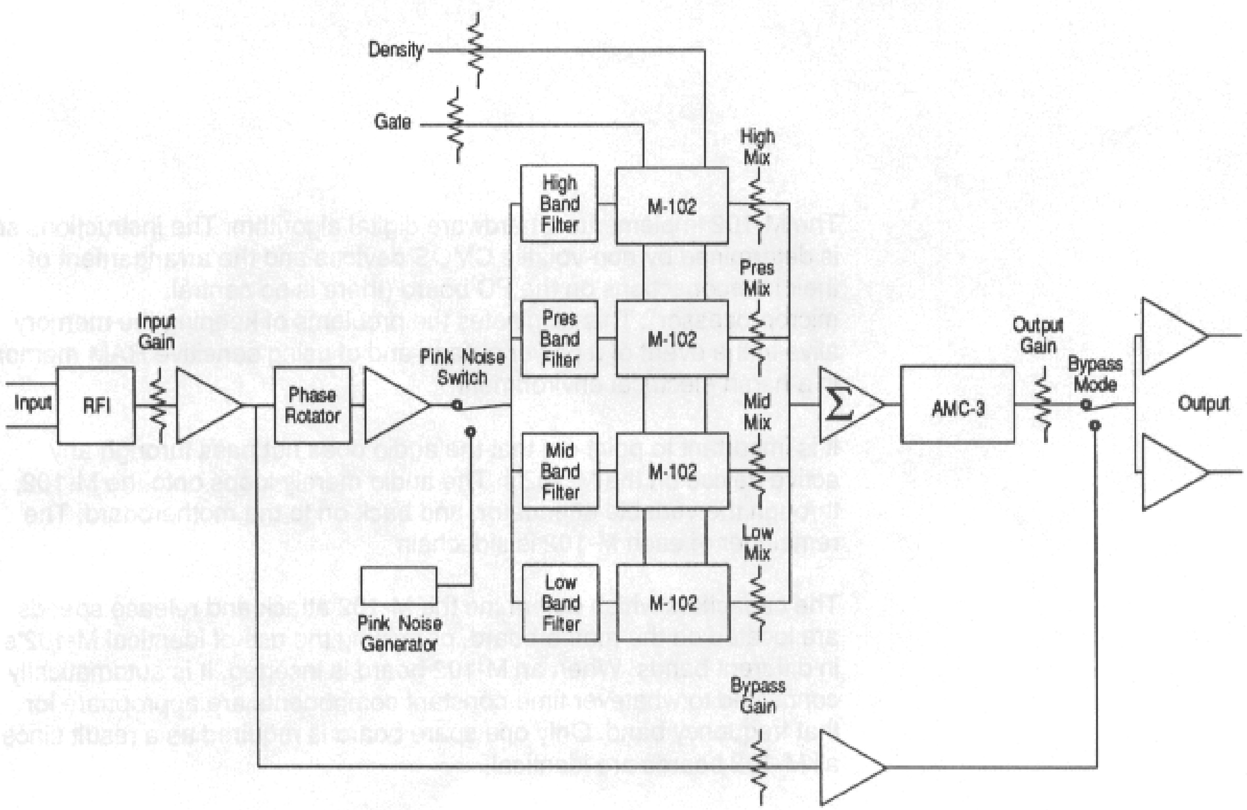
This completes the basic troubleshooting section. If the steps described here do not allow you to isolate and repair the Prism II AM, call Gentner Customer Support at (801) 975-7200 for further assistance.

5

Theory of Operation

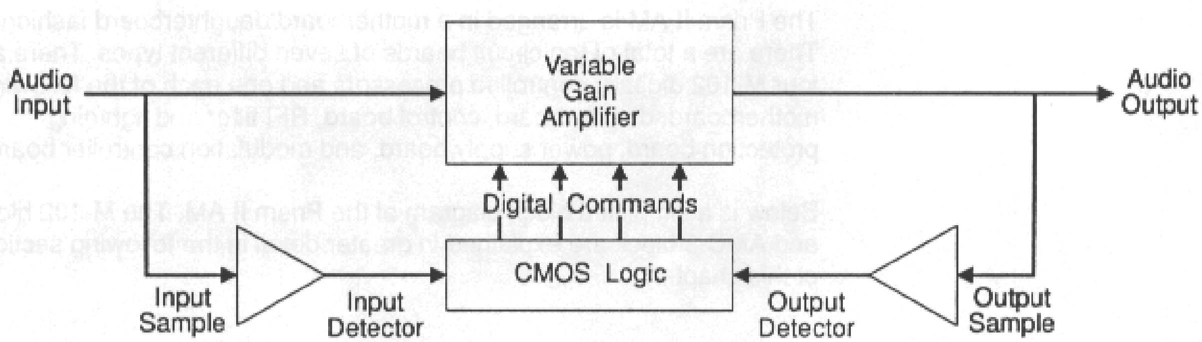
The Prism II AM is arranged in a motherboard/daughterboard fashion. There are a total of ten circuit boards of seven different types. There are four M-102 digitally controlled processors and one each of the following: motherboard, display board, control board, RFI filter and lightning protection board, power supply board, and modulation controller board.

Below is a simplified block diagram of the Prism II AM. The M-102 block and AMC-3 block are explained in greater detail in the following sections of this chapter.



The M-102 Processor

The heart is four digitally controled, model M-102 processor boards. The input and output signals of each fo the M-102's are level-detected as shown below. This information, plus reference voltages from the motherboard, is 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.



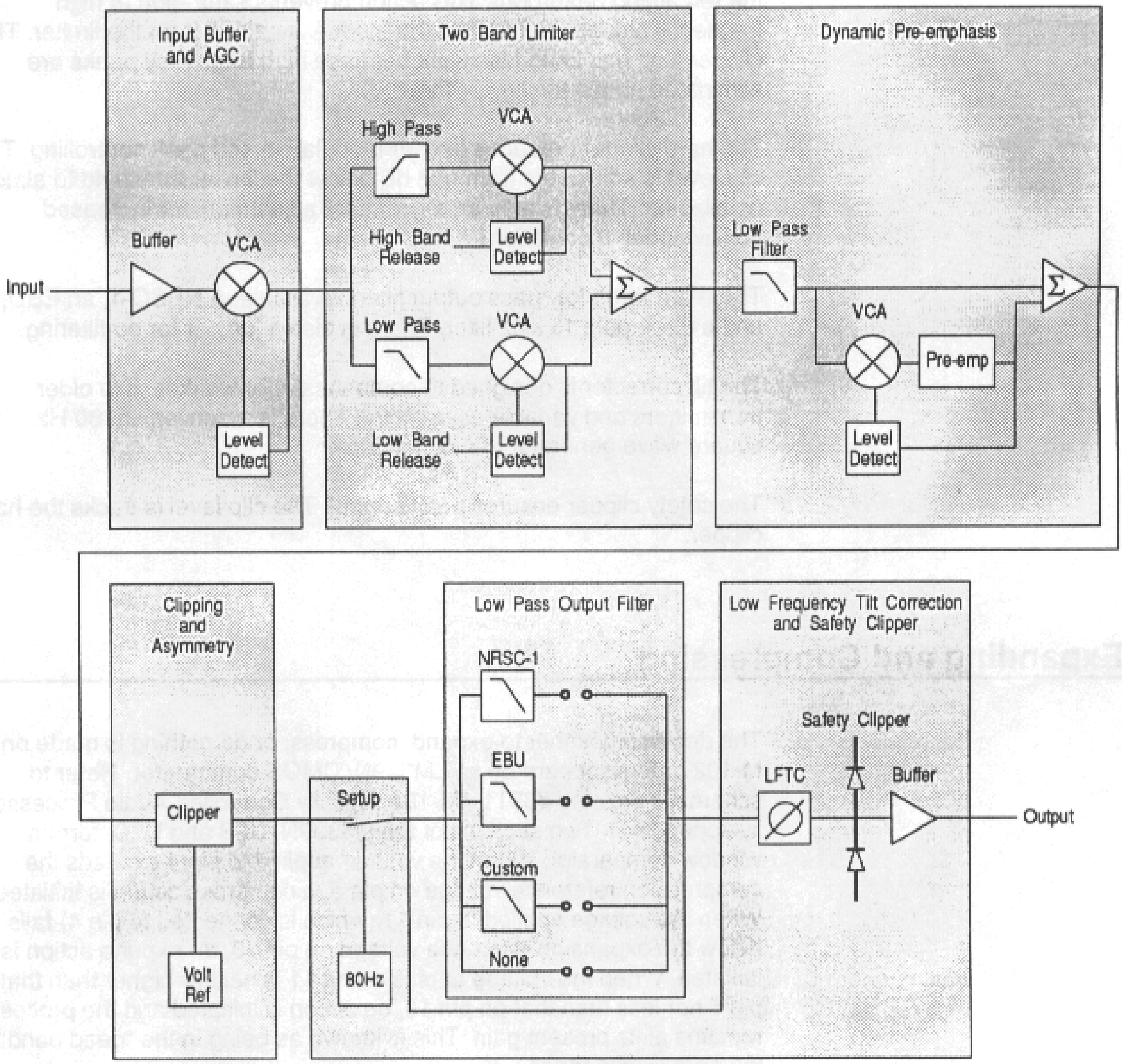
The M-102 implements a hardware digital algorithm. The instructions set is determined by non-volatile CMOS devices and the arrangement of their interconections on the PC board (there is no central microprocessor). This eliminates the problems of keeping the memory alive in the event of a power outage 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-102's. The audio merely loops onto the M-102, through the variable attenuator, and back on to the motherboard. The remainder of each M-102 is sidechain.

The capacitors which determine the M-102 attack and release speeds are located on the motherboard, permitting the use of identical M-102's in different bands. When an M-102 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 since all M-102 boards are identical.

The AM Modulation Controller (AMC-3)

The AMC-3 card is a high quality final amplitude controller for use in AM broadcasting. The AMC-3 employs an input AGC, a two band limiter, dynamic pre-emphasis, a peak clipper, a low pass output filter, a low frequency tilt corrector , and a safety clipper.



The input AGC has been designed to drive the limiter with a constant level. When the mix level potentiometers are adjusted, the audio level that feeds the AMC-3 changes. The AGC slowly adjusts this change to ensure a consistent audio level feeds the limiter.

The two band limiter employs the SSM-2120 Dynamic Range Processor. The high and low pass filters are of the single pole type, to eliminate combining effects in the stop band when the two bands are summed back together.

The dynamic pre-emphasis has four user selectable pre-emphasis times. The pre-emphasis circuit adjusts the amount of signal gain according to the remaining headroom. This action prevents saturation of high frequency signals and assists the loudness gained from the limiter. The clipper also has to do less work becuase high frequency peaks are eliminated above the limiter threshold.

The hard clipper produces denser modulation and peak controlling. The clip level is adjustable from one dB below the limiter threshold to almost no clipping. There is also an asymmetry adjustment for increased positive peak modulation.

There are three low pass output filters available. A NRSC-1, an EBU, and a three pole 15 kHz filter. There is also a jumper for no filtering.

The tilt corrector is designed to compensate for weakness in older transmitters and variable asymmetry. There is a convenient 80 Hz square wave generator to aid in setup.

The safety clipper ensures peak control. The clip level is tracks the hard clipper.

Expanding and Compressing

The decision whether to expand, compress, or do nothing is made on the M-102 processor card by an LM339N, CMOS comparator. Refer to schematic drawing #8511, "M-101 Digitally Controlled Audio Processor" in Appendix D. 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, you might expect that you 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 Prism II AM with this control by changing the size of the window comparator's permissible area.

Integrated Circuit Substitutions

The Prism II AM utilizes two families of Texas Instruments BIFET 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. ICs 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 ICs are really equivalents for the TL081 family. Only Exar 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 Prism II AM. Substitutions other than these may compromise the signal to noise ratio of the Prism II AM.

NUMBER OF SECTIONS	TEXAS INSTRUMENT 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	None	MC34004

The Prism II AM 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.

A

Forms

This appendix contains a Setup Log form for your use and convenience to assist you in adjusting the controls of the Prism II AM.

The form provided in this appendix is for your convenience and is exempt from the copyright notice in this manual and may be copied.

	MIX LEVELS (dBm)									
Date & Time	Low Band	Mid Band	Pres Band	High Band	Input Gain	Gate	Density	Output Gain	Comments	Initials
Example	4.0	4.0	2.0	-1.0	12 o'clock	0.5 VDC	4.25 VDC	12 o'clock	Starting Point	

B

Specifications

The following are static (steady-state) test data. Because broadcast audio processing systems are dynamic (non-steady-state) systems, the performance of such a device cannot be accurately reflected here. The proper method for evaluating audio processing equipment is by in-circuit evaluation and testing. The following is for informational purposes only.

Description

The Prism II AM is a digitally controlled audio processor for AM broadcast applications.

Topology

Parallel discriminate

Number of Bands

Four

Crossover Frequencies

153 Hz, 860 Hz, and 4900 Hz
(each band is 2½ octaves wide)

Input Impedance

600 ohm resistive terminating
(can be changed to 10K bridging)

Input level

0 dBm to +12 dBm as shipped from the factory (+4 nominal)
(can be modified to work with inputs as low as -10 dBm)

Output impedance

Dynamic Source Impedance:

with lightning protection circuit operational: 200 ohms
with lightning protection circuit deactivated: <1 ohm

Recommended load:

600 ohms or greater

Output Level

Nominal operating level:

+10 dBm

At saturation:

with lightning protection circuit operational: 18 volts peak-to-peak, +18.2 dBm
with lightning protection circuit deactivated: 24 volts peak-to-peak, +20.8 dBm

Harmonic Distortion

<0.2% at +10 dBm output at 2 kHz

Signal to Noise Ratio

-60 dB below +10 nominal output, weighted

Dimensions

Height:

1¾ inches (44.45 mm)

Width:

19 inches (482.6 mm)

Depth:

12¼ inches (311.2 mm)

Weight

7.5 pounds (3.4 kg)

Limiter Crossover

360 Hz

Limiter Drive

0 to +6 dB

Release Times

High:

1.1 sec to 100 msec

Low:

1.1 sec to 100 msec

Attack Times

High:

0.5 msec

Low:

2.0 msec

Clipping

+2 dBv above Limiter Threshold (max) to +11 dBv above Limiter Threshold (min)

Asymmetry

95% to 150%

Pre-emphasis

None
50 µsec
75 µsec (NRSC)
100 µsec (EBU)

Output Filters

None	flat to 20kHz
Custom	15 kHz corner, 18 dB/octave
NRSC	9.5 kHz corner, NRSC compliant
EBU	4.2 kHz corner, 54 dB/octave

Index

L

M

N

V

W